Session 3aAAa

Architectural Acoustics and Speech Communication: At the Intersection of Speech and Architecture II

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Chair's Introduction—8:00

Invited Papers

8:05

3aAAa1. Vocal effort and fatigue in virtual room acoustics. Pasquale Bottalico, Lady C. Cantor Cutiva, and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Lansing, MI 48910, pb@msu.edu)

Vocal effort is a physiological entity that accounts for changes in voice production as vocal loading increases, which can be quantified in terms of Sound Pressure Level (SPL). It may have implications on potential vocal fatigue risk factors. This study investigates how vocal effort is affected by room acoustics. The changes in the acoustic conditions were artificially manipulated. Thirty-nine subjects were recorded while reading a text, 15 out of them used a conversational style while 24 were instructed to read as if they were in a classroom full of children. Each subject was asked to read in three different reverberation time RT (0.4 s, 0.8 s, and 1.2 s), in two noise conditions (background noise at 25 dBA and Babble noise at 61 dBA), in three different auditory feedback levels (-5 dB, 0 dB, and 5 dB), for a total of 18 tasks per subject presented in a random order. The subjects answered questions addressing their perception of vocal fatigue on a visual analog scale. Babble noise and the order of task presentation increased SPL and self-reported fatigue. The SPL increased when the RT and the auditory feedback level decreased further clarifying how vocal effort changes within various conditions.

8:25

3aAAa2. Relationship of the difference of the speech rate of an announcement at the railway station and the listening impression. Sohei Tsujimura (Structures Technol. Div., Railway Tech. Res. Inst., 2-8-38 Hikari-cho, Kokubunji-shi, Tokyo 185-8540, Japan, tsouhei@rtri.or.jp)

The purpose of this study is to clear the influence of the speech rate of an announcement on the listening impression. In this study, a subjective experiment in which the elderly and the young adults participated was conducted in the simulated station. In this experiment, the speech rate of an announcement was varied from 4.5 to 9.5 mora/s at 1 mora/s interval, and subjective evaluations ("Listening difficulty," "Loudness," and "Strangeness") of the announcements were made under three types of background noise conditions (L_A 65 dB, 70 dB, and 75 dB). The relationships of the speech rate of an announcement and the subjective evaluation such as "Listening difficulty," "Loudness," and "Strangeness" were investigated, and the optimum speech rate of an announcement at the railway station was discussed. As a result, it was suggested that "Listening difficulty" has the lowest evaluation value when the speech rate is 6.5 mora/s or 7.5 mora/s. Furthermore, in this speech rate range, "Strangeness" as an announcement at the railway station was not felt. It was demonstrated that the optimum speech rate of an announcement at the railway station is in the range from 6.5 mora/s to 7.5 mora/s.

8:45

3aAAa3. Room acoustics and speech perception: Considerations for the developing child. Lori Leibold (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68124, lori.leibold@boystown.org)

What children hear in noisy rooms is not the same thing that adults hear. Despite precocious maturation of the peripheral auditory system, the ability to hear and understand speech in the presence of competing backgrounds sounds is not fully developed until adolescence. This talk will review results of behavioral studies of auditory masking in children, with a focus on the development speech perception in complex acoustical environments. Data will be presented that support the hypothesis that children's increased susceptibility to auditory masking relative to adults is related to immature perceptual processing such as sound source segregation and selective attention. Findings from studies investigating the extent to which children benefit from acoustic cues thought to facilitate sound source segregation will be highlighted.

Session 3aAAb

Architectural Acoustics and Signal Processing in Acoustics: Advanced Analysis, Simulation, and Auralization in Room Acoustics I

Michael Vorländer, Cochair
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Tetsuya Sakuma, Cochair
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Toshiki Hanyu, Cochair
Junior College, Department of Architecture and Living Design, Nihon University, 7-24-1, Narashinodai, Funabashi 274-8501, Japan

Chair's Introduction—9:20

Invited Papers

9:25

3aAAb1. Can you trust your numeric simulations—How to verify your code and validate your model. Lauri Savioja (Dept. of Comput. Sci., Aalto Univ., PO Box 15500, Aalto FI-00076, Finland, Lauri.Savioja@aalto.fi), Sebastian Prepeliţă, Pierre Chobeau (Dept. of Comput. Sci., Aalto Univ., Espoo, Finland), and Jonathan Botts (ARiA, Culpeper, VA)

In many fields where numeric simulations are utilized, there are standard practices on how to make sure that the simulation results are of high quality. In room acoustic simulations this seems to be quite rare although there are several factors that affect the accuracy of the simulations. First of all, there should be guarantees that the mathematical model, typically partial differential equations, actually model the physical phenomena under investigation. This can be made sure by validation and it can take place, for example, by comparing the simulation results to some reference solution or to measurement data. This is typically not a concern for the linearized wave equation although adding realistic boundary conditions makes it more challenging. Another essential factor affecting the correctness and reproducibility of the results is the quality of the implementation. Code verification is a procedure that aims to guarantee that an implementation is free of errors. In this paper, we review some common practices of code verification. As a practical example, we show verification studies conducted with our finite-difference time-domain solver. In addition, we show convergence rates obtained with the same solver to demonstrate the order of the accuracy of the underlying models.

9:45

3aAAb2. Crowd noise simulation system for sound environment evaluation of public spaces. Tetsuya Sakuma (Graduate School of Frontier Sci., The Univ. of Tokyo, 5-1-5 Kashiwanoha, Kashiwa 277-8563, Japan, sakuma@k.u-tokyo.ac.jp) and Yukiko Nishimura (Inst. of Technol., Shimizu Corp., Tokyo, Japan)

Aiming at the prediction and evaluation of sound environment of public spaces, a simulation-based auralization system is developed for reproducing background crowd noise. The system combines room acoustics modeling using a ray tracing method with a six-channel sound field reproduction system in an anechoic room. As sound sources, footsteps and voices of pedestrians moving on a floor, and HVAC noise from a ceiling are modeled. In a preliminary examination simulating several existing public spaces, a general correspondence to real sound field are confirmed in noise level and auditory impressions such as noisiness, liveness, and so on. Next, simulating a variety of imaginary rooms with changing room dimensions, absorption and pedestrian density, a subjective experiment of auditory pairwise comparisons is carried out. The results show that the noisiness fairly corresponds with the noise level, whereas the feeling of reverberation does not clearly related to the reverberation time, which may be due to the continuous overlap of sounds.

10:05–10:20 Break

10:20

3aAAb3. Binaural simulation using six channel reproduction based on the finite difference time domain room acoustic analysis. Shinichi Sakamoto (5th Dept., Inst. of Industrial Sci., The Univ. of Tokyo, 4-6-1, Komaba, Meguro-ku, Tokyo 153-8505, Japan, sakamo@iis.u-tokyo.ac.jp), Takatoshi Yokota, and Sakae Yokoyama (Kobayasi Inst. of Physical Res., Tokyo, Japan)

For auralization of room acoustics, various simulation-reproduction systems, such as ray tracing, cone tracing, image source methods, and wave-based numerical analysis for sound field simulation, and transaural system, multi-channel loudspeaker system for sound field reproduction, have been developed. As a sound recording-reproduction system, the six channel system has been developed, in

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which a sound signal is recorded by orthogonally located six directional microphones having a cardioid characteristics and the recorded signals are reproduced by orthogonally arranged six loudspeakers. To obtain binaural signals for auralization with headphones, the six channel reproduction was combined with binaural recording using dummy head microphones. The proposed concept can be efficiently applied to room acoustic simulation. As a room acoustic calculation method, finite-difference time-domain method was employed in this study. The calculated room impulse responses were reproduced by six channel reproduction system, and the binaural signals are recorded using dummy head microphones. In this presentation, the concept of the reproduction system is first stated and some examples on room acoustic problems are introduced.

10:40

3aAAb4. Evaluation of spatial impression of sound field in a concert hall based on the sound intensity using musical sounds. Toshiki Hanyu (Dept. of Architecture and Living Design, Junior College, Nihon Univ., 7-24-1, Narashinodai, Funabashi, Chiba 274-8501, Japan, hanyu.toshiki@nihon-u.ac.jp), Akiho Matsuo (Graduate School of Sci. and Technol., Dept. of Architecture, Nihon Univ., Funabashi, Chiba, Japan), and Kazuma Hoshi (Dept. of Architecture and Living Design, Junior College, Nihon Univ., Funabashi, Chiba, Japan)

Spatial impression is one of the most important factor for evaluating sound fields in concert halls. Objective indices, such as the lateral energy measures, for evaluating the spatial impression are standardized in the ISO-3382. These indices can be measured by using a figure-of-eight pattern microphone. On the other hand, details of spatial information of sound field can be depicted by measuring instantaneous sound intensity in the sound field. However, thus far, there is no index using the sound intensity. In this study, first the relationship between the existing objective indices and the sound intensity is examined. In particular, the existing objective indices are redefined based on the sound intensity. Secondarily, some objective indices are newly defined by using the sound intensity. These intensity-based indices can basically be calculated by using the sound intensity of impulse responses in a room. Third, potential of measuring the intensity-based indices by using musical sounds is examined. We investigated the intensity-based indices in several sound fields by using both impulse responses and musical sounds. Especially, the dependence of signals used in the measurements on the indices was examined.

11:00

3aAAb5. Acoustic feedback considerations in simulations of sound systems in rooms. Wolfgang Ahnert and Stefan Feistel (Ahnert Feistel Media Group, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu)

Computers are employed since about 30 years in the simulation of room- and electro-acoustics. Simulation tools have always considered sources, usually loudspeakers, but not microphones, which is the reason that acoustic feedback has not been considered by any simulation tool yet. As a new feature, a database for microphones is introduced analogous to the database of loudspeakers and other general sources. It is possible in this way to simulate the regeneration of sound over the close system loop, and estimate the maximum gain before feedback, and related influence on maximum sound pressure level and speech intelligibility. The paper briefly describes the important features of the new microphone database. Afterwards, the basic mechanism of acoustic feedback is explained, and the dependence of the feedback threshold of different sound sources and receivers as well as the acoustic properties of the room. By calculating the feedback threshold the headroom of the acoustic gain before feedback may be predicted. These threshold values are calculated by simulation and compared with measurements. This will be done for different loudspeaker-microphone arrangements in a selected conference hall.

11:20

3aAAb6. Sampling the sound field in auditoria using a large scale microphone array. Ingo B. Witew and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstrasse 5, Aachen 52074, Germany, ingo.witew@akustik.rwth-aachen.de)

Acoustical measurements in auditoria can be laborious with microphones to be placed in large sampling areas that are divided by rows of seating and separated by balconies. As a result, practical studies are often based on measurements of a relatively small number of selected source-receiver combinations. The goal of this paper is to present a new measurement, visualization, and analysis approach for complex wave fields. The measurement apparatus, capable of automatically sampling the sound field in auditoria over a surface of 5.30 m x 8.00 m in high resolution, is described. Based on data collected with the microphone array, a case study of how sound is reflected and scattered from a concert hall's boundaries is shown. The comparison of repeated measurements with and without the presence of chairs allows a new perspective on grazing sound propagation over theatre seating ("seat dip effect"). The presentation will conclude with the discussion of spatial fluctuations of acoustic properties as a factor of measurement analysis and uncertainty.

11:40

3aAAb7. Determination of boundary conditions for room acoustics simulation by application of inverse calculation models. Lukas Aspöck, Thomas Maintz, and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen, Kopernikusstr. 5, Aachen D-52074, Germany, las@akustik.rwth-aachen.de)

Simulation models for room acoustics simulation usually promise to deliver precise results for a certain frequency range. However, correct results can only be guaranteed if accurate input parameters are provided. Standardized methods for the determination of boundary conditions, the impedance tube and the reverberation chamber, include measurement uncertainties which might lead to different simulation results. These measurement methods also only capture valid results for special situations, e.g., normal or random wave incidence. This insufficient description of the boundary conditions makes it challenging to validate simulation models by comparing them to measured results. Even for rather simple room acoustic situations, state-of-the-art simulations fail to match measurements using in-situmeasurements or textbook values for the boundary conditions. In geometrical acoustics, these deviations can not only explained by insufficient measurement methods for the input data but also by different handling of reflection, scattering, and diffraction. To resolve the mismatch between measured and simulated results, optimization concepts have been proposed to determine input parameter sets for the simulation models. This work discusses two inverse approaches, evaluates their practicability for typical room acoustic scenarios, and presents the application in a comparison of room acoustic simulation software, the international round robin of auralization.

Session 3aAB

Animal Bioacoustics: Session in Honor of Whitlow Au I

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Marc Lammers, Cochair
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Chair's Introduction—7:50

Invited Papers

7:55

3aAB1. Something just clicked: Whitlow Au and the sonar of dolphins. Robert C. Gisiner (IAGC, 1225 North Loop West, Houston, TX 77008, bob.gisiner@iagc.org)

I suspect that no one, especially not Whit, imagined in 1970 that he would become the world's authority on dolphin biosonar, also known as active echolocation. Since that time Whit has amassed an astonishing list of hundreds of publications on the subject, including the seminal text on marine mammal echolocation, The Sonar of Dolphins, published in 1993. It may surprise many of the younger members of ASA to know that there was a Cold War "arms race" for knowledge about dolphin sonar and that Whit and his U.S. colleagues carried on a lively and mutually enriching correspondence with Soviet colleagues against a background of government intrigues and limitations on travel and conference attendance out of concerns about a "biosonar gap" every bit as worrisome to military leaders as the nuclear arms race (well, maybe not quite as worrisome). Since then Whit has branched out to many other fields of bioacoustics and mentored a diverse array of students at the University of Hawaii, but I will focus on the years at the Navy's marine mammal facility in Kaneohe, Hawaii, where I had the good fortune to work alongside Whit for several years.

8:15

3aAB2. The engineer in field biologist's clothing: Whitlow Au's academic years. Marc Lammers (Hawaii Inst. of Marine Biology, 46-007 Lilipuna Rd., Kaneohe, HI 96744, lammers@hawaii.edu) and Kelly Benoit-Bird (Monterey Bay Aquarium Res. Inst., Moss Landing, CA)

In 1993, the same year that The Sonar of Dolphins was published, Whitlow Au transitioned from the U.S. Navy's Naval Ocean Systems Center to the Hawaii Institute of Marine Biology. Thus began a second professional act, so to speak, of field-based research and mentoring the next generation of marine bioacousticians. An electrical engineer doing field biology may not seem like the natural order of things at first, but the combination of Whitlow's engineering and acoustic prowess coupled with the enthusiasm and energy of bright young biologists resulted in a powerful research team. Thanks to Whitlow's ingenuity and commitment to advancing the state of the art, Hawaii's coastlines and marine habitats around the world were explored like never before. A lack of funding or the availability of appropriate instrumentation were never an obstacle, but rather became opportunities for creativity and invention. Whitlow and his students' research on dolphins, their prey, snapping shrimp, humpback whales, and other taxa led to many pioneering breakthroughs both in biology and technology. While Whitlow's scientific legacy could be measured by his more than 200 publications, it is the hundreds of lives he influenced though his academic and professional service that will ultimately be his greatest mark.

8:35

3aAB3. Bioacoustics research in Asia, promoted by Whitlow W. L. Au. Ding Wang, Kexiong Wang (Inst. of Hydrobiology, CAS, 7 Donghu South Rd., Wuhan, Hubei 430072, China, wangd@ihb.ac.cn), SONGHAI LI (Inst. of Deep-Sea Sci. and Eng., Sanya, China), and Tomonari Akamatsu (National Res. Inst. of Fisheries Sci., Fisheries Res. Agency, Japan Fisheries Res. and Education Agency, Kanagawa, Japan)

The sonar of dolphins (1993) written by Whitlow W. L. Au accelerated underwater bioacoustics research, especially in Asian odontocetes. Before this book, very limited ultrasonic recording of odontocetes and bioacoustics research of marine mammals have been done in Asia. The Yangtze freshwater cetaceans provided opportunities for Asian researchers to study biosonar behavior, passive acoustic monitoring methods, and auditory physiology of odontocetes. The above works were accomplished with the Whitlow Au's preceding studies as basic references. The passive acoustic monitoring method developed from the Yangtze finless porpoises has been applied internationally in recent years. With encouragement and supports from Whitlow Au, biosonar research, auditory sensitivity measurement, and long term acoustic monitoring have been extensively conducted on Chinese white dolphins and oceanic finless porpoises, which are the key species in coastal environmental assessment, especially for the windmill farm construction area in Asia. In the meantime, Whitlow dispatched students to help our researches and also accommodated Asian students in Hawaii. So far, there are more than five Ph.D. students from Asia in underwater bioacoustics field supervised or partially supervised by Whitlow Au. Some of them have already grown up as leading researchers in marine mammal bioacoustics in Asia.

8:55

3aAB4. Biosonar detection range of mesopelagic patches by spinner dolphins in Hawaii. Whitlow Au (Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, wau@hawaii.edu), Marc O. Lammers (Univ. of Hawaii, Kaneohe, HI), and Jakob Jung (Bremen Univ. of Appl. Sci., Kaneohe, Hawaii)

Spinner dolphins (*Stenella longirostris*) in the near-shore waters of the Hawaiian islands forage on the mesopelagic boundary community (mbc) of organisms consisting of myctophids, mid-water shrimp, and small squids. They forage at night in a coordinated fashion swimming parallel to shore hunting for patches of prey that they can encircle and herd into a tight three-dimensional patch. A profiler housing a broadband echo-ranger that projected dolphin-like biosonar signals was used to measure the target strength of the mbc. Echoes consisted of a number of highlights bunched together with target strength between -45 and -55 dB based on a dolphin's integration window of 264 ms. Noise values collected by an autonomous acoustic recorder at midnight in the location where the profiler data were obtained were used to estimate the biosonar detection range of spinner dolphins for mesopelagic patches. The receiving directivity index and the width of the auditory filter for *Tursiops truncatus*were used to estimate the biosonar detection ranges of *Stenella longirostris* searching for mbc patches. Using the sonar equation, the biosonar threshold detection range of spinner dolphins was estimated to be approximately 100 plus m, more than sufficient range for the animals to formulate their prey herding behavior.

Contributed Papers

9:15

3aAB5. Using acoustics to examine odontocete foraging ecology: Predator-prey dynamics in the mesopelagic. Kelly J. Benoit-Bird (Monterey Bay Aquarium Res. Inst., 104 COEAS Admin Bldg., Corvallis, OR 97331, kbenoit@coas.oregonstate.edu), Brandon Southall (Southall Environ. Assoc., Inc., Aptos, CA), and Mark A. Moline (Univ. of Delaware, Lewes, DE)

From his expertise in biosonar, Whitlow Au brought a wealth of ideas on sonar use, design, and context to the study of wild cetaceans, resulting in great contributions to our understanding of odontocete foraging ecology. This contribution follows that foundation, using an integrated approach comprising echosounders deployed in a deep-diving autonomous underwater vehicle, ship based acoustics, visual observations, direct prey sampling, and animal-borne tags to explore the behavior of Risso's dolphins foraging in scattering layers off California. Active acoustic measurements demonstrated that Risso's dolphins dove to discrete prey layers throughout the day and night. Using acoustic data collected from the AUV, we found layers made up of distinct, small patches of animals of similar size and taxonomy adjacent to contrasting patches. Prey formed particularly tight aggregations when Risso's dolphins were present. Squid made up over 70% of the patches in which dolphins were found and more than 95% of those at the deepest depths. Squid targeted by dolphins in deep water were also larger, indicating significant benefit from these rare, physically demanding dives. Careful integration of a suite of traditional and novel tools is providing insight into the ecology and dynamics of predator and prey in the mesopelagic.

9:30

3aAB6. Echolocation behavior of the Icelandic white-beaked (*Lagenorhyncus albirostris*) dolphins: Now and then. Marianne H. Rasmussen (Husavik Res. Ctr., Univ. of Iceland, Hafnarstett 3, Husavik, Iceland 640, Iceland, mhr@hi.is), Jens Koblitz (BioAcoust. Network, BioAcoust. Network, Neuss, Germany), and Peter Stilz (BioAcoust. Network, BioAcoust. Network, Hechingen, Germany)

First studies of the echolocation behaviour of free-ranging white-beaked dolphins (*Lagenorhyncus albirostris*) were conducted in Faxafloi Bay in the Southwestern part of Iceland in the years 1997 to 1999. However, the sighting rate of white-beaked dolphins has decreased in that area since then and the current studies were conducted in the Northeastern part of Iceland. The aim of this study was to investigate the difference between normal clicks compared to clicks from buzz sequences. The recordings of the Icelandic white-beaked dolphins were conducted using a vertical linear 16-hydrophone array in Skjalfandi Bay, Northeastern Iceland during August 2015 and June 2016. The hydrophones were connected to NI-Boards and to a laptop computer on board using a sample rate of 1 MHz per channel. The group size of the dolphins varied from three individuals up to 30 animals in the area during the recordings. The dolphin echolocation clicks were recorded and it was possible to track individuals and to estimate beam-pattern from their clicks. Estimated beam pattern from 45 regular clicks gave -3dB BW of 9.6 degrees and maximum source level was 208 dB re. 1mPa. In addition buzzes with short inter-click-intervals down to 2.5 ms were recorded on all 16 channels.

9:45–10:00 Break

10:00

3aAB7. Echolocation behavior of endangered fish-eating killer whales (*Orcinus orca*) recorded from digital acoustic recording tags (DTAGs): Insight into subsurface foraging activity. Marla M. Holt, M. Bradley Hanson, Candice K. Emmons (NOAA NMFS NWFSC, 2725 Montlake Blvd East, Seattle, WA 98112, Marla.Holt@noaa.gov), Deborah A. Giles (Wildlife, Fish, & Conservation Biology, Univ. of California, Davis, Davis, CA), Jeffrey T. Hogan (Cascadia Res. Collective, Olympia, WA), and David Haas (Marine Sci. and Conservation, Duke Univ., Durham, NC)

Killer whales are apex predators with diet specializations that vary among ecotypes. Resident killer whales use broadband echolocation clicks to detect and capture fish prey in their underwater environment. Here, we describe the echolocation behavior of endangered Southern Resident killer whales using DTAGs to determine subsurface foraging activity and to assess the effects of vessel and noise on foraging behavior. We deployed 29 DTAGs on individually-identified killer whales and collected complimentary field data over four seasons in summer habitat. DTAGs had two hydrophones that each recorded sound at sampling rates of 192 or 240 kHz, and other sensors to reconstruct whale movement. Prey remains were opportunistically collected during tag deployments to validate feeding. Echolocation signals of the tagged whale were inferred from spectral content and the angle of arrival that corresponded to tag placement. Preliminary results reveal that individuals produced steady click trains during shallow dives then dove to deeper depths while clicking at higher repetition rates that graded into bottom-associated low-level buzzes before ascent occurred. These results, together with movement data, are reliable subsurface foraging cues in this endangered population that can be used to assess vessel effects on foraging behavior.

10:15

3aAB8. Echolocation parameters of toothed whales measured at sea. Jens C. Koblitz (BioAcoust. Network, Eichenallee 32 a, Neuss 41469, Germany, Jens.Koblitz@web.de), Peter Stilz (BioAcoust. Network, Hechingen, Germany), Lisa Steiner (Whale Watch Azores, Horta, Portugal), and Marianne H. Rassmussen (The Univ. of Iceland's Res. Ctr. in Húsavík, Husavik, Iceland)

Historically, data on toothed whale echolocation parameters and abilities were collected from captive animals. Acoustic parameters under investigation were inter-click-interval, spectral content, source level, directionality, emission direction, including correlation and variation of those parameters. Technological advances over the past decade have allowed collecting data on those parameters from animals at sea using acoustic recording tags or hydrophone arrays. Using a vertical, linear array of 16 hydrophones, echolocation clicks from harbor porpoises, white-beaked dolphins, common dolphins, and bottlenose dolphins were recorded around Iceland and the Azores. The animal's position at click production was computed for each click based on the time of arrival differences. Intensity and spectral differences at the array allowed measuring source levels, beam width, and spectral variation at different angles relative to on-axis. Advancing knowledge on the use and variation of echolocation signals of toothed whales in their natural habitat will allow widespread and effective use of acoustic monitoring.

10:30

3aAB9. Echolocation detection of fishing hooks and implications for the Hawaiian longline fishery. Aude F. Pacini, Paul E. Nachtigall, Adam B. Smith, Rock Owens, and Stephanie Vlachos (School of Ocean and Earth Sci. and Technol., Hawaii Inst. of Marine Biology, 46-007 Lilipuna Rd., Kaneohe, HI 96744, aude@hawaii.edu)

Interactions between marine mammals and fisheries have a biological and economic impact that is often detrimental to both fishermen and species of concern. False killer whale bycatch in the Hawaii longline fishery has exceeded the potential biological removal (PBR) triggering the designation of a take reduction team under the Marine Mammal Protection Act (MMPA). As an attempt to understand the importance of acoustic cues in depredation events, this study presents preliminary data looking at the echolocation ability of a false killer whale (Pseudorca crassidens) to detect a longline fishing hook at various distances. Using a go/no-go paradigm, the whale was trained to report the presence of the hook at distances varying in 50 cm increments. A total of 28 sessions of 25 trials each were collected and echolocation signals were recorded using a nine element acoustic array. Number of clicks, acoustic parameters, decision time and performance were recorded. The subject successfully reported the presence of the hook up to 6 m. This work presents evidence that false killer whales can easily detect fishing gear which could influence how they interact with longline fishery.

10:45

3aAB10. The characteristics of dolphin clicks compared across recording depths and instruments. Marc Lammers (Hawaii Inst. of Marine Biology, 46-007 Lilipuna Rd., Kaneohe, HI 96744, lammers@hawaii.edu), Julie N. Oswald (Bio-Waves, Inc., Encinitas, CA), Anke Kuegler (Marine Biology Graduate Program, Univ. of Hawaii, Honolulu, HI), and Eva M. Nosal (Abakai Int., LLC, Waianae, HI)

The identification of delphinid species on the basis of the characteristics of their acoustic signals is an important aspect of many passive acoustic monitoring efforts. The development of species classifiers relies on the assumption that species-specific signal characteristics will be consistent across different recording scenarios, including depth and instrumentation. However, this assumption has largely remained untested. Here, we report on an effort to examine whether and how the properties of echolocation clicks obtained from different delphinid species vary as a function of recording depth and the instrument employed. Field recordings of seven species of dolphins were obtained off Kona, Hawaii, and San Diego, CA, using a 250 m vessel-based vertical array composed of five microMARS recorders, two SoundTrap recorders, and four C75 broadband dipping hydrophones (Cetacean Research Technology). The clicks obtained were characterized on the basis of their spectral properties and duration for each recording depth and also compared among different instruments deployed at the same depth. Both depth and recording instrumentation influenced the click characteristics observed. However, the click properties of some species varied to a greater degree than others. These results suggest that developing species-specific classifiers based on dolphin clicks should be approached with caution and carefully validated.

11:00

3aAB11. From dolphin sonar to commercial scientific acoustic systems. Lars N. Andersen (Underwater Sci., Simrad, Kongsberg Maritime AS, P.O. Box 111, Horten 3191, Norway, lars.nonboe.andersen@simrad.com)

Dolphins have remarkable sonar systems for underwater navigation, object detection, object inspection, and general understanding of the environment. It is easy to be inspired by these advanced biosonar systems when designing new advanced commercial sonar systems. Similarly, it is also easy to be inspired when discussing bioacoustics with Whitlow Au. His work on dolphin sonar systems has been and still is of significant inspiration for many working with both biological and manmade sonar systems. Personal examples on how Whitlow Au has inspired development of commercial scientific echo sounders will be given. Characteristics of dolphin sonar and commercial scientific acoustic systems will be discussed.

11:15

3aAB12. The echolocation beam of bottlenose dolphins (*Tursiops truncatus*): High-resolution measurements of horizontal beam patterns and nearfield/farfield transitions. Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmf.org), James J. Finneran (U.S. Navy Marine Mammal Program, San Diego, CA), Brian K. Branstetter, Patrick W. Moore, Cameron Martin, and Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA)

The work of Whitlow Au and colleagues has demonstrated that dolphin biosonar forms a highly directional, forward-facing beam. In our recent studies, we have expanded upon previous work by making biosonar beam measurements using high-resolution hydrophone arrays with up to 48 hydrophones. Bottlenose dolphins were trained to echolocate on both physical targets and phantom echo generators, with clicks simultaneously recorded on all hydrophones at a sampling rate of 2 MHz. Target ranges (and simulated target ranges for phantom echoes) were varied in order to examine the resulting effects on the spatial characteristics of the acoustic field. The directivity index of the echolocation beam increased with increasing click level and center frequency, and recordings from extreme off-axis azimuths displayed a two-pulse pattern best explained by internal reflections off of the premaxillary bones. High-density hydrophone arrays placed near echolocating dolphins' heads demonstrated that a transition from the geometric nearfield to the farfield occurs at approximately 0.3 to 0.4 m from the melon. The results were remarkably similar to the earlier findings of Au and colleagues, and provide further information on the spatial characteristics of the acoustic field associated with dolphin biosonar. [Funding from ONR]

11:30

3aAB13. Transmission beam characteristics of a spinner dolphin (*Stenella longirostris*). Adam B. Smith, Aude F. Pacini (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, 1933 Iwi Way, Honolulu, HI 96816, adambsmi@hawaii.edu), Paul E. Nachtigall (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, Kailua, HI), Leo Suarez (Philippines Marine Mammal Stranding Network, Subic Bay, Philippines), Lem Aragonez (Inst. of Environ. Sci. and Meteorol., Univ. of the Philippines, Subic Bay, Philippines), Carlo Magno (Ocean Adventure, Subic Bay, Philippines), Gail Laule (Philippines Marine Mammal Stranding Network, Subic Bay, Philippines), and Laura N. Kloepper (Biology, St. Mary's College, South Bend, IN)

Transmission beam characteristics have been described in a small number of odontocete species, providing insight into the biological and ecological factors that have influenced the design of the outgoing echolocation beam. The current study measured the on-axis spectral characteristics and transmission beam pattern of echolocation clicks from a small oceanic delphinid, the spinner dolphin (*Stenella longirostis*). A formerly stranded individual was rehabilitated in captivity and trained to station underwater in front of a 16 element hydrophone array. Preliminary analysis of a subset of recorded clicks showed on-axis spectral characteristics with a mean center frequency of 68 kHz and a mean peak frequency of 52 kHz. The dolphin exhibited both a circular beam shape and also varying degrees of a dorso-ventrally compressed transmission beam. The mean angular beamwidth for all clicks was 16.6 and 14.3 degrees in the horizontal and vertical planes, respectively, though some clicks exhibited horizontal beamwidths that were almost twice as wide as the vertical beamwidth. The overall beam was generally broader than the beams of previously described species, with a mean directivity index for all clicks of 21.5 dB. These results are the first reports of transmission beam characteristics from a spinner dolphin.

11:45

3aAB14. Biosonar radiation field on the forehead of a Risso's dolphin during prey capture. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu), Hsin-Yi Yu (Inst. for Ecology and Evolutionary Biology, National Taiwan Univ., Taipei, Taiwan), Whitlow W. Au, Adam Smith (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kaneohe, HI), I-Fan Jen (Farglory Ocean Park, Hualien, Taiwan), Wei-Cheng Yang (Dept. of Veterinary Medicine, National Chiayi Univ., Chiayi, Taiwan), Ying-Ching Fan (Farglory Ocean Park, Hualien, Taiwan), Paul E. Nachtigall (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kaneohe, HI), and Lien-Siang Chou (Inst. for Ecology and Evolutionary Biology, National Taiwan Univ., Taipei, Taiwan)

On-animal suction cups with embedded hydrophones allow examination of how signals on the forehead of echolocating odontocetes relate to the internal anatomical structure and the transmission beampattern. Risso's dolphin (*Grampus griseus*) is an interesting species for this investigation due to the presence of a unique vertical groove in the middle of their forehead. In this study, a linear array of six broadband suction cup hydrophones were attached along the forehead groove of an adult female Risso's dolphin trained to catch freshly thawed dead squid in front of an eight-element far-field hydrophone array. The animal's movement was simultaneously observed using an underwater video camera. A total of nine successful prey captures were recorded. During each catch, the animal first emitted regular echolocation clicks, which quickly transitioned into buzzes (clicks with distinctively high repetition rate) until prey capture. The amplitude and relative time of arrival of these signals across all channels were analyzed. For a subset of trials, the relative amplitude distribution across channels vary significantly between regular clicks and buzzes in a manner that may be explained by beampattern changes. These observations were investigated jointly with data from the hydrophone array and interpreted in light of anatomical structure of the melon.

WEDNESDAY MORNING, 30 NOVEMBER 2016 — KAHILI, 7:50 A.M. TO 12:00 NOON

Session 3aAO

Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Ocean Acoustic Tomography: Active and Passive, Theory and Experiment II

Bruce Howe, Cochair
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Arata Kaneko, Cochair
Graduate School of Engineering, Hiroshima University, 1-4-1 Kagamiyama, Higashi-Hiroshima 739-8527, Japan

Hiroyuki Hachiya, Cochair
Tokyo Institute of Technology, 2-12-1 S5-17, Ookayama, Meguro-ku, Tokyo 152-8550, Japan

Invited Papers

7:50

3aAO1. Recent progress in coastal acoustic tomography in China: Experiments and data assimilation. Xiao-Hua Zhu, Ze-Nan Zhu, Xiaopeng Fan, Wen-Hu Liu, Chuanzheng Zhang, Menghong Dong, Yu Long, and Yun-Long Ma (Second Inst. of Oceanogr., State Oceanic Administration, 36 Baochubei Rd., Hangzhou 310012, China, xhzhu@sio.org.cn)

We will review three recent coastal acoustic tomography (CAT) experiments carried out in China: (1) A 15-day tomography experiment that was carried out for the first time in the Qiongzhou Strait to measure the major tidal current constituents, residual currents, and volume transport; (2) A high precision CAT experiment with 11 CAT systems performed in the winter of 2015 in the Dalian bay, China. The number of successful reciprocal transmission lines reached 51, which may be the highest number in ocean acoustic tomography history. The CAT results showed a very high accuracy of velocity measurements with a root-mean-square difference of 4 cm/s compared with moored ADCP measurements; (3) Rapid sampling CAT measurements were used to map the structure of nonlinear tidal (M_4 and

M_6) currents. The results indicate that M_4 is predominantly generated by the advection terms, while friction mechanisms are predominant for generating M_6. Finally, we will introduce the CAT data assimilated into an unstructured triangular grid ocean model (FVCOM) using the ensemble Kalman filter scheme. The assimilated velocities agreed better with independent ADCP data than those obtained by inversion and simulation, indicating that data assimilation of the CAT data is the optimal method for reconstructing the current field.

8:10

3aAO2. Real-time monitoring of coastal current and throughflow by ocean acoustical tomography with a time reversal responder. Hong Zheng (Zhejiang Ocean Univ., No. 1 Haida South Rd., Lincheng Changzhi Island, Zhoushan, Zhejiang 316022, China, seahzheng@msn.com) and Arata Kaneko (Hiroshima Univ., Higashi-Hiroshima, Japan)

The conventional Coastal Acoustic Tomography (CAT) system works in a controlled timing of sound transmission from acoustic stations surrounding an observation site. Data received in each station are sent to an integrated data center via communication network and a real-time mapping of current and sound speed is realized. In a new time reversal mirror CAT system (TRM-CAT) proposed, data received at a few land stations by time reversal responders gather all data transferred to the data center. The SNR of received signals is increased and the cost of field experiment is reduced. The TRM-CAT consists of two types of station, land and response stations. The land station is a conventional CAT station that sends coded signals to the response station and receives time reversal signals released. The response station work as a responder, which does not transmit any acoustic signals, until a signals coming from the land station are recorded in a memory, and then reconstructed as a time reversal data to transmit toward the land station at a pre-determined timing. One-way travel time and travel time difference are determined by a matched filter with codes. As a results, the average sound speed and flow velocity can be calculated in land stations. The TRM-CAT is designed and the system simulation is presented. Application to monitoring volume and heat transports through straits and rivers in real time is planned now. Finally, the TRM-CAT serves as a cost effective system for real-time mapping of velocity structures in the coastal sea.

Contributed Papers

8:30

3aAO3. An acoustic tomography method for imaging ocean structure with reverberated waves. Robert A. Dunn (Univ. of Hawaii at Manoa, 1680 East-West Rd., Honolulu, HI 96822, dunnr@hawaii.edu)

Seismic wide-angle imaging using ship-towed acoustic sources and networks of ocean-bottom seismographs is a common technique for exploring earth structure beneath oceans. In these studies, the recorded data are dominated by acoustic waves propagating as reverberations in the water column. Ignored by the earth scientist, this data offer an alternative approach for determining the structure of the oceans and advancing understanding of ocean heat content and mixing processes. A method, referred to as ocean acoustic reverberation tomography, is developed that uses the traveltimes of reverberated waves to image ocean acoustic structure. To demonstrate the feasibility of applying this method to existing and future data collected as part of marine seismic studies, a synthetic example is devised and the maximum resolution of the method is explored in the presence of different levels of data noise. Reverberation tomography offers significant improvement over classical ocean tomography in that it can produce images of ocean sound speed over the entire vertical height of the water column and along long survey profiles (100 + km), on a scale much finer (100s of meters) than traditional tomography. Variations in acoustic wave speed of $<$1 m/s are possible to detect with existing technology.

8:45

3aAO4. Toward underwater channel impulse response estimation using sources of opportunity. Kay L. Gemba, Santosh Nannuru, William Hodgkiss, and Peter Gerstoft (MPL, Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 446, La Jolla, CA 92037, gemba@ucsd.edu)

Passive acoustic tomography exploits the acoustic energy generated by sources with unknown spectral content such as sources of opportunity (e.g., cargo ships) to study the ocean. However, separating the channel impulse response (CIR) from a possibly moving, random radiator is non-trivial for coherent processing and the amount of unknown variables may leave the problem intractable. Using an incremental approach, we study the sparse CIR estimation problem for a single receiver. Second, we add multiple receivers, relax source spectrum knowledge, and study processor performance in resolving possibly coherent multipath arrivals across the vertical line array. Third, we discuss the extension to include multiple VLAs and sequential processing for a source with known location. Application of the Noise Correlation 2009 Experiment and the Santa Barbara Channel Experiment 2016 are discussed.

9:00

3aAO5. Deducing environmental mismatch from matched-mode processing ambiguity surface sidelobes. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Julien Bonnel (Lab-STICC(UMR CNRS 6285), ENSTA, Bretagne, France), Chris Verlinden (SIO, UCSD, New London, Connecticut), Margaux Thieury (Lab-STICC(UMR CNRS 6285), ENSTA, Bretagne, France), and Aileen Fagan (U.S. Coast Guard Acad., New London, CT)

The application of non-linear time sampling to broadband acoustic signals propagating in shallow water has made modal filtering possible using a single hydrophone. As a result, incoherent matched-mode processing (MMP) techniques are now practical using only single-hydrophone data. When MMP ambiguity surfaces are constructed from pairs of modes and plotted as a function of range and frequency, both the mainlobe and sidelobes form striations that embed information about the type and amount of environmental mismatch present between the modeled and true environment. Thus the degree of symmetry that a two-mode frequency-averaged ambiguity surface displays around the mainlobe provides a metric for identifying environmental mismatch. Acoustic invariant theory, combined with simulations, demonstrate how mismatched waveguide replicas can be used to (1) estimate the true bottom interface sound speed, (2) determine whether a downward-refracting waterborne profile is present in data, and (3) estimate waterborne sound speed profile slopes. The theory also explains why certain mismatched environments can generate very high MMP correlations, even with broadband signals. In a shallow water environment with a downward-refracting sound speed profile, taking full advantage of this approach requires reformulation of non-linear time-sampling methods to extract refractive modes. [Work sponsored by the North Pacific Research Board.]

Invited Papers

9:15

3aAO6. CANAPE-2015 (Canada Basin acoustic propagation experiment): A pilot tomography experiment in the Beaufort Sea.
Matthew Dzieciuch (SIO/UCSD, 9500 Gilman Dr., IGPP-0225, La Jolla, CA 92093-0225, mad@ucsd.edu) and Peter F. Worcester (SIO/UCSD, San Diego, CA)

During the summer of 2015, test transmissions were made from a ship-suspended source to a moored vertical line array in the Beaufort Sea. These were carried out at a variety of ranges in order to prepare for a year-long (2016-2017) tomography experiment. A major concern was the expected transmission loss (TL) under the ice for current climatic conditions in the Arctic. A greater proportion of smooth first year ice (rather than rough multi-year ice) was expected due to climate change. This should lead to less scattering and a lower TL. Acoustic propagation in the Artic includes propagation in a weak shallow duct as well as arrivals of deeper turning rays. The TL of these different regimes is compared to that of an experiment in a tropical ocean, the Philippine Sea. The initial analysis shows that the Arctic TL scales with range to the fourth power. This large TL could be because the actual ice conditions in the experimental area had a high proportion of multi-year ice in 2015.

9:35

3aAO7. Capabilities and challenges of ocean acoustic tomography in Fram Strait.
Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Thormøhlensgate 47, Bergen N-5006, Norway, hanne.sagen@nersc.no) and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

The Fram Strait is of great importance in ocean climate monitoring, as it is the only deep-water connection between the Arctic and Atlantic Oceans. Even though an extensive array of oceanographic moorings has been operated in Fram Strait since 1996 to monitor the transports through the Strait, the small spatial scales of the flow are poorly resolved, leading to large uncertainties. Beginning in the 2005-2010 DAMOCLES (Developing Arctic Modeling and Observing Capabilities for Long-term Environmental Studies) project, underwater acoustic methods were introduced to improve the monitoring of Fram Strait. A 2008-2009 pilot study with a single acoustic path was followed during 2010-2012 in the ACOBAR (ACoustic technology for OBserving the interior of the ARctic Ocean) project by the implementation of a multipurpose acoustic network with a triangle of acoustic transceivers for ocean acoustic tomography, ambient noise, and glider navigation. The measurements were continued during 2014-2016 in UNDER-ICE (Arctic Ocean Under Melting Ice), with eight acoustic paths crisscrossing the Fram Strait at 78-800N. Tomography in Fram Strait is demanding. The sound-speed field has a weak sound channel with little geometric dispersion, making it difficult to resolve and identify individual arrivals. The strong oceanographic variability in space and time reduces the coherence of the received signal and the stability of the arrival pattern. The status of tomography in Fram Strait will be summarized, focusing on the capabilities and challenges.

9:55–10:10 Break

Contributed Papers

10:10

3aAO8. Acoustic noise interferometry and ocean dynamics.
Oleg A. Godin (Phys. Dept., Naval Postgrad. School, NPS 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu)

Noise interferometry relies on averaging long time series to reveal the deterministic features of the noise cross-correlation function, which contain information about the propagation medium. Unlike the solid earth, the ocean exhibits considerable variability at the time scales that are shorter than the necessary noise averaging times of hours and days. This variability is a primary factor that limits useful noise averaging times and underlies the more limited applicability of acoustic noise interferometry than of the better established seismic interferometry. This paper aims to quantify the restrictions, which are imposed on the noise interferometry by the ocean variability, and to identify possible applications of passive acoustic remote sensing to characterizing ocean dynamics. A theory will be presented that quantifies the coherence loss of measured noise cross-correlations due to gravity waves on the ocean surface, tidally induced changes in the water depth, and internal gravity waves. It is found that temporal variations of the ocean parameters lead to frequency-dependent suppression of deterministic features in the noise cross-correlation functions. The coherence loss limits the resolution of deterministic inversions. On the other hand, the passively measured coherence loss can be inverted to statistically characterize ocean dynamics at unresolved spatial and temporal scales.

10:25

3aAO9. Inferring salinity microstructure using high-frequency reciprocal acoustic transmission.
Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu)

The Connecticut River Estuary is highly stratified, strongly influenced by fresh water discharge and tides, and characterized by hydrographic features such as dynamic fronts, plumes, internal waves, shear instability, and the incoming salt-wedge. An experiment was conducted to measure the impact of these hydrographic features on the propagation of high-frequency sound, and to use the path-averaged acoustic fields to infer properties of salinity microstructure. Two 120 kHz, 4-element arrays, each element with reciprocal transmission capabilities, were deployed for 5 days mid-water-column, spanning multiple tidal cycles and a range of discharge conditions. Extensive environmental data were collected, including ADCP profiles, CTD profiles, salinity microstructure at various depths, and broadband acoustic backscatter (100-600 kHz). Results from this experiment show that 1) shear instabilities dominated the structure of the sound propagation during the ebb tide and were coherent features across the propagation path length, 2) salinity stratification could be deduced from reciprocal transmissions, 3) the slope of the shear instabilities could be inferred by combining the ADCP data and arrival times at different depths, and 4) and the incoming salt-wedge was the dominant hydrographic feature affecting the sound propagation during the flood tide. [This work was funded by the ONR Ocean Acoustics Program.]

10:40

3aAO10. Information content of ocean travel-times in the Philippine Sea. Brian Powell, Sarah Williamson, and Xiaofeng Zhao (Oceanogr., Univ. of Hawaii, 1000 Pope Rd., MSB, Honolulu, HI 96822, powellb@hawaii.edu)

The Philippine Sea is a highly energetic region of varying scales from planetary down to tidal. Significant eddy activity is present throughout the region with enhanced mesoscale activity centered around 22N. Tidally driven internal waves emit from the Luzon Strait into the Philippine Sea and are the most energetic in the world. These multiple dynamical scales combine to have significant influence on the density structure and sound speed in the region. In 2010, a large deep-water acoustic transmission experiment was conducted to investigate the acoustics in such a dynamic environment. A total of seven transceivers were used for providing significant coverage over the western Philippine Sea. In this talk, we use a data-assimilative numerical ocean model to investigate how travel-times from the acoustic array covary with the ocean. Using both predictions and analyses from the model, we investigate how well the model is capable of resolving acoustic rays and travel-times without the constraint of the acoustic observations. We then work to quantify how the ocean covaries with acoustic travel-times to help us to understand the variability in the ocean that controls the propagation and what the propagation can tell us about the oceanic variability.

10:55

3aAO11. Deblurring acoustic signals for statistical characterization in applications of ocean acoustic tomography. Viktoria Taroudaki (Biomedical Image Computing Group, Dept. of Pediatrics, Univ. of Washington, 1959 NE Pacific St., Seattle, WA 98195, victtar@uw.edu), Costas Smaragdakis, and Michael Taroudakis (Dept. of Mathematics and Appl. Mathematics & IACM, Univ. of Crete and FORTH, Heraklion, Greece)

The signal characterization method suggested by Taroudakis et al. (J.Acoust. Soc. Am. **119**, 1396-1405 (2006)) based on the statistics of its 1-D wavelet transform coefficients and successfully applied for inverting acoustic signals in applications of acoustical oceanography has been proven to be sensitive to noise contamination of the signal, but still, it provides good inversion results if an appropriate denoising strategy is applied. In this work, the statistical signal characterization is applied to signals which are both blurred and noise contaminated. Deblurring of the signal is achieved by means of a technique introduced by Taroudaki and O' Leary (SIAM J. Sci. Comput. **37**(6), A2947-A2968 (2015)) for image deblurring, and it is based on a statistical near optimal spectral filtering technique that takes advantage of the singular values of the approximated blurring matrix and the Picard Parameter of the signal that allows for estimation of the additive noise properties and estimation of the error. The study is extended to cases when no accurate knowledge of the blurring mechanism is available. It is shown by typical simulated experimental data that the combination of deblurring and simple denoising strategies provide good results with respect to both signal characterization and subsequent inversions.

11:10

3aAO12. Acoustic mode travel time variability in the Philippine Sea. Tarun K. Chandrayadula (Ocean Eng., IIT Madras, 109 B Ocean Eng., IIT Madras, Chennai, Tamil Nadu 600036, India, tkchandr@iitm.ac.in), John A. Colosi (Naval Postgrad. School, Monterey, CA), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., San Diego, CA)

During the 2009 NPAL (North Pacific Acoustic Laboratory) PhilSea deep water acoustic propagation experiment, a low frequency source transmitted broadband chirps to a mid-water spanning hydrophone array at a distance of 185 km every three hours for approximately one month. In addition, transmissions took place every five minutes during some time periods. The source and the receiver array contained conductivity and temperature sensors. The motions of the source and receiver moorings were measured using long-baseline acoustic navigation systems. The experiment site was oceanographically dynamic and contained significant internal tide activity. This work builds a model to compare the acoustic and oceanographic variabilities. The acoustic observations are used to estimate the mode travel times, and then related to the internal tide displacements at the source and the receiver arrays. This paper presents the work in three parts. The first part uses matched subspace detectors to estimate the mode travel times. The second uses the environmental observations to build an internal tide model. The third constructs a linear perturbation model to relate the internal tides to the adiabatic mode travel times. The discrepancies between the model and the observations are attributed to factors such as limitations of the perturbation model for the travel times, range variability not accounted for in the adiabatic mode propagation, and signal processing issues.

11:25

3aAO13. Deep water sound sources for ocean acoustic tomography and long-range navigation. Andrey K. Morozov and Douglas C. Webb (Teledyne, 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com)

The first test of the tunable organ-pipe was successfully conducted on 11 Sept. 2001. The sound source was efficient, powerful, and had unlimited operational depth, as well as a minimum level of high frequency harmonic content. The projector uses a narrow-band, highly efficient sound resonator, which is tuned to match the frequency and phase of a reference frequency-modulated signal. Since 2001, many deep-water ocean acoustic experiments have used this type of sound source. The source was successfully used for ocean acoustic tomography and long range navigation. Recently, it was shown that a bottom-deployed swept frequency array can be used for high-resolution seismic imaging of deep geological formations. The long-term operating experience of the sound sources demands the development of an improved sound source system. A finite-element analysis that shows the structural acoustics of the tunable resonator has been conducted to improve the acoustics of the sound source. The analysis gave a better solution for a tuning mechanism with an octave frequency range. The development of the electric and mechanical systems of the sound source will be discussed. Teledyne Marine Systems continues innovating promising solutions for ocean acoustic tomography, navigation, and seismo-acoustic applications.

Invited Paper

11:40

3aAO14. Recent progress in coastal acoustic tomography. Arata Kaneko (Graduate School of Eng., Hiroshima Univ., Hiroshima University, Higashi-Hiroshima 739-8527, Japan, akaneko@hiroshima-u.ac.jp), Fadli Syamsudin, Yudi Adityawarman (Agency for the Assessment and Application of Technol. (BPPT), Jakaruta, Indonesia), Hong Zheng (Zhejiang Ocean Univ., Zhoushan, China), Chen-Fen Huang, Naokazu Taniguchi (National Taiwan Univ., Taipei, Taiwan), Xiaohua Zhu (State Oceanic Administration, Second Inst. of Oceanogr., Hangzhou, China), Ju Lin (Ocean Univ. of China, Qingdao, China), and Noriaki Gohda (Graduate School of Eng., Hiroshima Univ., Higashi-Hiroshima, Japan)

Coastal acoustic tomography (CAT) which was proposed by Hiroshima University in 1990s as a shallow-sea application of ocean acoustic tomography (OAT) is developed as a mirror-type CAT (MCAT) for measuring deep strait throughflows in Indonesian archipelago seas in real time. MCAT system is composed of a land station (M0) connected to a 100 m submarine cable edged by a 5 kHz subsurface transceiver and triangular-arrayed bottom-moored stations (M1, M2, and M3). Reciprocal data are first obtained among three station pairs (M1M2, M2M3, and M3M1). Data received at M1 from M2 and M3 are transferred to the land station (M0) by the first

mirror reflection, resulting in travel time summations (t21 + t10, t31 + t10) (t21: travel time from M2 to M1). Data obtained at the offshore stations M2 and M3 are transferred to the nearshore station M1 by the first mirror reflection, resulting in travel time summations (t12 + t21, t13 + t31, t23 + t31, and t32 + t21). All the first mirror data at M1 are also transferred to M0 by the second mirror reflection, adding a travel time by t10 for the above four travel time summations. Finally, travel time differences are calculated for M0M1, M1M2, M2M3, and M3M1 by subtracting the summation data at M0. They are converted to path-average currents along the four transmission lines, succeeding in strait throughflow estimate. The mirror reflection function of MCAT can develop in an underwater transfer system of sound from moving sources by adding a hydrophone.

WEDNESDAY MORNING, 30 NOVEMBER 2016 CORAL 1, 8:00 A.M. TO 12:00 NOON

Session 3aBA

Biomedical Acoustics: Quantitative Ultrasound I: From Micro to In-vivo

Jonathan Mamou, Cochair
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Tadashi Yamaguchi, Cochair
Chiba University, 1-33 Yayoicho, Inage, Chiba 2638522, Japan

Invited Papers

8:00

3aBA1. Emerging quantitative ultrasound applications: From diagnosing disease to monitoring of therapy. Michael L. Oelze (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Mathews, Urbana, IL 61801, oelze@uiuc.edu)

Ultrasound has traditionally been considered an imaging modality characterized by high spatial resolution but low contrast. Conventional ultrasonic imaging may be sensitive to the detection of anomalous tissue features, but the ability to classify these tissue features often lacks specificity. However, recent advances in quantitative ultrasound (QUS) have emerged which can improve diagnostics by providing new sources of image contrast and specific numbers tied to tissue state. QUS imaging can encompass a wide variety of techniques including spectral-based parameterization, elastography, flow estimation, and envelope statistics. Spectral-based techniques include the estimation of the backscatter coefficient, estimation of attenuation, and estimation of scatterer properties such as the correlation length associated with an effective scatterer size and the acoustic concentration of scatterers. Envelope statistics include the estimation of the number density of scatterers and quantification of coherent to incoherent signals produced from the tissue. In this talk, successful applications of QUS will be discussed demonstrating the ability of spectral-based QUS and envelope statistics approaches to improve medical diagnostics including cancer detection and classification of solid tumors in the breast or thyroids, grading of fatty liver disease and monitoring and assessment of therapy in solid tumors.

8:20

3aBA2. Selected quantitative ultrasound successes. Aiguo Han and W. D. O'Brien, Jr. (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois, 405 N. Mathews Ave., Urbana, IL 61801, han51@illinois.edu)

The objective of soft tissue quantitative ultrasound (QUS) is to improve diagnostic ultrasound imaging via quantitative outcomes. Over the past three or so decades, there have been an increasing number of QUS successes. Four QUS successes will be highlighted, from theoretical development to clinical applications. Significant ultrasonic scattering mechanistic details have resulted from the spatial correlation among the scatterers (modeled by the structure function) as well as from the individual scatterers (modeled by the form factor), for which there have been successes with both theory and applications. In collaboration with colleagues at the University of California, San Diego, nonalcoholic fatty liver disease (NAFLD) is being assessed from model-free QUS parameters from which both the backscatter coefficient and the attenuation coefficient show dependencies with the liver fat content in human subjects. Likewise, in collaboration with colleagues at the University of Illinois at Chicago, spontaneous preterm birth is being assessed as early as 20 weeks of gestation in pregnant women from the model-free QUS parameter attenuation coefficient. [NIH Grants NIH R01DK106419 and R37EB002641.]

8:40

3aBA3. Backscatter coefficient estimation in human thyroid carcinoma *In vivo*. Julien Rouyer (Pontificia Universidad Católica del Perú, Av. Universitaria 1801, San Miguel, Lima 32, Peru,), Rosa Laimes, Claudia Salazar, Joseph Pinto, Jorge Guerrero (Oncosalud, AUNA, Lima, Peru), and Roberto Lavarello (Pontificia Universidad Católica del Perú, San Miguel, Lima 32, Peru, lavarello.rj@pucp.edu.pe)

Spectral-based quantitative ultrasound (QUS) characterizations may provide a potential alternative to invasive gold standard technique—fine-needle aspiration (FNA)—in the diagnostic management of the thyroid cancer. Such potentiality was previously demonstrated in mouse thyroid cancer models *ex vivo*. Recently, the feasibility of the backscatter coefficient (BSC) estimation of human thyroid *in vivo* was demonstrated for a controlled group of 20 healthy volunteers (24.4 $\pm$ 2.9 year-old). Among the volunteer, the integrated BSC parameter was found equal to $1.41\text{x}10^{-2}\text{sr}^{-1}.\text{cm}^{-1}$ in the 3 to 9 MHz range. The current study focuses on the estimation of the BSC in a clinical context. Preliminary results on patients having malignant lesions are proposed. A research ultrasound imaging system (SonixTouch with an L14-5/38 linear array, Ultrasonix Medical Corp.) was employed by the radiologists of an oncological center first to record radio frequency data of the lesion and second to guide the FNA. The reference phantom technique was used in the same fashion as for the previous healthy volunteer's study. The experimental protocol was approved by an Institutional Review Board, and all enrolled patients provided written informed consent. A controlled group of seven patients with a diagnosed papillary or follicular carcinomas were included in this pilot study to assess the ability of the technique to estimate the BSC and derived parameters in the lesions.

9:00

3aBA4. Quantitative ultrasound for tumour characterization and a priorichemotherapy response prediction. Gregory Czarnota (Sunnybrook Health Sciences Ctr., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, gregory.czarnota@sunnybrook.ca)

Previous studies have demonstrated that quantitative ultrasound (QUS) is an effective tool for monitoring breast cancer patients undergoing neoadjuvant chemotherapy. Here, for the first time, we demonstrate the clinical utility of pre-treatment QUS texture features in predicting the response of breast cancer patients to NAC. Using a 6 MHz center frequency clinical ultrasound imaging system, radiofrequency breast ultrasound data were acquired from 92 locally advanced breast cancer patients prior to their NAC treatment. QUS spectral parameters including mid-band fit, spectral slope, and spectral intercept, and average acoustic concentration and average scatterer diameter were computed from regions of interest in the tumor core and its margin. Subsequently, gray-level co-occurrence matrix textural features were extracted from the parametric images as potential predictive indicators. Results were compared with the clinico-pathological response of each patient determined at their treatment end. A composite QUS feature demonstrated a favourable response prediction with sensitivity, specificity, and AUC of 86%, 83%, and 0.77, respectively, using a Naïve Bayes classifier. The findings here suggest that QUS features of breast tumors are strongly linked to tumor responsiveness. The ability to identify patients that would not benefit from NAC would facilitate clinical management that has minimum patient toxicity and maximum outcome.

9:20

3aBA5. Quantitative ultrasound of the uterine cervix. Timothy J. Hall (Medical Phys., Univ. of Wisconsin, 1005 WIMR, 1111 Highland Ave., Madison, WI 53705, tjhall@wisc.edu), Helen Feltovich (Medical Phys., Univ. of Wisconsin, Provo, UT), Ivan M. Rosado-Mendez, Lindsey C. Drehfal, Quinton W. Guerrero (Medical Phys., Univ. of Wisconsin, Madison, WI), and Mark L. Palmeri (Biomedical Eng., Duke Univ., Durham, NC)

We are developing methods to objectively describe the acoustic and viscoelastic properties of the *in vivo* cervix to monitor the structural changes that occur during pregnancy. Clinicians commonly use digital palpation to subjectively judge cervical softness knowing the cervix softens prior to parturition. Premature softening of the cervix is a risk factor for preterm birth which has enormous morbidity and financial consequences. However, this is a challenging tissue for quantitative ultrasound (QUS) investigations because its microstructure is layered and pseudo-aligned within each layer. Also, data are typically acquired with a transducer that is in contact with the outer layer of the cervix, so QUS analysis is necessarily done very close to the transducer surface. To avoid violating assumptions used in QUS algorithm development, we use a prototype transducer that allows us to perform QUS parameter estimation as close as 3mm from the transducer surface. Further, we systematically test for various forms of echo signal coherence instead of assuming the backscatter is from diffuse scattering. Preliminary results in humans and non-human primates suggest the cervix significantly softens in early pregnancy, but the underlying collagen microstructure remains intact until very near delivery. Methods for this analysis and results will be briefly described.

9:40–10:00 Break

10:00

3aBA6. Compressing and shearing the transplanted kidney. Matthew W. Urban (Dept. of Radiology, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu), Carolina Amador, and Sara Aristizabal (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN)

Kidney transplantation is one treatment for end-stage renal disease. Transplant rejection is typically evaluated with an invasive biopsy. Noninvasive methods for evaluating renal transplants would be beneficial for frequent evaluation. We propose the use of shear wave elastography (SWE) to evaluate viscoelastic properties in renal transplants. Further, we also use acoustoelasticity to measure the nonlinear modulus of the renal transplants. Acoustoelasticity is the phenomenon where the shear wave velocity (SWV) in a material varies with applied stress and is quantified by a nonlinear modulus, A. By applying pressure with the ultrasound transducer, stress is applied to the kidney under investigation and B-mode imaging was used to measure induced strain and SWE was used to make measurements of SWV. Using relationships involving the measured strain and shear moduli at different compression levels, we can obtain estimates of A. In our ongoing study, we use a General Electric Logiq E9 to perform SWE measurements in renal transplant patients coming for protocol biopsies. We have studied 63 patients with SWE to quantify viscoelastic properties of the transplanted kidney. We have obtained data from 19 patients for estimation of the nonlinear modulus, A. We compared the measured viscoelastic and nonlinear shear parameters with clinical variables such as the serum creatinine, Banff biopsy scores, and resistive index. We continue to evaluate these methods and their efficacy as indicators for transplant rejection.

10:20

3aBA7. Passive elastography: A seismo-inspired tomography of the human body. Stefan Catheline (INSERM U1032, 151 cours albert thomas, Lyon 69003, France, stefan.catheline@inserm.fr)

Elastography, sometimes referred as seismology of the human body, is an imaging modality recently implemented on medical ultrasound systems. It allows to measure shear waves within soft tissues and gives a tomography reconstruction of the shear elasticity. This elasticity map is useful for early cancer detection. A general overview of this field is given in the first part of the presentation as well as latest developments. The second part, is devoted to the application of noise correlation technique in the field of elastography. The idea, as in seismology, is to take advantage of shear waves naturally present in the human body due to muscles activities, heart beating, and arteries pulsatility to construct shear elasticity map of soft tissues such as the liver, the thyroid, or the brain. It is thus a passive elastography approach since no shear wave sources are used.

10:40

3aBA8. Elasticity imaging of inhomogeneous media using inverse filtering with multiple shear wave generation. Tsuyoshi Shiina (Human Health Sci., Graduate School of Medicine, Kyoto Univ., 53 Kawahara-cho Shogoin, Sakyo-ku, Kyoto, Kyoto 6068507, Japan, shiina.tsuyoshi.6w@kyoto-u.ac.jp)

Most diseases like cancer cause to change of tissue mechanical properties described by elasticity. Elastography visualizes these changes and aids in early and differential diagnosis of diseases. Shear wave imaging is one of principles of elastography, and using the speed of shear wave generated by mechanical vibration or the acoustic radiation force impulse. Conventional shear wave imaging uses time-of-flight method; however, it causes artifact by reflection and refraction of shear wave. We proposed a method for shear wave elasticity imaging that combines inverse filtering with multiple shear wave sources induced by acoustic radiation force. Shear waves are induced by pushing pulse and recorded by ultrafast imaging, and repeated at multiple pushing points that are sparsely located. An inverse filter can be applied to virtually focus shear waves on an arbitrary point. Shear wave speed is obtained by a measured shear wavelength and its image is constructed by scanning the focal point. An inverse filter can focus a point in a reverberant field. Thus, the proposed method is not based on the assumption of propagating direction, and improved estimation accuracy can be expected. The feasibility of the proposed method was demonstrated in phantom experiments.

11:00

3aBA9. Quantitative diagnosis method based on model of tissue structure change for liver fibrosis. Hiroyuki Hachiya, Shinnosuke Hirata (Tokyo Inst. of Technol., 2-12-1 S5-17, Ookayama, Meguro-ku, Tokyo 152-8550, Japan, hachiya@ctrl.titech.ac.jp), and Tadashi Yamaguchi (Chiba Univ., Chiba, Japan)

To realize a quantitative diagnosis method of liver fibrosis, we have been developing a modeling method for the probability density function of the echo amplitude. We developed a scatterer distribution model of diseased livers considering the liver lobule structure, because computer simulations are effective for obtaining information on the continuous stage of diseases. We examined relationship between tissue characterization and ultrasonic B-mode image which is simulated by this scatterer distribution model. From these analyses, we proposed a multi-Rayleigh model with three Rayleigh distributions, corresponding to the distribution of the echo amplitude from hypoechoic, normal, and fibrous tissues. We showed quantitatively that the proposed model can model the amplitude distribution of liver fibrosis echo data adequately. We also found that fibrous indices can be estimated stably using the proposed model. To evaluate the liver fibrosis more accurately, each pixel's amplitude value in the clinical echo image was transformed into the fibrotic probabilities by using the multi-Rayleigh model. Clinical echo images of liver fibrosis were analyzed, and the relationship between those probabilities and the stage of liver fibrosis were discussed. We conclude that the proposed method is valid for the quantitative diagnosis of liver fibrosis.

11:20

3aBA10. Speed of sound, attenuation, and acoustic impedance of hepatic lobule in diseased rat liver observed by scanning acoustic microscopy with 250 MHz. Kenji Yoshida (Ctr. for Frontier Medical Eng., Chibar Univ., 1-33 Yayoicho, Inageku, Chiba, Chiba 263-8522, Japan, kenyoshi1980@chiba-u.jp), Zhihao Deng, Kazuyo Ito (Graduate School of Eng., Chiba Univ., Chiba, Japan), Jonathan Mamou (Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), Hitoshi Maruyama (Dept. of Gastroenterology and Nephrology, Chiba Univ., Chiba, Japan), Hiroyuki Hachiya (Tokyo Inst. of Technol., Tokyo, Japan), and Tadashi Yamaguchi (Ctr. for Frontier Medical Eng., Chibar Univ., Chiba, Japan)

Based on the analysis of echo amplitude envelope and power spectrum of backscattering echo, our group has attempted the characterization of various types of tissue such as liver, lymph node, and skin. For example, the scatterer size estimation of rat liver showed that the estimated size was 30-50 μm. However, it was not investigated how the obtained parameters related to the histology in detail. To resolve this problem, it would be invaluable to understand the spatial distribution of acoustic properties, e.g., speed of sound, attenuation, and acoustic impedance in the scale from micro- to millimeter range. Thus, we measure and accumulate the data using ultrasound with center frequency of 1- 500 MHz. This report introduces the acoustic properties of specimen prepared from two different types of rats: normal liver and liver fibrosis models. A scanning acoustic microscopy using 250-MHz ultrasound with spatial resolution of 7 μm was employed. After the measurement, H&E staining and Masson trichrome staining were conducted to obtain histology. Image registration of SoS image and histology image enabled to separately evaluate SoS of cell nucleus, cytoplasm, and fibrotic tissue. The result demonstrated that there was no significant difference in SoS among cell nucleus, cytoplasm, and fibrotic tissue.

11:40

3aBA11. Cultured tumor model inspection for anticancer drugs using acoustic impedance microscope. Sachiko Yoshida, Rahma H. Rahayu, Kyoichi Takanashi, Kenta Kishikawa, Hirofumi Kurita, Kazunori Takashima, Naohiro Hozumi (Toyohashi Univ. of Technol., 1-1, Hibarigaoka, Tempaku-cho, Toyohashi 4418580, Japan, syoshida@ens.tut.ac.jp), Kazuto Kobayashi (Honda Electronics Co., Ltd., Toyohashi, Japan), and Seiji Yamamoto (Hamamatsu Univ. School of Medicine, Hamamatsu, Japan)

High-frequency ultrasonic microscope is useful to observe intracellular structure of cultured living cells. We have observed the dynamical change of intracellular actin filaments in several cells, and reported the difference of stabilities between cancer cells and ordinary cells. The dynamics of actin filaments are deeply related to cancer cell viability. In this study, we applied this acoustic microscopy to antitumor drug screening. One of the cultured brain cancer models was established with both normal glial cells and C6 glioma cells. Normal glial cells were labeled by endogenous fluorescent protein to be identified. Both cells were co-cultured on a 75 micrometer-thick polystyrene substrate. Acoustic pulse wave with 320 MHz was focused on the interface between the cell and the substrate, and sent through the substrate. Either classical anticancer drug, cytochalasin B, or novel clinical anticancer drug, temozolomide, was applied to the cultured cancer model, and observed their intracellular acoustic changes using the acoustic microscope. Before and after drug treatment, the cultures were observed with the laser confocal microscopy to confirm survival of each cell. Another of the models was established with fluorescent protein-labeled breast cancer cells. Our model system would be useful to inspect anticancer drugs without whole animals.

WEDNESDAY MORNING, 30 NOVEMBER 2016 SOUTH PACIFIC 4, 8:15 A.M. TO 11:45 A.M.

Session 3aEA

Engineering Acoustics: Acoustic Analysis

Rumi Ito, Cochair
Ritsumeikan University, 1-1-1, Noji-Higashi, Kusatsu 525-8577, Japan

Daniel J. Mennitt, Cochair
Electrical and Computer Engineering, Colorado State University, 1201 Oakridge Drive, Suite 100, Fort Collins, CO 80525

Contributed Papers

8:15

3aEA1. An evaluation on voice intelligibility for factory noise reduction based on integration of active noise control and auditory masking. Rumi Ito, Takahiro Fukumori, Masato Nakayama, and Takanobu Nishiura (Ritsumeikan Univ., 1-1-1, Noji-Higashi, Kusatsu, Shiga 525-8577, Japan, is0100is@ed.ritsumei.ac.jp)

Factory noise has been focused upon as one of the social problems. Factory workers exposed to loud, continual noises feel strong discomfort by hearing the loud factory noises for a long time. We have previously proposed an integration method of the active noise control (ANC) and the auditory masking for a noise reduction. The ANC reduces the sound pressure level of the noise in a lower-frequency band by emitting the secondary signals, and the auditory masking reduces discomfort which caused the spectral peaks in a higher-frequency band by emitting the secondary signals. We confirmed that the proposed method is effective for reducing discomfort. However, the method might disturb conversations among the workers by emitting secondary signals. In this study, we therefore evaluated the voice intelligibility of the noise-reduced speech obtained by applying the proposed method to the noisy speech recorded in the factory. As a result of the evaluation experiment, we confirmed that the proposed method can keep voice intelligibility almost the same as the intelligibility of the clean speech in the amid factory noise. In addition, we investigated the relationship between the voice intelligibility and each method under the condition that the controlling frequency-bands are set at to 0–500 Hz for ANC and 500–5500 Hz for auditory masking.

8:30

3aEA2. Head-related transfer function personalization based on spectral notch estimation with stereo images of pinna. Zhuan Zuo, Taku Yoshimura, Takahiro Fukumori, Masato Nakayama, and Takanobu Nishiura (Ritsumeikan Univ., 1-1-1, Noji-higashi, Kusatsu 525-8577, Japan, is0291hr@ed.ritsumei.ac.jp)

Three-dimensional (3-D) sound field reproduction systems such as surround sound reproduction systems are rapidly being developed in recent few years. However, these systems need a lot of space for the placement of loudspeakers. For this reason, a binaural system has been studied as one of the systems, which controls the sound image localization with the head-related transfer functions (HRTFs) and just by using only a headphones. However, the individual differences in body features would cause the variations of HRTFs which greatly affect the direction perception. For this reason, a method based on the spectral notch estimation from the pinna shape has been studied for the personalization of HRTFs. In this study, we propose a new method, in which the depth information used for the spectral notch estimation is measured with stereo images of the pinna. Two cameras are utilized together as a stereo camera to take the disparate stereo images so that we can use the disparity of them to measure the depth information of the pinna. As a result of an evaluation experiment, we could reduce the cost of actual measurement and the burden to the subjects by the proposed method.

8:45

3aEA3. Arrangement of control points for sound spot generation in the multi-point control method. Kazuya Yasueda (Graduate School of Sci. and Technol., Ryukoku Univ., 1-5 Yokotani, Setaoe-cho, Otsu-shi, Shiga 520-2123, Japan, t16d001@mail.ryukoku.ac.jp), Daisuke Shinjo, and Akitoshi Kataoka (Faculty of Sci. and Technol., Ryukoku Univ., Otsu-shi, Japan)

In this research, we considered control point arrangement by using small-scale linear loudspeaker array. The multi-point control method is proposed for sound reproduction in a limited area/direction. This method is based on boundary surface control principle. The desired signal is reproduced in the target point by the sound pressure control in the control points. According to the MINT (multiple-input/output theorem), the signal is reproduced with accuracy in the target point if the number of sound sources more than the number of control points exists. However, a lot of speakers make it difficult to use in real environment. Therefore, small-scale loudspeaker array is preferable. If the number of control points is greater than the number of sound sources, the filter coefficient is approximated by the least squares-method. This approximation affects the sound pressure suppression and sound quality depending on the arrangement of control point. We examined the number of control points arranged. The input signal is the speech signal that has 300-3400 Hz frequency band. We evaluated some arrangement pattern by sound pressure, SD (Spectral Distortion), and PESQ (Perceptual Evaluation of Speech Quality). As a result, good results were obtained than conventional arrangement.

9:00

3aEA4. Relation between area of shielding bone-conduction microphone and quality of speech communication in noisy environment. Kojiro Takahashi, Kenji Muto (Graduate School of Eng. and Sci., Shibaura Inst. of Technol., 3-7-5 Toyosu, Muto lab, Koto-ku, Tokyo 135-8548, Japan, ma15051@shibaura-it.ac.jp), and Kazuo Yagi (Tokyo Metropolitan Univ., Arakawa-ku, Tokyo, Japan)

The doctor does not receive the transmitted voice of the patient when the patient becomes unwell in MRI examination, because the voice is faint and MRI acoustical noise is loud. Our purpose is to improve the quality of speech communication of bone-conduction microphone to transmit the patient's faint voice to the doctor. We propose the theoretical formula using the characteristic of vibration propagation for calculating the quality in the case of the shielding microphone. In this report, the improvement level of speech communication quality in the case of the shielding microphone is estimated by the proposed formula in order to discuss the accuracy of the proposed formula. Here, the improvement level of shielding was calculated with a phantom. The phantom consists of a ceramic plate and a silicon rubber representing bones and muscles, respectively. The vibration of vertical direction on the phantom was measured with accelerometers to calculate the improvement level. The results showed that the quality of speech communication is improved by the shielding area, such as an earmuff, to the level of 3 dB.

9:15

3aEA5. Metasurfaces for perfect transmission of sound from air into water. Eun Bok, Haejin Choi, Jong J. Park, and Sam H. Lee (Phys., College of Sci. 332, 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, skzz430@gmail.com)

Three orders of magnitude difference in acoustic impedance between air and water has so far prevented effective sound transmission through the interface. A conventional method for an antireflection coating is a quater-wave plate. Such a coating would be too thick and bulky because acoustic waves in audible range are long. More importantly, one cannot find a suitable coating material among the naturally occurring materials, the acoustic impedance of which is required to be a geometrical mean of those of air and water. Here, we present fabrication of acoustic metasurfaces of thickness much smaller than the wavelength, which allows perfect transmission of sound through the air/water interface. The metasurface is a two dimensional array of unit cells which are consisting of membrane and cavities. Our work demonstrates usage of metamaterials for impedance matching in extreme conditions.

9:30

3aEA6. Gain and directivity of exponential horn receivers. Daniel J. Mennitt (Elec. and Comput. Eng., Colorado State Univ., 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, daniel_mennitt@partner.nps.gov) and Kurt Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

The self-noise of acoustical sensors limits their capacity to monitor extremely quiet environments and measure the subtle, adventitious cues that animals routinely rely upon. Although primarily used in sound production, horns also can amplify sound prior to transduction by a microphone. Given the small size of microelectromechanical microphones, substantial gain can be achieved at wavelengths larger than the horn. An analytical model of an exponential horn embedded in a rigid spherical housing was formulated to describe the gain relative to a free-field receiver as a function of frequency and angle of arrival. Through comparison with experiment and numerical models, the directivity of the horn receiver is shown to be adequately described by the analytical model up to a critical wavelength, beyond which physical assumptions are violated to some degree. Numerical models, based on the equivalent source method, describe the acoustic scattering within and around the horn and provide a means for identifying the mechanisms of deviation in gain at high frequencies and large angles of arrival.

9:45

3aEA7. Large-scale loudspeaker array system for sound field reproduction using high-speed 1 bit signal processing. Yusei Yamanaka, Daiki Takeuchi, Dai Kuze, Yusuke Ikeda, and Yasuhiro Oikawa (Waseda Univ., 407-2 59-gokan, Okubo Shunjuku-ku, Tokyo 169-8555, Japan, sunset@akane.waseda.jp)

Recently, many techniques of physical sound field reproduction, such as Wave Field Synthesis, Higher Order Ambisonics and Boundary Surface Control, have been studied. Since the number of loudspeakers generally increases to control with higher spatial resolution and higher frequency, the system becomes too complicated to be constructed. On the other hand, high-speed 1 bit signal processing has been studied widely. The 1 bit audio system has interesting features such as simple circuit implementation, simple signal transmission, and no D/A converter. In this study, we proposed the large-scale loudspeaker array system with high-speed 1 bit signal. The system consists of a master and hub systems with sample rate of 4 MHz. The master system reads multi-channel 1 bit signal from SSD and transmits it to each hub system. Each hub system drives 32 loudspeakers by CMOS drivers. The master and each hub system are connected with a LAN cable to implement only hub system near the loudspeakers. It is easy to scale the size of system with flexibility thanks to the simplicity of the circuit and signal flow. We conducted preliminary experiments to control its sound directivity. In the future, we are planning to construct 1536-channel sound reproduction system with our proposed system.

10:00–10:15 Break

10:15

3aEA8. Evaluation on overlapping barriers design using SoundPLAN. Sheying Sun, Neil Morozumi, and Richard Patching (Patching Assoc. Acoust. Eng. Ltd., #20, 2150 - 29th St. NE, Calgary, AB T1Y 7G4, Canada, ssun@patchingassociates.com)

Traffic noise comes along with road developments. To overcome traffic noise problems, noise barriers, low noise pavements, low noise vehicles, traffic control measures, and proper land uses have been proposed. Being considered to have great benefits of easy installation, better noise reduction performance, and ability to soothe annoyed residents, noise barriers have become the most prevalent noise control measures adopted by most agencies. However, any defects in noise barriers may allow unnecessary noise propagation and thus degrade their performance. Consequently, noise barriers should be constructed and maintained with care to uphold their designed noise reduction capability. Overlapping barriers are sometimes necessary for maintenance or community access. There is apparently a need to evaluate the effect of the performance of overlapping barriers. Inappropriate design may cause severity of degradation of acoustical effectiveness. Acquiring the appropriate parameters of the overlapping barrier design will

serve as a reference for decision makers to properly allocate the gap location and select the sound design. This analysis focuses on the evaluation of barrier overlapping design and receiver regions in the vicinity of an overlap gap using the SoundPLAN software package. The contributions from the various parameters such as materials, barrier height, gap, and overlapping sizes were investigated.

10:30

3aEA9. The development of wavenumber extended multi-dimensional interpolation techniques for aeroacoustic problems. Seokjong Jang (Seoul National Univ., Daehak-dong, Gwanak-gu, Seoul 151-742, South Korea, clickcoco@snu.ac.kr)

One of the main concerns of computational aeroacoustics is how to solve aeroacoustic problems with complex geometry. The dissipation/dispersion errors should be minimized to obtain accurate acoustic solutions because numerical solutions for aeroacoustic problems are over-susceptible. The wavenumber extended multi-dimensional interpolation techniques were developed in order to satisfy those requirements. The size of a stencil used for standard multi-dimensional interpolation can be determined. Adding the number of stencils with optimized interpolation coefficients enables obtaining the distribution function optimizing the relationship between dispersion and dissipation. Through the optimization process in frequency domains, this technique minimizes the dispersion/dissipation errors generated by the data transfer between multi-grid systems. In addition, a proper limiting process was proposed to remove a numerical instability and oscillations in a discontinuous region. Several problems were numerically simulated by applying the wavenumber extended multi-dimensional interpolation. The results of a advanced technique was validated by being compared with the existing conventional interpolation techniques.

10:45

3aEA10. Numerical investigation on radiation characteristics of aero-intake noise with ground effects. HanAhChim Choung (Mech. and Aerosp. Eng., Seoul National Univ., Daehak-dong, Seoul, Gwanak 151-742, South Korea, good8160@snu.ac.kr)

A hybrid method for the prediction of far-field Aero-intake noise from gas turbine engine with ground effects has been developed. The near-field predictions were obtained by solving the linearized Euler equations with computational aeroacoustic techniques composed of high-order finite difference scheme, non-reflecting boundary condition, overset grids, and parallel computational methods. Kirchhoff integral method was used for the prediction of far-field directivity patterns. In order to predict the noise propagation, ray theory is applied with Geographical Information System (GIS) terrain profile data. Then, the noise level on the ground has been predicted and analyzed, including the effects of air absorption, ground reflection, and diffraction.

11:00

3aEA11. Determination of the acoustical properties of sisal and coconut fiber samples based on the flow resistivity experimental assessment. Key F. Lima, Nilson Barbieri (Mech. Eng., PUCPR, Imaculada Conceição 1155, Curitiba, Paraná 80215901, Brazil, keyflima@gmail.com), Renato Barbieri (Mech. Eng., UDESC, Joinville, Santa Catarina, Brazil), and Vinicius A. Grossl (Mech. Eng., PUCPR, Curitiba, Paraná, Brazil)

Absorbent materials are fibrous or porous and must have the property of being good acoustic dissipaters. The noise reduction process occurs due to transformation of a part of the sound energy into heat. The attenuation is caused by friction of the viscous air fluid inside the irregular arrangement of fibers during the sound wave propagation. This study assesses the complex wave number and the complex characteristic impedance of two natural fibers with the classical empirical equations of Delany and Bazley (1970). These properties are determined based on experimental evaluation of the flow resistivity of sisal and coconut coir fiber samples. The experimental assessment procedure was performed according to the guidelines of ASTM C522-03 (2009). Samples evaluated have approximately identical densities, different diameters, and thicknesses. The diameters were 62.0, 71.5, and 80.0 mm and thicknesses were 5, 10, 15, and 20 mm. The purpose of testing seven dimensions is to verify the effect of these differences on the flow resistivity experimental measurements. This evaluation is due the standard ASTM does not provide guidelines for the sample dimensions.

11:15

3aEA12. Multi-physics modeling of conformal, solid-state, and thin-film thermophones. Mahsa Asgarisabet and Andrew R. Barnard (Mech. Eng. - Eng. Mech., Michigan Technolog. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, arbarnar@mtu.edu)

Thin-film thermophones are thermoacoustic loudspeakers made from new materials such as aligned carbon nanotubes or graphene. They are solid state devices, in that there are no moving parts associated with the sound generation mechanism. Lumped parameter models have been developed for planar thermophones and agree well with experimental results. One benefit of thin-film thermophones is that they can be designed such that the transducer can be conformal to complex geometries. However, the lumped parameter models are not appropriate for complex geometries or electrical/ thermal transients. An electrical-thermal-acoustic multiphysics model of these transducers has been created using COMSOL Multiphysics to address the issues present with the lumped parameter models. Development and validation of the model will be discussed.

11:30

3aEA13. Study on coupled analysis for compact acoustic systems using finite-difference time-domain method based on Kirchhoff-Love plate theory and elastic wave. Hayato Takimoto, Yoshinobu Kajikawa (Faculty of Eng. Sci., Kansai Univ., 3-3-35, Yamate-cho, Suita, Osaka 564-8680, Japan, k730406@kansai-u.ac.jp), Masahiro Toyoda (Faculty of Environ. and Urban Eng., Kansai Univ., Suita, Osaka, Japan), and Takashi Miyakura (Hosiden Corp., Yao, Osaka, Japan)

In this paper, we propose a coupled analysis method for compact acoustic systems (e.g., micro speakers, electret condenser microphones) using finite-difference time-domain (FDTD) method based on the Kirchhoff-Love plate theory and elastic wave. Acoustic systems are usually analyzed through any numerical analysis technique based on wave propagation. However, an exact frequency response is rarely obtained for compact acoustic systems. This is because the standard FDTD method cannot treat the change of the frequency response due to the air viscosity and cannot also treat the wave propagation in a solid. Thus, we cannot obtain exact frequency responses for compact acoustic systems through the analysis of coupled fluid-solid model based on standard FDTD method. In order to solve these problems, we utilize a coupled analysis method based on Kirchhoff-Love plate theory and elastic wave FDTD method considering air viscosity. The analysis method models diaphragm and voice coil as a two-dimensional plate model based on Kirchhoff-Love plate theory and air as a three-dimensional model based on elastic wave FDTD method considering the air viscosity. In this paper, we demonstrate the effectiveness of the analysis method through some comparisons between calculated and measured frequency responses.

Session 3aMU

Musical Acoustics: East and Western Instruments: Contrasts and Connections

James P. Cottingham, Cochair
Physics, Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52402

Toru Kamekawa, Cochair
Musical Creativity and the Environment, Tokyo University of the Arts, 1-25-1, Senju, Adachi-ku, Tokyo 120-0034, Japan

Chair's Introduction—8:35

Invited Papers

8:40

3aMU1. Eastern and Western free reed instruments: Acoustics, history, and culture. James P. Cottingham (Phys., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

There are two families of free reed instruments. Mouth-blown instruments employing free reeds coupled to pipe resonators have been used throughout East and Southeast Asia since ancient times. The *sheng*, *sho*, *khaen, and bawu* are typical examples. Details of the origin and development of these instruments are not known, but are closely connected with the history and prehistory of a multitude of ethnic groups. The instruments of the Western free reed family are of modern origin, originating in Europe around 200 years ago. These include the reed organ, harmonium, harmonica, and the accordion-concertina family. This paper explores significant differences in the acoustical design of the two groups of instruments and surveys historical developments of each, including examples of possible connections and influences between the two families. Interesting developments involving the two families of free reed instruments occurred as elements of their original musical cultures became mixed in the nineteenth and twentieth centuries.

9:00

3aMU2. Western and non-western approaches to control timbre using multi-resonator/generator systems of wind instruments. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Wind instruments are often modeled as a coupled generator-resonator system, which for the case of the clarinet consists of a reed and a cylindrical pipe. However, the fact that the throat and oral cavities behind the tone generator also play a critical role is often neglected. While the resonance pipe of a wind instrument can be adjusted in length through keyholes or valves, its width is fixed. The opposite can be observed for the human resonance system where the length is fixed but the dimensions of the cross-section are variable. This important fact is routinely used to shape the timbral qualities of tones. In many non-western music cultures, time-varying timbral modifications are more important than melodic aspects, for example, in traditional digeridoo practice. The effects of vocal tract control including Tongue Controlled Embouchure (TCE) will be analyzed and discussed using extended saxophone techniques with adaptive mouthpieces [Braasch, 2014, J. Acoust. Soc. Am. **135**, 2245].

9:20

3aMU3. Finite-difference model of mode shape changes of the Myanmar *pat wain*drum circle with tuning paste on drum head. Rolf Bader (Univ. of Haburg, Inst. of Systematic Musicology, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

The tuning process of drums differ considerably between Western and Southeast Asian drums. While Western drums, like tom-toms, snare, or bass drums of a rock or jazz drum kit, are tuned by changing the tension of the drum head, many drums in Southeast Asia are tuned by adding a paste consisting of rice and ash. As an example, the tuning of the Myanmar *pat wain*, a unique drum circle of a *hsain wain* orchestra is investigated. The *pat wain* has 20 pitched drums and is played with great virtuosity. It was investigated by the author during a field trip in northern Myanmar during its tuning process. The drum head is modeled using a time-dependent Finite-Difference Method (FDM) to estimate pitch and mode changes due to adding the paste. The results are compared to the field recordings which show considerable damping and pitch glides with low-pitched drums. Also the tuning of the drum circle is compared to different tuning systems showing similarities to a tuning pattern of Cambodian *roneat deik*metalophone.

9:40

3aMU4. Transforming rhythmscapes. Rohan Krishnamurthy (Music, Ohlone College, 544 Sunrise Circle, Kalamazoo, MI 49009, rohan.krishnamurthy@rochester.edu)

Originating in India over two millennia, the mridangam is one of the most ancient, complex, and versatile hand drums in the world. The pitched nature of the two-sided drum and a complex split finger technique enable the creation of over a dozen unique tones. The acoustics of the mridangam, especially the tonal drumhead, will be discussed. Musical excerpts will demonstrate how the timbres and sound structures of the mridangam have intertwined with diverse musical genres from around the world, including global percussion ensembles, symphony orchestras, jazz ensembles, and Hollywood scores. The recent phenomenon of real-time online music education will also be discussed as a seminal mode of disseminating the mridangam tradition across the globe. The presentation will conclude with a live, interactive performance.

10:00–10:15 Break

10:15

3aMU5. String-pressing force of Japanese Koto. Tamaki Ando (Koto Player, 1-3-17 Sakuragaoka, Setagaya-ku 156-0054, Japan, antama@d6.dion.ne.jp), Satoshi Obata, and Eriko Aiba (The Univ. of Electro-Communications, Chofu, Tokyo, Japan)

The koto is a traditional Japanese stringed musical instrument and is similar to the zither. The koto typically has 13 strings and is played by plucking the strings using three fingers (thumb, index, and middle fingers) with picks. Each string is strung over a movable bridge, and the support point can be changed by adjusting the position of the bridge. As the distance between the support point and the right end of the string determines a pitch, the koto is tuned by adjusting the bridge position. In addition, players can change the pitch by pressing the string with their left hand. This technique is called oshide and it is important but difficult to apply accurately for koto players. The subjective difficulties of oshide change depending on the position of the strings (on the near or far side of a player's body), string's material (silk or polyester), condition (old or new) of the string, and so on. In order to ensure good-quality performance, expert koto players modify their string-pressing manner. However, there are no previous studies on this. In this presentation, we present our results of the string-pressing force applied by koto players using oshide under various conditions.

Contributed Papers

10:35

3aMU6. Differences between audio engineers and players in the subjective evaluation on the recording position of the *Koto* (Japanese harp). Shun Saito (Music and PsychoAcoust., Res. Area of Creativity of Music and Sound, Div. of Musicology and Music Studies, Graduate School of Music, Tokyo Univ. of the Arts, Oosumidai1-12-13, Isehara-shi, Kanagawa-ken 259-1105, Japan, s2316907@ms.geidai.ac.jp), Toru Kamekawa, and Atsushi Marui (Music and PsychoAcoust., Res. Area of Creativity of Music and Sound, Div. of Musicology and Music Studies, Graduate School of Music, Tokyo Univ. of the Arts, Tokyo, Japan)

The objective of this study was to examine the appropriate recording position for the *Koto* (Japanese harp). The sound of the *Koto* used as experiment stimuli was recorded at 54 different spherical microphone positions. A group of audio engineers and a group of *Koto* players rated the recorded stimuli on thirteen attribute rating scales. The result of evaluation was performed in an analysis of variance using two factors, namely, the recording position and the evaluator group. In the results, significant differences between the two groups were not found in almost all attributes. However, it should be noted that in the assessment of a few number of attributes, significant differences were observed in interaction among the recording positions and between the groups. For determining suitable factors within the attributes, a factor analysis was performed. The factor scores from the results of the factor analysis showed that the tendencies of each group in regards to the recording positions were different, and results showed that some recording positions had higher scores. It was found that factor one of each group tended to have a higher score in the upward recording position on the players side.

10:50

3aMU7. Modes of vibration of a sheng reed. Thomas W. Kirk (St. Olaf College, St. Olaf College, 1500 St. Olaf Ave., Northfield, MN 55057, kirk@stolaf.edu), James P. Cottingham (Phys., Coe College, Cedar Rapids, IA), and Eric Sindelar (Phys., Coe College, Cedar Rapids, MN)

The sheng is an Asian mouth organ consisting of a number of bamboo pipes enclosed in a wind chamber with brass free reeds at the end of each pipe inside the chamber. The reed in each pipe typically includes a small wax tuning weight near the free end of the reed tongue. The determination of modes of vibration of these reeds, especially how the tuning weight changes those modes of vibration, is the topic of this paper. Recent research has investigated the acoustics of sheng pipes and compared the measured impedance of sheng pipes augmented with pipe resonators with a theoretical model. This research employs finite element analysis software to construct a theoretical model of sheng reeds and compare that model with measured vibrational analysis data gathered with a laser vibrometer. The primary emphasis is on how well the predicted resonance frequencies compare with measured resonance frequencies of the reed as indicated in a response curve. Also compared are the predicted nodal and antinodal positions along the length of the reed with the measured nodal positions. [Research supported by National Science Foundation REU grant PHY-1004860.]

11:05

3aMU8. Organologic and acoustic similarities of the European *violin* and the Chinese *erhu*. Florian Pfeifle (Univ. of Hamburg, Neue Rabenstrasse 13, Hamburg 20354, Germany, Florian.Pfeifle@uni-hamburg.de)

The Chinese *erhu* is a bowed spike lute with a recorded history dating back to the Song dynasty (960-1279). It is played in different sociocultural as well as musical settings and in different styles of Chinese music. In modern Chinese orchestras it fills the part of the lead voice in the bowed instruments section, comparable to the role of the European *violin* in classical orchestras. Even though the design of both instruments differs in several central aspects, they exhibit structural and acoustical similarities. In this work, organologic as well as acoustical similarities of both instruments are highlighted with a special focus on the excitation mechanisms of both instruments. To this end, a set of measurements using high-speed camera recordings of the bowing and the string motion are evaluated and compared. Differences in the resulting "Helmholtz" motion are explained and attributed to the specific string attachment and the structure of the bow respectively. Timbre similarities are shown for several different bowing gestures and the expressive palette of both instruments is compared. Organologic similarities are highlighted through an analysis of the historical evolution of structural components of both instruments again with a strong focus on the bow.

Session 3aNS

Noise, Psychological and Physiological Acoustics and ASA Committee on Standards: Current Issues in Hearing Protection and Hearing Conservation I

Elliott H. Berger, Cochair
Personal Safety Division, 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650

Jinro Inoue, Cochair
Department of Health Policy and Management, University of Occupational and Environmental Health, Japan, 1-1 Iseigaoka, Yahatanishi-ku, Kitakyushu 807-8555, Japan

Chair's Introduction—8:00

Invited Papers

8:05

3aNS1. Review of the 2016 Edition of ANSI S12.6 and its place in the panoply of standards on the measurement of real-ear attenuation at threshold since 1957. Elliott H. Berger (Personal Safety Div., 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650, elliott.berger@mmm.com)

The American standard specifying the procedure for the measurement of real-ear attenuation at threshold (REAT), often termed the gold standard in measuring hearing protector attenuation, was recently approved as an updated version, ANSI 12.6-2016. REAT was first standardized worldwide in the late 1950s in an American standard ANSI Z24.22-1957 and the method has evolved with time. Changes have affected the electroacoustic requirements for the sound field, instrumentation, audiometric method, and permissible background noise, but more importantly have also improved the specification of how the experimenter works with and fits the test subjects. So too, estimates of uncertainty are now included, and in the 2016 version they have been clarified and brought into harmony with ISO 4869-1. The author, who has been the chair since 1985 of the ANSI working group responsible for S12.6, will compare and contrast the various methods and present representative data as well as a discussion of the expanded uncertainties that are specified. Those values, for the 1/3-octave band test bands from 125 Hz to 8000 Hz, vary from approximately 1.5-2 dB for earmuffs and 2-3 dB for earplugs for within-laboratory testing, to 4-6 dB for earmuffs and 6.5-8 dB for earplugs for between laboratory measurements.

8:25

3aNS2. Field fit testing with E-A-Rfit Dual-Ear validation system in China. Yufei Liu (Person Safety Div., 3M, No.9, Nanxiang Er Rd., Sci. City, Guangzhou, Guangdong 510663, China, sliu9@mmm.com), Wei Gong (Jiangsu Provincial Ctr. for Disease Control and Prevention, Nanjing, China), and Min Yang (Guangdong Province Hospital for Occupational Disease Prevention and Treatment, Guangzhou, China)

This presentation introduces hearing protection fit testing on over 1000 workers in 6 different factories, each with different hearing conservation practices in China. The 3M™ E-A-Rfit™ Dual-Ear validation system was used to measure the Personal Attenuation Rating (PAR), on three different types of Hearing Protection Devices (HPDs), including a foam earplug, a premolded earplug, and a headband earmuff at the work site. A follow up visit to repeat fit-testing was conducted approximately 6 months after the initial visit. Field attenuation of HPDs obtained by workers' and the effects of training toward improving the attenuation were observed. In addition, PARs for the Peltor™X4A earmuff users were obtained at several points throughout the work shift to capture different fits. Result shows that there is wide variability in PARs and poor agreement between PAR and labelled noise attenuation ratings obtained from laboratory measurement. Hearing protection fit testing is essential to verify the sufficiency of attenuation. Training together with fit testing was shown to improve the PAR. **Key words:** Hearing Protection Device (HPD), fit testing, Personal Attenuation Rating (PAR), China.

8:45

3aNS3. Personal attenuation ratings reported using Fit Check Solo™: Is background noise a concern? Taichi K. Murata (Oak Ridge Inst. for Sci. and Education, WPAFB, OH), Hilary Gallagher (Battlespace Acoust. Branch, Air Force Res. Lab., 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, hilary.gallagher.1@us.af.mil), Elizabeth A. McKenna (SpecPro Tech. Services, WPAFB, OH), Gregory G. Wolff (Air Force Satellite, Defense Health Agency, WPAFB, OH), Sarah C. Sullivan (Hearing Ctr. of Excellence, Dept. of Defense, WPAFB, OH), Matthew R. Ankrom (Oak Ridge Inst. for Sci. and Education, WPAFB, OH), William J. Murphy (National Inst. of Occupational Safety and Health, Cincinnati, OH), Nourhan K. Abouzahra (Oak Ridge Inst. for Sci. and Education, WPAFB, OH), and Quintin A. Hecht (School of Aerosp. Medicine, U.S. Air Force, WPAFB, OH)

It is well known that some personnel, especially Service Members, operate in hazardous noise-environments. To reduce the risk of hearing loss and hearing related disabilities, hearing protection devices (HPDs) are worn to decrease the level of noise at the ear. Noise attenuation provided by HPDs can vary greatly depending on user fit. Fit test systems were developed to measure the noise attenuation performance of HPDs and to report a personal attenuation rating (PAR) based on that particular fit. Elevated levels of background noise in a typical office-type setting (in comparison to a laboratory) could however affect PAR results. A study completed by the Air Force Research Laboratory compared PAR results of 3M EAR Classic earplugs and Howard Leight Airsoft earplugs from data collected using ANSI S12.6 methodology in the laboratory and data collected using Fit-Check Solo™ in an office environment, a controlled noise office environment, and the laboratory. Results demonstrated that PARs were affected by increased background noise; however, the significance varied based on the number of frequencies used in the PAR calculation. Experimental methodologies, PAR results, and background noise levels will be presented.

9:05

3aNS4. Modeling the effects of hearing protection on speech recognition as function of noise level and hearing status. Christian Giguère (Audiology/SLP Program, Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H8M5, Canada, cgiguere@uottawa.ca) and Elliott H. Berger (Personal Safety Div., 3M, Indianapolis, IN)

The ability to communicate verbally when wearing hearing protectors is an important aspect to consider for compliance in the use of these devices as well as for safety and operational efficiency in the workplace. Speech recognition with hearing protectors is influenced by many factors such as the overall amount of noise reduction, the slope of the attenuation-frequency function and the presence of hearing loss [Giguère and Berger, Int. J. Audiol. **55**, S30-S40 (2016)]. Interestingly, results from subjective and modelling studies sometimes show a benefit of using passive hearing protectors on speech recognition in noise compared to unprotected listening, especially with normal-hearing individuals in high-noise levels. The purpose of this modeling study is to further explore the conditions promoting such a facilitative effect of hearing protectors on speech recognition, with a particular attention to the noise level in the environment and the hearing status of the user. This information is important to better guide the selection of the most appropriate hearing protection device, given the characteristics of the user and workplace noise.

9:25

3aNS5. Evaluation of the improvement in hearing Japanese speech while wearing earplugs advertised as enabling communication. Toshitaka Yokoya, Jinro Inoue, Shinji Kohata, Shoko Kawanami, and Seichi Horie (Health Policy and Management, Industrial Ecological Sci., Iseigaoka 1-1, Yahata-Nishi-Ku, Kitakyushu, Fukuoka 8078555, Japan, yokoya1117@med.uoeh-u.ac.jp)

Workers often refuse to wear earplugs because of the difficulty hearing conversations in noisy workplaces. Little research has been performed to explore whether multiple syllables of the Japanese language are difficult to hear through earplugs. New earplugs recently became available that could be used to communicate clearly in noisy workplaces and simultaneously prevent personal noise exposure. This study aimed to evaluate the hearing of speech using three types of earplugs in a noisy environment. The subjects were 10 people without hearing loss. We conducted this experiment in the anechoic room at the University of Occupational and Environmental Health, Japan. We set up a pink noise environment at 80 dBA, 85 dBA, and 90 dBA, and evaluated the hearing of speech with and without earplugs. The subjects were asked to distinguish standard Japanese words comprising two syllables recorded on a CD-ROM (TY-89) established for the evaluation of the fitness of hearing aids. If the speech level required for 90% intelligibility with those new earplugs was lower than that without earplugs, or lower than that with reference earplugs reported previously, the new earplugs can protect hearing and lead to good communication. A comparison of the new and reference earplugs will be discussed.

9:45–10:00 Break

10:00

3aNS6. Uncertainty of hearing protector impulsive attenuation measurements. Cameron J. Fackler, Elliott H. Berger, and Michael E. Stergar (3M Personal Safety Div., 7911 Zionsville Rd., Indianapolis, IN 46268, elliott.berger@mmm.com)

Measurement of the impulsive protection of a hearing protection device (HPD) is currently standardized in ANSI/ASA S12.42-2010 as a measurement of impulse peak insertion loss (IPIL). IPIL is a time-domain metric, determined as the difference between open-ear and occluded-ear peak sound pressure levels of an impulse, measured without and with an HPD in place on an acoustic test fixture. To characterize potential level-dependent protection of an HPD, the IPIL test is repeated at several free-field impulse peak sound pressure levels, typically ranging from approximately 130-170 dB. Based on our experiences measuring IPIL with both a shock tube and a rifle serving as the impulse source, as well as a review of existing literature reporting IPIL studies, we discuss and attempt to quantify the uncertainty of such measurements. Sources of uncertainty include the repeatability of the impulse source, characteristics of the sound field created by the impulse source, spectral characteristics of both the impulse source and the HPD's attenuation, variability between HPD samples and repeat fitting of the HPD to the acoustic test fixture, and variability in the measurement instrumentation and data analysis procedures. Our estimates will be compared to those provided in the uncertainty annex of S12.42.

10:20

3aNS7. Angle dependent effects for impulse noise reduction for hearing protectors. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov), Daniel Adams (Dept. Communciation Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and Pamela S. Graydon (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH)

The exposure at the ear in response to a forward-propagating wave depends upon the angle of incidence at the head, the nominal sound pressure level of the impulse and the attenuation of hearing protection (if worn). The unoccluded and occluded responses of an acoustic test fixture equipped with two G.R.A.S. IEC 60711 couplers 1/4 inch microphones were measured in 15 degree increments for impulses with nominal peak sound pressure levels of 150 and 160 decibels. The attenuation was assessed in a variety of ways: Impulse Peak Insertion Loss (IPIL), change in A-weighted Equivalent Energy, and change in the Auditory Hazard Unit. Generally, the L_{Aeq} was quite similar to the (IPIL). However, the change in AHUs predicted less attenuation than was actually observed. The lower performance for AHUs may be attributable to the nonlinear hazard growth for the unoccluded ear. These results will be compared with continuous noise assessments of the effect of angle on protection.

10:40

3aNS8. Middle ear muscle contractions from non-acoustic elicitors. Gregory A. Flamme, Stephen M. Tasko, Kristy K. Deiters (Speech Pathol. and Audiol., Western Michigan Univ., 1903 W. Michigan Ave., MS 5355, Kalamazoo, MI 49008, greg.flamme@wmich.edu), William A. Ahroon (US Army Aeromedical Res. Lab., Ft. Rucker, AL), and William J. Murphy (Div. of Appl. Res. and Technol., National Inst. for Occupational Safety and Health, Cincinnati, OH)

High-level sounds can elicit middle ear muscle contractions (MEMC), which are commonly known as acoustic reflexes. Tactile stimulation to the face can also elicit MEMC, and it is plausible that MEMC could co-occur with voluntary eye closure gestures. In this paper, we shall present preliminary MEMC results from human volunteers receiving controlled tactile stimulation (nitrogen gas, 10 kPa, 200 ms duration) to four locations on the face and who close the eye ipsilateral to the MEMC detection probe. The MEMC were detected via changes in total energy reflected in the ear using a filtered (0.2 to 8 kHz) click train. Concomitant muscle activity was measured using electromyography. The morphology and magnitude of the MEMC from these non-acoustic stimuli will be described. If non-acoustic MEMC behaviors do not extinguish and are not highly susceptible to fatigue, they could represent an opportunity to mediate exposure to short-duration noises or help explain between-person differences in noise exposure outcomes.

11:00

3aNS9. Going global with hearing conservation: Regulations and trends. Laurie Wells (Personal Safety Div., 3M Co., 817 W. 4th St., Loveland, CO 80537, Laurie.Wells@mmm.com)

Noise is recognized universally as an occupational hazard; however, there is not a globally accepted regulatory approach toward protecting the noise-exposed workforce. The regulatory differences for hearing conservation around the world make it challenging for multinational companies to set policies for corporate hearing loss prevention programs. This presentation compares and contrasts selected aspects of various hearing conservation regulations to the United States Occupational Safety and Health Administration requirements. The following jurisdictions are included: Australia/New Zealand, Brazil, China, European Union, India, Japan, and Mexico. Details were collected from English translations of a country regulation as well as by consultation with an experienced, in-country resident whenever possible. Information includes noise exposure limits, noise control, hearing protection device use, standards, and attenuation derating schemes, audiometric testing and hearing shift criteria, and employee training requirements. Studying these criteria will also spark discussion as to best practices and emerging trends in hearing conservation.

11:20

3aNS10. Introduction of a hearing conservation program in a Japanese enterprise. Naoko Sasaki, Takayuki Ogasawara (Labor Relations and Corporate Services, Mitsubishi Fuso Truck and Bus Corp., 10, Ohkura-cho, Nakahara-ku, Kawasaki, Kanagawa Pref. 211-8522, Japan, naoko.sasaki@daimler.com), Ikuo Denda (Personal Safety Tech. Dept., 3M Japan Ltd., Sagamihara, Japan), Jinro Inoue, and Seichi Horie (Dept. of Health Policy and Management, Univ. of Occupational and Environ. Health, Japan, Kitakyushu, Fukuoka Pref., Japan)

To prevent occupational hearing loss, the Ordinance of Industrial Safety and Health in Japan requires employers to measure sound levels in eight hazardous areas in the workplace. In addition, the Guidelines for the Prevention of Noise-Induced Impairments require employers to measure the sound levels in noisy workplaces or provide audiometric testing for employees exposed to occupational noise. However, measurement of personal noise exposure is not a legal requirement in Japan. The Japanese Working Environment Measurement Standards define how to measure occupational noise using a sound level meter (SLM). However, it is difficult to evaluate real noise exposure for employees who use a sound-emitting industrial tool. We measured this exposure using a dosimeter at a Japanese manufacturing enterprise producing transport machinery. The sound levels at some workshops that were evaluated as 85 dBA using a SLM were shown to be more than 90 dBA using a dosimeter. Following these results, we are introducing a Hearing Conservation Program at the enterprise. We would like to report how to develop this new noise management system in Japan and manage personal noise exposure.

Contributed Paper

11:40

3aNS11. Thoughts on MIL-STD-1474E noise limits. William A. Ahroon (Auditory Protection and Performance Div., U.S. Army Aeromedical Res. Lab., Bldg. 6901, Farrel Rd., Fort Rucker, AL 36362-0577, william.a.ahroon.civ@mail.mil) and Gregory A. Flamme (Dept. of Speech Pathol. and Audiol., Western Michigan Univ., Kalamazoo, MI)

Department of Defense Design Criteria Standard Noise Limits (MIL-STD-1474E), published in 2015, is a standard developed for the acquisition of military materiel specifies the maximum permissible noise levels produced by military systems and the test requirements for measuring these levels. It is intended for use by equipment designers and manufacturers, covering typical operational conditions and provides noise limits shall not be exceeded if the materiel is to be acceptable to the procuring activity. It is not intended to be used for health hazard assessments for personnel exposed to impulsive noises. Appendix B established the acquisition requirements for impulsive noise limits. Oddly, the standard allows the use of two different methods, although the U.S. Army must use a method that utilizes the Auditory Hazard Assessment Algorithm for Humans. This paper describes some obstacles in using this appendix for acquisition of weapons systems, including the curious use of two completely different methods for measuring impulsive noise limits, the middle-ear assumptions of the required Army method, and the limits of assumed risk of hearing loss from exposure to impulsive noises.

WEDNESDAY MORNING, 30 NOVEMBER 2016 SOUTH PACIFIC 2, 8:50 A.M. TO 12:00 NOON

Session 3aPA

Physical Acoustics and Biomedical Acoustics: Acoustic Micro- and Nanofluidics I

James Friend, Cochair
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Tony Jun Huang, Cochair
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Invited Paper

8:50

3aPA1. Controlling sample and particle behaviour in microfluidic systems using acoustic forces. Adrian Neild, Tuncay Alan, Muhsincan Sesen, David Collins, and Jason Brenker (Dept. Mech. & Aero Eng, Monash Univ., Clayton, VIC 3800, Australia, adrian.neild@monash.edu)

Acoustic forces have been widely used to control microparticles within microfluidic systems. Such acoustofluidic systems have been very successful in tasks such as cell sorting, however, to date efforts have been mostly limited to single phase systems. Just as the contrast in acoustic impedance between a fluid and suspended particle means that acoustic forces can be exerted on the particle, a contrast also exists at the interface between two immiscible fluids. This work explores ways in which such acoustically generated forces can be used in digital (two phase) microfluidic systems. Digital microfluidic systems have gathered significant interest because of the potential of single cell analysis and on-chip chemical reactions. In the paradigm of a lab-on-a-chip, each droplet is analogous to a test tube; however, due to the rapidity of producing large numbers of droplets, reaction based systems can run into difficulties as each test tube contains the same sample. In order to gain control over droplet behavior, acoustically actuated systems have been developed to produce individual droplets, selectively merge droplets, and to steer them at a junction, enabling control of the type of chemical in each droplet in a sequence, chemical reaction initiation and guidance of droplets around a chip.

Contributed Paper

9:10

3aPA2. Negative radiation torque using "acoustical sheets". Farid G. Mitri (Area 52 Technol., Chevron, 5 Bisbee Court, Santa Fe, NM 87508, f.g.mitri@ieee.org)

An interesting effect of negative acoustical radiation torque generation and direction reversal of the spinning of an absorptive cylinder is reported [F.G. Mitri, Wave Motion **66**, 31-44 (2016)]. The emergence of the acoustic spin radiation torque experienced by the circular cylinder in a nonviscous host fluid is a consequence of acoustic attenuation inside its absorptive material. Depending on the direction of the shift from the center of a focused Hermite-Gaussian "acoustical-sheet" beam of first-order, the spin torque is negative or positive, causing the rotation of the cylinder in the polar plane in either the clockwise or the anticlockwise direction, respectively. Notice

that at the center of the beam, the torque vanishes as required by symmetry. In the off-axial configuration, the spin torque sign reversal is dependent on the variation of absorption of both the longitudinal and shear waves inside the viscoelastic material. The effect of changing the non-dimensional size parameter *ka* (where *k* is the wavenumber and *a* is the cylinder radius) has a substantial effect on the negative spin torque generation. Potential applications are in acousto-fluidics, particle manipulation, and handling and imaging microscopy to name a few.

Invited Paper

9:25

3aPA3. Ultrasonic tissue micro-engineering: Making tumor models for optimizing immuno-therapy. Martin Wiklund, Karl Olofsson (Appl. Phys., Royal Inst. of Technol., KTH-Albanova, Stockholm 10691, Sweden, martin@biox.kth.se), Valentina Carannante (Microbiology, Tumor and Cell Biology, Karolinska Inst., Stockholm, Sweden), and Björn Önfelt (Appl. Phys., Royal Inst. of Technol., Stockholm, Sweden)

We demonstrate a method for tissue micro-engineering based on ultrasound-supported three-dimensional (3D) cell culture in a multi-well microplate. The method can be used for producing various tissue-mimicking 3D structures in parallel, and utilizes ultrasonic standing wave trapping forces inside hundred micro-wells in a multi-well microplate fabricated in silicon and glass. We have analyzed and optimized the driving parameters of the ultrasound transducer attached to the microplate, and we demonstrate the production of different micro-engineered tumor models, including micro-scaled 3D solid models of liver, kidney, and skin tumors. The tumor models have been analyzed with flow cytometry in order to investigate the difference between 2D and 3D cultures in expression of protein ligands susceptible to recognition by natural killer (NK) immune cells. We have also designed an image-analysis-based method for quantification of the amount of NK cells needed to defeat a tumor having a certain size, shape, and composition. Our results are important for optimizing future personalized immuno-therapy methods for the treatment of various cancer diseases.

Contributed Paper

9:45

3aPA4. Neutral core-shell particles under acoustic radiation force. Jose P. Leao (Phys., Federal Univ. of Alagoas, Av. Lourival Melo Mota, sn, Maceio, Alagoas 57035-557, Brazil), Jose H. Lopes (Exact Sci., Federal Univ. of Alagoas, Maceio, Alagoas, Brazil), and Glauber T. Silva (Phys., Federal Univ. of Alagoas, Maceio, Alagoas, Brazil, glauber@pq.cnpq.br)

We theoretically analyze how a core-shell small particle can become nonresponsive under the acoustic radiation force in an ideal fluid. The particle is composed by a fluid core cloaked by an elastic or viscoelastic solid shell with considering its outer radius much smaller than the wavelength, e.g., the so-called Rayleigh scattering limit. By suitably choosing the core volume fraction (the core-to-shell volume ratio) one may suppress the radiation force. In case of a traveling plane wave, radiation force neutrality is related to scattering cancellation by the cloaking shell. However, neutrality is not achieved for absorptive cloaks. The somehow "intuitive" connection between scattering cancellation and radiation force suppression does not happen for a standing plane wave. Also, shell absorption does not change neutrality in this case. It is worth to mention that standing wave is *the leit motiv* of particle handling in different microfluidics applications. Several examples of radiation force neutrality are illustrated on a fluid core (air, fluorinert, and methanol) with a gold and polyethylene shell. In addition, we show that secondary radiation forces (Bjerknes force) in a many-particle system can also be suppressed by selecting an appropriate core volume fraction.

Invited Papers

10:00

3aPA5. Acoustomicrofluidics for micro-object Manipulation. Ghulam Destgeer and Hyung Jin Sung (Dept. of Mech. Eng., KAIST, Rm. 4120, N9, 291 Daehak-ro, Yuseong-gu, Daejeon, Daejeon 34141, Korea (the Republic of), destgeer@kaist.ac.kr)

Interaction of acoustic waves with fluids at a microscale have shown great potential in realizing efficient actuation of liquids for atomization/nebulization, mixing, pumping, gradient generation, etc. It has also enabled researchers working in various fields to dexterously manipulate suspended micro-objects—microparticles, cells, microorganisms—for concentration, separation, sorting, medium exchange, etc. Surface acoustic waves (SAWs)—travelling as well as standing—have been used to demonstrate several of the above mentioned applications with great success. In particular, standing SAWs along with other bulk acoustic waves-based techniques have been proven to efficiently and continuously separate micro-objects inside a microfluidic channel. Our focus has so far been to explore the unexplored use of travelling SAWs to unravel some of the associated advantages. We have used high frequency travelling SAWs—O(100 MHz)—to separate microparticles flowing through a microchannel or suspended within a sessile droplet. Contrary to the formation of pressure nodes in standing SAWs for micro-object manipulation, traveling SAWs generate a strong acoustic streaming flow and impart a direct acoustic radiation force on the suspended particles for efficient manipulation capabilities with all the additional advantages of their non-invasive nature, label-free manipulation, biocompatibility, low costs, and simplicity in operation.

10:20–10:35 Break

Contributed Paper

10:35

3aPA6. Temperature stable liquid gigahertz viscosity sensors by combining shear mode piezoelectric ScAlN thin film and AT-cut quartz crystal plate. Ko-hei Sano (Waseda Univ., 3-4-1 Ookubo, Shinjuku-ku, 63-06-08B, Tokyo 169-8555, Japan, k-sano@fuji.waseda.jp), Takeshi Mori (Nagoya Inst. of Technol., Nagoya, Japan), Rei Karasawa, and Takahiko Yanagitani (Waseda Univ., Tokyo, Japan)

We here propose the temperature stable liquid GHz viscosity sensors by combining shear mode piezoelectric thin film and AT-cut quartz crystal plate. In the ultrasonic viscosity sensors, longitudinal wave cannot be used because the longitudinal wave energy leaked into the liquid. Therefore, shear wave is required for the viscosity measurement. We recently reported new large shear wave piezoelectricity in the c-axis tilted ScAlN films. In this study, the resonator type viscosity sensor consisting of c-axis tilted ScAlN films (approximately 7 μm) grown on an AT-cut quartz crystal plate were fabricated. Because the thickness of AT-cut quartz crystal is large enough compared to that of ScAlN film, temperature coefficient of resonant frequency is approximately determined by the extremely temperature stable quartz crystal. This enable robust measurement for the external temperature change. Increase of the viscous penetration depth between the liquid and the resonator interface induces the decrease of resonant frequency. The change of liquid viscosity can be then determined by this frequency shift. We observed strong shear wave excitation at the 403 MHz by using a network analyzer in the air. When the sensor was immersed in the pure water, 50 kHz decrease of shear wave resonant frequency was clearly observed.

Invited Paper

10:50

3aPA7. Pattern deposition on an ultrasonic actuator. Ofer Manor (Chemical Eng., Technion - Israel Inst. of Technol., Technion, Haifa 3200003, Israel, manoro@technion.ac.il)

In recent years, one has observed an increasing interest in the pattern deposition of colloidal particles from an evaporating carrier liquid, a process associated with the coffee ring effect. In particular, various companies employ pattern deposition for the fabrication of medium resolution conducting circuits for various applications from touch screens to antennae. In this talk, I will discuss the pattern deposition of solute particles off an evaporating solution film atop an ultrasonic actuator. We use the actuator to generate a MHz vibration in the form of a Rayleigh surface acoustic wave (SAW), propagating in the solid substrate and under the solution film at a phase velocity of 4000 m/s. Transfer of momentum from the SAW to the neighboring solution translates to an acoustic stress, supporting various flow mechanisms, i.e., acoustic streaming, and altering the geometry of the evaporating film. The interplay between the capillary, acoustic, and evaporative stresses within the solution determines the transport dynamics of the solute particles in the solution and the state of pattern deposition on the solid substrate. Using different SAW power levels we alter the dynamic state of the deposition, changing the qualitative geometry of the deposited patterns from dots to stripes and from stripes to solid films.

Contributed Paper

11:10

3aPA8. Manipulation of microbubbles and targeted single cell sonoporation with by surface acoustic waves in a microfluidic device. Long Meng, Feiyan Cai, Juanjuan Chen, Lili Niu, Hairong Zheng (Paul C. Lauterbur Res. Ctr. for Biomedical Imaging, Inst. of Biomedical and Health Eng., Shenzhen Institutes of Adv. Technol., Chinese Acad. of Sci., ShenZhen, China), and Junru Wu (Phys., Univ. of Vermont, Cook Bldg., Burlington, VT 05405, jwu@uvm.edu)

A microfluidic device was developed to transport the aggregated microbubbles to a targeted cell at arbitrary location in liquid solution by introducing the phase-shift to a planar standing surface acoustic wave (SAW). The device consists of two perpendicular pairs of interdigital transducer (IDTs) and a polydimethyl-siloxane microchannel. By adjusting the relative phase between electric voltages applied to the independent IDTs, the microbubbles can be trapped, continuously moved or rotated due to the translation of pressure nodes. We further demonstrate that a SAW is capable of inducing microbubble cluster destruction at a desired location to achieve a single cell's reparable sonoporation. By controlling the position of the microbubble cluster relative to the targeted cell precisely, the effective size of the collapsing microbubbles is measured to be less than 0.68 times the diameter of microbubble cluster. The sonoporation efficiency and the cell viability are 82.4%$\pm$ 6.5% and 90%$\pm$8.7%, respectively, when the targeted cell is within the effective microbubble destruction region.

Invited Paper

11:25

3aPA9. An emulsification system using a microchannel and a piezoelectric transducer. Takefumi Kanda (Graduate School of Natural Sci. and Technol., Okayama Univ., 3-1-1 Tsushima-naka, Kita-ku, Okayama 700-8530, Japan, kanda@sys.okayama-u.ac.jp)

Ultrasonic emulsification devices have been used in many industrial and scientific fields. Recently, a nanoemulsion which contains small droplets having a diameter of sub-micron scale have attracted attentions. In usual, those emulsions are produced by using ultrasonic transducers oscillated to generate a cavitation field. However, such process has a difficulty to avoid contaminations. For a continuous process, we have generated the nanoemulsion by using microchannel devices and a piezoelectric transducer. The devices are made of stainless steel mainly. The micron-size droplets generated by a Y-type or T-type microchannel device, and are oscillated by a piezoelectric transducer in another microchannel device. The driving frequency of the transducer is over 2.2 MHz. We have succeeded in generating sub-micron size droplets which contain anti-cancer drug using this emulsification system. Additionally, we have evaluated the acoustic field in the microchannel by using a micro cavitation sensor. The micro sensor made of a piezoelectric polymer has been installed in the device. The sensor consists of some films deposited and patterned with a photolithography process. Using this sensor, the acoustic state in the microchannel has been evaluated from the acoustic power spectrum. From the results, we can conclude that the cavitation effect is not dominant in the emulsification process by this system.

Contributed Paper

11:45

3aPA10. Dynamic measurement of microbubble compressibility and interactions in an acoustofluidic device. Gianluca Memoli (Dept. of Eng., Computing and Design, School of Informatics, Eng. and Design, Univ. of Sussex, Brighton BN1 9RH, United Kingdom, g.memoli@sussex.ac.uk), Kate O. Baxter, Christopher R. Fury (Acoust. and Ionising Radiations, National Physical Lab., Teddington, United Kingdom), Philip H. Jones (Phys. and Astronomy, Univ. College London, London, United Kingdom), and Bajram Zeqiri (Acoust. and Ionising Radiations, National Physical Lab., Teddington, United Kingdom)

A key parameter for coated microbubbles in diagnostic and therapeutic applications is the non-linear response of their shell to acoustic pressures. Different measurements of this parameter exist, but most rely on the bubbles to be stuck on a surface or on the absence of acoustic excitation. In this work, we use the dynamic of coated microbubbles in an acoustofluidic device to measure the acoustic forces acting on them and the dynamic response of their shell to an increasing acoustic pressure. The device comprises of a microfluidic glass chip, where acoustical tweezers (operating in the range of 160-180 kHz) and optical tweezers can be used simultaneously. Comparing results from the calibrated optical tweezers, laser vibrometry and particle tracking allows a precise characterization of the acoustic field as a function of the driving voltage, at fixed frequencies and for pressures up to 4 kPa. Building on this result we extend the tracking technique to polymer-coated microbubbles and measure the acoustic forces acting on them over a larger range of pressures and frequencies (acoustic force spectroscopy). As pressure increases, we quantify secondary Bjerknes forces and present a dynamic characterization of microbubble compressibility with acoustic pressure. Finally, we present preliminary measurements on lipid-coated microbubbles.

WEDNESDAY MORNING, 30 NOVEMBER 2016 CORAL 5, 7:55 A.M. TO 11:50 A.M.

Session 3aPPa

Psychological and Physiological Acoustics and Speech Communication: Perspectives of Research in Overlooked Hearing Problems

Shigeto Furukawa, Cochair
NTT Communication Science Labs, 3-1 Morinosato-Wakamiya, Atsugi 243-0198, Japan

Michael G. Heinz, Cochair
Speech, Language, and Hearing Sciences & Biomedical Engineering, Purdue University, 500 Oval Drive, West Lafayette, IN 47907

Chair's Introduction—7:55

Invited Papers

8:00

3aPPa1. Predicting effects of hidden hearing loss using signal detection theory. Andrew J. Oxenham and Magdalena Wojtczak (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Recent physiological studies in animals have shown that noise-induced "temporary threshold shift" can result in a large and permanent loss of synaptic connections between the inner hair cells and the auditory nerve, without any measurable change in absolute thresholds. As audiometric absolute thresholds are the current clinical standard for diagnosing human hearing loss, this form of synaptopathy may remain undiagnosed, or hidden, with its perceptual consequences remaining unclear. Baseline expectations for perceptual consequences can be derived from a simple application of the principles of signal detection theory using a set of simplifying assumptions, such as independent and equal information from each auditory nerve fiber, and optimal integration of information. This approach shows that quite dramatic losses (e.g., 50%) lead to relatively small changes in predicted behavioral thresholds (1.5 dB in this case), which may not be easily detectable. However, synaptopathy that selectively affects specific neural populations, such as low-spontaneous-rate fibers, may result in larger and more measurable perceptual effects. Preliminary data from our lab also suggest that physiological measures, such as changes in wideband acoustic reflectance via the middle-ear muscle reflex, may provide a sensitive measure of synaptopathy in humans. [Work supported by NIH grant R01DC005216.]

8:20

3aPPa2. Isolating cochlear synaptopathy in people with impaired audiograms: An auditory brainstem and envelope-following response modeling study. Sarah Verhulst, Viacheslav Vasilkov, Anoop Jagadeesh, and Markus Pelz (Oldenburg Univ., Carl-von-Ossietzky Strasse 9-11, Oldenburg 26129, Germany, sarah.verhulst@uni-oldenburg.de)

Listeners with impaired audiograms likely suffer from mixtures of peripheral hearing pathologies that include outer-hair-cell loss and the more recently discovered cochlear synaptopathy. These mixed pathologies pose challenges for current audiological practice as objective brainstem EEG metrics might not be able to isolate these different hearing deficits. By simulating different degrees of outer-hair-cell and auditory-nerve fiber loss in a functional model of the human auditory periphery, we studied which auditory brainstem (ABR) and envelope-following response (EFR) measures and stimuli are most sensitive to either type of hearing deficit. We validated our model predictions using recordings from 32 listeners with normal or elevated hearing thresholds. The simulations show that high-frequency outer-hair-cell loss steepens the ABR wave-V latency-vs-intensity curve as well as the wave-V amplitude-vs-intensity curve, such that grouping listeners according to the ratio of these metrics (i.e., the ABR growth ratio) offers a way to factor out the outer-hair-cell aspect of hearing loss. Individual differences within an ABR growth ratio group are sensitive to other aspects of peripheral hearing loss such as cochlear synaptopathy. We furthermore simulated EFRs to pure-tone and broadband stimuli of different modulation depths to study which EFR metrics are sensitive to synaptopathy in hearing impaired listeners.

8:40

3aPPa3. Characterizing impairments in compression and filter shape to establish their role in hidden hearing loss. Toshio Irino, Toshie Matsui (Wakayama Univ., 930 Sakaedani, Wakayama 640-8510, Japan, irino@sys.wakayama-u.ac.jp), and Roy D. Patterson (Univ. of Cambridge, Cambridge, United Kingdom)

In the study of hidden hearing loss, it is important to characterize an individual's peripheral losses accurately while minimizing measurement time for clinical applications. In this talk, we show how peripheral losses involved in compression and auditory filtering can be estimated simultaneously and accurately using the compressive GammaChirp (cGC) filter and notched noise (NN) maskers. Sets of thresholds were gathered with asymmetric maskers and fitted with the cGC filter model using several constraints associated with cochlear filtering. The slopes of the resultant input-output functions were within the range established with temporal masking curves above 1000 Hz. Studies of the fitting process also revealed how to reduce the number of threshold measurements without unduly increasing estimation error. These improvements led to the extension of the cGC filter model to explain hearing impairment of cochlear origin. The model also led to a hearing impairment simulator which can analyze everyday sounds including speech and music with a given individual's cochlear hearing impairment, and then resythesize the sounds to provide a normal hearing listener with a realistic experience of the sound perception problems facing that hearing impaired individual. By isolating peripheral losses, the simulator will assist the specification of hidden hearing losses.

9:00

3aPPa4. Conversion of amplitude modulation to phase modulation on the basilar membrane and its implication in the perceptual consequences of disrupted temporal coding. Sho Otsuka and Shigeto Furukawa (NTT Commun. Sci. Labs., 3-1, Morinosato Wkamiya, Atsugi-shi, Kanagawa Pref. 243-0198, Japan, otsuka.s@lab.ntt.co.jp)

Disrupted fidelity of temporal-structure representation by auditory-nerve (AN) firing is a key component of hidden hearing loss. However, impacts of disturbed temporal processing on everyday listening are yet to be identified. Here we propose a new role of the AN's temporal processing, namely, encoding amplitude modulation (AM). When an AM signal with a constant-frequency carrier is fed to a generic nonlinear amplifier, the carrier phase of the output signal can be modulated, a phenomena referred to as AM-to-PM conversion. By simulating the basilar-membrane (BM) vibration and measuring otoacoustic emissions, we demonstrated that the AM-to-PM conversion takes place on the BM, too. The size of AM-induced PM depends on the stimulus level, which can be accounted for by the characteristics of BM's compressive nonlinearity. Psychoacoustic measurements further indicated the association between the listener-dependent patterns of the level dependence of AM-induced PM and of AM detection performance. These results imply that the AM-induced PM on the BM can be used as a cue for AM processing in the auditory system and that disrupted AN sensitivity to PM should have negative impacts on AM processing.

Contributed Paper

9:20

3aPPa5. Hidden hearing loss and computational models of the auditory pathway: Predicting speech intelligibility decline. Christopher Smalt, Thomas F. Quatieri (MIT Lincoln Lab., 244 Wood St., Lexington, MA 02420, Christopher.Smalt@ll.mit.edu), and Mark A. Parker (Steward St. Elizabeth Medical Ctr., Brighton, MA 02135)

Recent animal studies have shown that noise exposures cause a permanent loss of low spontaneous rate auditory nerve fibers (ANFs) and reduction of auditory brainstem response wave-I amplitudes (Kujawa and Liberman, 2009). This phenomenon is thought to create difficulties understanding speech in noise in humans, although there is currently no established clinical technique to measure cochlear synaptopathy. The goal of this research is to utilize computational models of the auditory periphery and auditory cortex to study the effect of low spontaneous rate ANF loss on the cortical representation of speech intelligibility in noise. The auditory-periphery model of Zilany *et al.* (2009, 2014) is used to make predictions of auditory nerve responses to speech stimuli in noise. The resulting cochlear neurogram, a spectrogram-like output based on ANF outputs, is then used as a foundation for two different but related cortical representations of speech: the Spectro-Temporal Modulation Index (STMI; Elihali *et al.*, 2003) and 2D Fourier Analysis (Wang and Quatieri., 2012). Reducing the number of low spontaneous rate ANFs in the cochlear neurogram was found to cause a blurring of speaker-specific cortical components, increasing the difficulty of speaker separation in the time-frequency modulation domain. Suprathreshold deficits in speech intelligibility, as simulated by auditory pathway models, may be predicting the effect of ANF loss or degradation on listening performance, and could potentially be used as part of a diagnostic tool.

9:35

3aPPa6. Effects of hidden hearing loss on the representation of speech-in-noise in the gerbil auditory midbrain. David McAlpine, Jaime Undurraga (Linguist, Macquarie Univ., Australian Hearing Hub, 16 University Ave., Sydney, NSW 2109, Australia, david.mcalpine@mq.edu.au), Jose Garcia-Lazaro, and Roland Schaette (Ear Inst., Univ. College London, London, United Kingdom)

Increasing evidence indicates that noise exposure selectively damages high-threshold auditory nerve fibres, in the absence of damage to sensory hair cells. This "hidden hearing loss (HHL)"—undetected by conventional tests such as audiometry—is suggested to account for undiagnosed difficulties processing speech in background noise. Here, we demonstrate in the midbrain of gerbils exposed to a single, controlled noise insult, and in human listeners, evidence of increased neural gain in the central auditory pathways, indicative of damage to high-threshold ANFs following noise exposure. Neural responses were higher, and discrimination performance for 60-dB SPL "vowel-consonant-vowel" stimulus (VCVs in background noise (speech-shaped, +12 to -12 dB signal-to-noise ratio) enhanced, in noise-exposed animals compared to controls. Conversely, for 75-dB SPL VCVs, neural discrimination was better in control, compared to noise-exposed, animals. A similar pattern was evident in human listeners, and was positively correlated with the ratio of wave V (generated by the midbrain) to wave I (generated by the auditory nerve) of the auditory brainstem response. The data are consistent with the hypothesis that reduced neural output from the cochlea at higher sound levels increases central gain through homeostatic mechanisms that seek to normalize neural activity.

9:55–10:15 Break

10:15

3aPPa7. No evidence for hidden hearing loss due to noise exposure in young adults with a normal audiogram. Karolina Kluk, Garreth Prendergast, Hannah Guest, Kevin J. Munro, Agnès Léger (Manchester Ctr. for Audiol. and Deafness (ManCAD), The Univ. of Manchester, Oxford Rd., Manchester m139pl, United Kingdom, karolina.kluk@manchester.ac.uk), Deborah A. Hall (National Inst. for Health Res. (NIHR) Nottingham Hearing Biomedical Res. Unit, Nottingham, United Kingdom), Michael Heinz (Dept. of Speech, Lang., & Hearing Sci., Purdue Univ., West Lafayette, IN), and Christopher J. Plack (Manchester Ctr. for Audiol. and Deafness (ManCAD), The Univ. of Manchester, Manchester, United Kingdom)

Cochlear synaptopathy, or "hidden hearing loss," refers to a loss of synapses between inner hair cells and auditory nerve fibers, and affects primarily low-spontaneous-rate fibers that encode moderate-to-high intensity sounds. In rodent models, cochlear synaptopathy has been shown to occur as a result of noise exposure even when threshold sensitivity is unaffected. We tested 126 young humans with normal audiograms and a wide range of lifetime noise exposures quantified using a questionnaire. Our electrophysiological and behavioral measures were designed to be sensitive to the contribution of low-spontaneous-rate fibers. We predicted that our test battery would reveal that noise exposure is associated with attenuated electrophysiological responses, and elevated behavioural thresholds, especially at higher levels and frequencies. However, there was no relation between noise exposure history and auditory brainstem response wave I amplitude, nor performance on psychophysical tasks including modulation detection, inter-aural phase difference discrimination, frequency discrimination, and intensity discrimination. There was also no relation between noise exposure and speech-in-noise performance nor self-report listening ability. Only the 16 kHz audiometric thresholds were correlated positively with noise exposure for females, but not for males. Our results suggest that cochlear synaptopathy due to noise exposure is not easily identifiable in young humans with a normal audiogram. [This work was supported by the MRC UK (MR/L003589/1).]

10:35

3aPPa8. Functional hearing impairments in normal-hearing listeners with a history of deployment-related blast exposure. Douglas S. Brungart (Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil), Lina Kubli (Army Public Health Ctr. (Provisional), Bethesda, MD), Sandeep Phatak, Melissa J. Kokx-Ryan, Lynn M. Bielski, LeeAnn Horvat, and Kenneth W. Grant (Walter Reed NMMC, Bethesda, MD)

Over the past decade, DoD and VA audiologists have reported large numbers of relatively young adult patients who have normal-audiometric thresholds but who report greater than expected difficulty understanding speech in noisy environments. One common theme in the history of these patients is that they self-report having experienced exposure to explosive blast as part of their military service. Recent studies in our laboratory suggest that some blast-exposed patients with normal-hearing thresholds have greater than expected difficulty in tasks that require spatial sound perception (auditory localization and N_0S_π tone detection) or those that involve the perception of time-compressed speech in reverberant environments. They also tend to report more hearing difficulties compared to non-blast exposed patients on a questionnaire that evaluates spatial perception, speech perception, and perceived sound quality. These laboratory results appear to be confirmed by preliminary results from roughly 2000 active-duty service members who participated in a brief hearing test and questionnaire at the time of their annual hearing test. Within this sample, blast-exposed listeners were 3-4 times more likely to experience subjective or objective hearing deficits than those with no blast history. These results suggest a form of hidden hearing loss that may be associated with exposure to blast. [The opinions presented are the private views of the authors and are not to be construed as official or as necessarily reflecting the views of the DoD.]

10:55

3aPPa9. Effects of aging on the absolute pitch judgment and frequency following responses of electroencephalograms. Minoru Tsuzaki, Satomi Tanaka (Kyoto City Univ. of Arts, 13-6 Kutsukake-cho, Oe, Nishikyo-ku, Kyoto 610-1197, Japan, minoru.tsuzaki@kcua.ac.jp), and Junko Sonoda (none, Weimar, Germany)

Absolute pitch (AP) possessors can name the pitch class of the note simply by hearing a periodic tone. It has been reported that the AP judgment can shift by one or two semitones when AP possessor become old. We confirmed this age-related AP shift by a series of psychophysical experiments with piano sounds as well as synthesized complex tones. AP possessors whose ages ranged from 20s to 50s participated in the experiments. The results showed that the aged participants were likely to assign the higher pitch classes than young participants for the piano sounds. Similar tendencies were observed for the synthesized, complex sounds if they contained lower order harmonics. The hearing levels, OAEs, and the frequency following responses (FFRs) of the EEG were also measured for the same participants. The hearing levels and OAE levels which could be indices in the cochlear mechanical properties could not explain the age-related AP shift. The phase locking values of the FFRs obtained at 98 and 220 Hz could neither be a good predictor of the AP shift. Although no plausible model to predict it exists, the age-related AP shift might bring another route of overlooked hearing problems.

Contributed Paper

11:15

3aPPa10. Discrimination and identification of environmental sounds among people with hearing impairment. Yuuki Yuno, Masaki Matsubara (Library Information and Media Studies, Univ. of Tsukuba, 1-2, Kasuga, Tsukuba, Ibaraki 305-8550, Japan, s1621640@u.tsukuba.ac.jp), Kei Tabaru (Education, Ibaraki Univ., Mito, Japan), Hiroko Terasawa (Library Information and Media Studies, Univ. of Tsukuba, Tsukuba, Japan), and Rumi Hiraga (Industrial Information, Univ. of Technol., Tsukuba, Japan)

We report that people with hearing impairment (HI) can better discriminate environmental sounds than identifying the same sounds. In order to investigate the perception/recognition of environmental sounds by people with HI, researchers often choose identification task as a measurement tool. However, since identification ability is developed upon discrimination ability, the discrimination ability should be studied too. We used 9 environmental sounds as stimuli for both discrimination and identification tasks, and 16 participants with HI joined the experiment. In the discrimination task, the participants listened to the paired short stimuli, and reported the perceived difference. In the identification task, the participants listened to each stimulus and reported the name of the sound source. The order of discrimination and identification tasks is counterbalanced among the participants, and we also tested 10 people with normal hearing for comparison. The results showed a clear contrast between the discrimination and identification tasks by people with HI. People with HI outperformed in discrimination tasks (97.0 % correct answer on average) than in identification tasks (35.5 % correct answer on average). This means that people with HI can discriminate most of the sounds that they cannot identify, suggesting the potential for the acquisition of identification ability.

Invited Paper

11:30

3aPPa11. Individualized assessment of suprathreshold hearing and relationship to cochlear synaptopathy. Hari M. Bharadwaj (Athinoula A. Martinos Ctr. for Biomedical Imaging, Massachusetts General Hospital, 149 Thirteenth St., Boston, MA 02129, hari@nmr.mgh.harvard.edu), Leonard Varghese, Golbarg Mehraei (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA), Christopher A. Shera (Eaton Peabody Lab. of Auditory Physiol., Massachusetts Eye and Ear Infirmary, Boston, MA), and Barbara G. Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

Threshold audiometry is currently the foundation of clinical audiology. Yet, many individuals with normal hearing thresholds (NHTs) have difficulty communicating in noisy settings. We previously documented that even among listeners with NHTs and no hearing complaints, large individual differences exist in the ability to perceive subtle temporal features of clearly audible sounds, and to selectively process target speech in the presence of competing sounds. Critically, we find that these suprathreshold differences in individual perceptual sensitivity correlate with physiological measures from the brainstem and auditory nerve. These measures, including brainstem envelope-following responses, auditory brainstem responses, and the middle-ear muscle reflex, show that some perceptually relevant physiological differences arise very early along the auditory pathway. Cochlear synaptopathy—a loss of afferent synapses and nerve terminals innervating the cochlea—is well documented in noise-exposed and aging animals (Kujawa & Liberman, Hear. Res. 2015). Cochlear synaptopathy in humans may account for both large individual differences and hearing difficulties in human listeners with NHTs. This presentation will summarize updated human evidence consistent with cochlear synaptopathy, discuss some of the challenges that remain with this interpretation, and review the search for clinical markers of such damage.

Session 3aPPb

Psychological and Physiological Acoustics: From Cochlea to Cortex (Poster Session)

Olga Stakhovskaya, Chair
Audiology and Speech Center, Walter Reed National Military Medical Center, 4954 North Palmer Road, Bldg. 19, R. 5607, Bethesda, MD 20889

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

3aPPb1. *In-vivo* intra-cellular characterization of "OFF" responses in the superior para-olivary nucleus. Tom P. Franken (Univ. of Leuven, Leuven, Belgium), Mark Sayles (Purdue Univ., West Lafayette, IN), Philip H. Smith (Univ. of Wisconsin - Madison, Madison, WI), and Philip X. Joris (Univ. of Leuven, University of Leuven, Leuven, Belgium, philip.joris@med.kuleuven.be)

The superior para-olivary nucleus (SPN) is an auditory-brainstem OFF channel. Typically, SPN neurons are inhibited during sound presentation and fire at sound offset. Release from sound-induced inhibitory potentials is the hypothesized mechanism, but this has not previously been directly observed *in vivo*. We obtained *in vivo* axonal recordings from 33 SPN neurons in the chinchilla, and patch recordings from 17 SPN cell bodies in the Mongolian gerbil. We retrieved several cells anatomically by labeling with biocytin or neurobiotin. Labeled SPN neurons had large dendritic trees, and axons heading into the lateral lemniscus. Their responses typically showed offset spiking to contralateral, and sometimes to ipsilateral stimulation. Spikes in response to amplitude-modulated tones were highly synchronized, up to hundreds of Hz (vector strength >0.9). Some SPN neurons were ITD sensitive, either to envelope or low-frequency fine-structure components. Patch-clamp recording allowed us to study sub-threshold potentials. Large IPSPs were generated by contralateral, and often also by ipsilateral sounds, with few EPSPs. Individual phase-locked IPSPs were discernable to low-frequency tones. High-frequency tones elicited a large onset IPSP, followed by sustained inhibition. For amplitude-modulated tones, IPSPs were locked to the stimulus envelope, with spikes at the offset of inhibitory events.

3aPPb2. Against "broadening with sound level". Mark Sayles (Purdue Univ., Purdue University, West Lafayette, IN 47907, sayles.m@gmail.com), Bertrand Fontaine (Univ. of Leuven, Leuven, Belgium), Philip H. Smith (Univ. of Wisconsin - Madison, Madison, WI), and Philip X. Joris (Univ. of Leuven, Leuven, Belgium)

Spectral resolution is fundamental to audition. Auditory-nerve-fiber responses provide a window onto cochlear tuning. Typically, resolution is quantified using threshold tuning curves. Superficially, these suggest dramatic and systematic filter broadening with increasing intensity, a view often invoked to explain psychoacoustic phenomena. We recorded auditory-nerve-fiber spike times from normal-hearing chinchillas in response to 0.01-20 kHz wideband noise. We gathered data over a wide intensity range from each fiber. Linear filter estimates were obtained at each level by reverse correlation. We characterized their magnitude and phase response. We considered responses from 175 fibers (CF range: [0.1, 3.0] kHz; all SR groups), with noise level varying over a 40 to 100-dB range. 10-dB bandwidth varied minimally with sound level over a 40 to 50-dB range. Some fibers demonstrated phase pivoting with level; however, most did not follow this simple scheme. Common teaching is "auditory filters broaden with level," usually in reference to threshold tuning curves. Here, gain functions constructed from responses to a fixed-level input show this broadening is not as severe as threshold tuning curves might suggest. Broadening is very subtle over a large range of "low" to "mid" input intensities (<70-dB SPL), before rapidly increasing at "high" levels (>80-dB SPL).

3aPPb3. The efficiency of the cytoarchitecture of the organ of Corti for active cochlear amplification. Hamid Motallebzadeh and Sunil Puria (Eaton Peabody Lab., Harvard Med. School, 243 Charles St., Boston, MA 02114, Hamid_Motallebzadeh@meei.harvard.edu)

A "*Y*"-shaped structure of cells formed by the basally tilted force generating outer hair cells (OHC) and apically tilted passive phalangeal process (PhP) connected to a Deiter cell (DC) has been hypothesized to be an essential organ of Corti (OoC) building block for cochlear amplification (Yoon *et al.*, 2011). We developed a COMSOL finite-element model of the mouse cochlea, taking into account the spatial arrangement of *Y*-shaped elements and the 3-D fluid-structure interaction. The model was validated by comparison with previously reported basilar membrane (BM) displacement for both passive and active cases (Lee *et al.*, 2015). The baseline model can reproduce an increase of 40 dB gain of the BM (re stapes velocity) for low level sounds. However, the absence of the PhP results in an unstable response of the BM for active cases. The model with the roles of the PhP and OHC switched, could not reproduce the active gain. Finally, the model with reversed titling orientation of OHC and PhP cells generated significantly lower gains (specially for lower frequencies), but also results in broader resonance peaks. These results demonstrate the efficiency of the natural cytoarchitecture of the OoC. [Work supported by NIH grant R01 DC 07910.]

3aPPb4. Simulating cochlear nonlinearity in an excised, active *in vitro* segment. Amir Nankali, Aritra Sasmal, and Karl Grosh (ME Dept., Univ. of Michigan, Ann Arbor, MI 48105, nankali@umich.edu)

Hearing relies on a series of coupled acoustical, electrical, and mechanical interactions in the auditory periphery. There exists a distinct nonlinear amplification process inside the cochlea that enables sound processing of a broad range of frequencies and intensities. The precise operating principles of the active mechanics underlying nonlinear cochlear amplifier remains an open question in biophysics. Using the experimental protocol devised by Chan & Hudspeth of an excised cochlear segment as a model problem, we develop a computational model for studying the active *in vitro* response of the organ of Corti (OoC) to acoustical stimulation. Both experiment and theory show that there exists a traveling wave even for a very small, 700 mm cochlear segment. However, we show that the contribution of the traveling wave on the partition dynamics is insignificant in this preparation as the phase accumulation is less than one-tenth of a cycle. This finding enables us to reduce the macroscopic fluid dynamics of the configuration to a loading added mass on the OoC and study the cochlear local dynamics. The two hypothesized active mechanisms of the cochlea (hair bundle motility and somatic electromotility) are included and the contribution of each mechanism on the cochlear amplifier is investigated. It is observed that in our model the OHC somatic electromotility is sufficient to predict the experimental results. [This work was supported by NIH grants DC-004084 and T32DC-00011.]

3aPPb5. Evolutionary elongation of the time window of auditory cortical processing: Comparison of effects of stimulus time parameters on human and macaque scalp auditory evoked potentials. Kosuke Itoh (Univ. of Niigata, Asahimachi 1-757, Niigata 951-8585, Japan, itoh@bri.niigata-u.ac.jp), Masafumi Nejime, Naho Konoike, Katsuki Nakamura (Kyoto Univ., Inuyama, Japan), and Tsutomu Nakada (Univ. of Niigata, Niigata, Japan)

Auditory cortex integrates sound signals over time for obtaining neural representations of auditory events, the time window of which affects perception and cognition. Species differences between human and macaque monkey in the time window of auditory cortical processing were investigated by comparing (1) how the cortical auditory evoked potentials (CAEP) increase in amplitude with the duration of a sound (2-200 ms), and (2) how the CAEPs to a repetitive stimulus train recover from neural adaptation with an increase in interstimulus silent interval (30-1620 ms). A virtually identical noninvasive scalp recording method was used in the two species for rigorous comparisons. First, the integration windows for the human N1 and N2 were 50 ms or longer, but the macaque N1 (mN1) and macaque N2 (mN2) were elicited with their full amplitudes by a stimulus as short as 2 ms. Second, the human N1 to a repeated stimulus continued to recover from neural adaptation as the interstimulus interval increased to 1620 ms, and a clear elicitation of N2 required an interval longer than 500 ms; in the macaque, by contrast, mN1 fully recovered from adaption by an interval shorter than 1 s, and a clear mN2 was elicited with a 30 ms gap. These findings are consistent an evolutionary elongation of the time window of human auditory cortical processing, which enables integrating greater amount of auditory information over time to obtain neural representation of complex auditory features that characterize language and music.

3aPPb6. Role of dopamine in learning beyond reinforcement: Boost of auditory learning and transfer by orally taken or gameplay-generated dopamine. Dinglan Tang (National Key Lab. of Cognit. Neurosci. and Learning, Beijing Normal Univ., Rm. 327, Yingdong Bldg., Beijing 100875, China, dinglantang@sina.com), David R Moore (Commun. Sci. Res. Ctr., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), Sygal Amitay (Medical Res. Council—Inst. of Hearing Res., Nottingham, United Kingdom), Jun-Yun Zhang (Psych. Dept., Peking Univ., Beijing, China), and Yu-Xuan Zhang (National Key Lab. of Cognit. Neurosci. and Learning, Beijing Normal Univ., Beijing, China)

We showed here that playing a video game, Tetris, in silence enhanced learning of contiguous auditory perceptual training and its transfer to working memory. The effect could not be explained by game play *per se*or the visual-spatial training involved, but rather resulted from across-time and modality interaction between game play and target training. The results led us to propose that game play releases reward signals like dopamine in the brain, which serves as a biological marker for significance of perceptual experiences and regulates learning. Consistent with this hypothesis, orally taking Madopar, a dopamine precursor, before training showed similar learning-enhancing effect as Tetris play. Madopar did not influence working memory training, indicating that the dopamine effect was perceptually but not cognitively mediated. Taken together, the results support the learning-regulating theory of dopamine, and point to a promising approach for increasing the efficiency and applicability of rehabilitative training with rewarding events like game play.

3aPPb7. Spatial relationship between two sounds in an oddball paradigm affects responses of neurons in the auditory midbrain to the two sounds. Chirag Patel and Huiming Zhang (Dept. of Biological Sci., Univ. of Windsor, 401 Sunset Ave., Windsor, ON N9B3P4, Canada, hzhang@uwindsor.ca)

Neural sensitivity to a novel sound (i.e., an occasionally occurring acoustic event) is important for hearing in a natural acoustic environment. An oddball paradigm is used to study this sensitivity. Such a paradigm is a quick succession of acoustic stimuli, with each stimulus being one of two qualitatively different sounds presented at respective probabilities (i.e., high probability standard sound and low probability deviant sound). We studied action potential discharges in the rat's auditory midbrain in response to oddball paradigms presented from free-field loudspeakers. Our results indicated that many neurons generated a stronger response to a sound presented as a deviant than as a standard stimulus when two sounds in an oddball paradigm were co-localized at the frontal midline. For many neurons, the response to a sound presented at frontal midline (either as a standard or a deviant sound) was enhanced or suppressed by relocating the other sound in an oddball paradigm from frontal midline to an off-midline angle. At extreme angles of separation, the number of neurons showing suppression exceeded that showing enhancement. Thus, a spatial separation between two sounds in an oddball paradigm affects responses to both sounds. Population results suggest that co-localization of two sounds helps maintaining neural sensitivity to both sounds.

3aPPb8. Human electrophysiology of endogenous auditory spatial attention. Erol J. Ozmeral, David A. Eddins, and Ann C. Eddins (Commun. Sci. and Disord., Univ. of South Florida, 3802 Spectrum Blvd., Ste. 210, Tampa, FL 33612, eozmeral@usf.edu)

Whereas perceiving the location of a stimulus in space relies on unique bottom-up processing, focusing attention to that location requires the availability and use of separate higher-level processes. It is unclear from previous work how selective attention might mediate and possibly sharpen spatial tuning, as behavioral tasks often lack an unattended comparison, and electrophysiological tasks often lack an attended comparison. We measured evoked responses to changes in the location of continuous noise in the free field when young, normal-hearing listeners were either passively listening or explicitly attending to one speaker location. Stimuli were presented from 1 of 5 frontal loudspeaker locations in the horizontal plane for 1.6 s before switching to another loudspeaker without pause. To ensure active attention for certain blocks, listeners were directed to press a button at the onset of a sound presented from the target location. Results show evidence of both facilitation for attended locations and suppression for unattended locations when compared to passive listening, and hemispheric distribution of activity was hemifield specific. Together, the data provide electrophysiological measures of a listener's left-right orientation and another step toward cortical models of auditory spatial attention.

3aPPb9. Machine classification of P1-N1-P2 responses elicited with a gated syllable. David J. Morris (Dept. of Nordic Studies and Linguist, Univ. of Copenhagen, Njalsgade120, Copenhagen-S 2300, Denmark, dmorris@hum.ku.dk) and Sigrid Klerke (Ctr. for Lang. Technol., Univ. of Copenhagen, Copenhagen-S, Denmark)

Onsets are generally considered to play an important role in the determination of syllable identity by human listeners. We measured P1-N1-P2 event-related potentials (ERP) from subjects ($n = 11$) in response to the syllable [sdʁɔ]. The complex onset of this syllable was sequentially gated at the consonant boundaries to yield three distinct tokens. Group ERPs to the three tokens were most different within a narrow latency window around 160 ms. This latency corresponded approximately to the lower portion of the ensemble N1-P2 deflection. Single-trial samples from the vertex channel at a range of poststimulus latencies around this point were submitted to multi-class classification by a support vector machine. The three tokens were used as class labels in a one-versus-one machine with a radial basis function kernel. Classification was trained and tested on subsets of the ERP data, and precision was found to be significantly better than chance. As the auditory stimuli used in this experiment differed only in initial consonant, these results indicate that syllable onset information is borne in the N1-P2 vertex response. They also demonstrate that a machine learning algorithm may be useful in retrieving syllable identity from single-trial ERP data.

3aPPb10. Investigation of the processing of ambiguous melodies in the brain by behavioral and neurophysiological experiments. Ryosuke Yuhara (Dept. of Information Environment, Tokyo Denki Univ., 2-1200, Muzaigakuendai, Inzai, Chiba 270-1382, Japan, 15jkm25@ms.dendai.ac.jp)

Melody recognition is not a mere succession of pitch and time perception but must be an active process to a make a meaningful line of tones. To investigate this process, we devised a form of illusory melody resembling but different from the ones used in the studies of streaming. A4 and E5 tones

are continuously presented while another tone M (middle tone, for example, C5 or C#5) between the two continuous tones is presented intermittently (0.25 s presentation and 0.25 s rest and repeating). One may hear either an illusory continuous melody of repeated M_{E5} or M_{A4} depending on the preceding unambiguous phrase of repetition of M_{E5} or M_{A4}. Assuming that this illusion reflects the active process of melody formation, we investigated 1) the transition time from the primary state (either one of the melody forms) fixed by the leading phrase behaviorally, and 2) possible neural correlates by measuring the MEG (magnetoencephalography) responses to the continuous tones separately by amplitude modulating them with different frequencies. MEG signal was filtered for the two modulation frequencies and the ASSR (auditory steady state responses) were obtained for different experimental conditions. The results so far showed wide variance among subjects but seem promising.

3aPPb11. Outer hair cell function in human ultrasonic perception. Tadao Okayasu, Tadashi Nishimura, Akinori Yamashita, Osamu Saito, Toshiaki Yamanaka, Hiroshi Hosoi, and Tadashi Kitahara (Otolaryngology-Head and Neck Surgery, Nara Medical Univ., 840, Shijo-cho, Kashihara, Nara 634-8522, Japan, tokayasu@naramed-u.ac.jp)

Ultrasound can be heard by bone conduction. It is hypothesized that this bone-conducted ultrasound (BCU) is perceived in the basal turn of the cochlea and is a result of direct ultrasonic stimulation of inner hair cells via basilar membrane vibrations without cochlear amplifier. In order to clarify the contribution of outer hair cell function to ultrasonic perception, the present study evaluated changes in hearing sensitivity for BCU and air-conducted audible sound before and after ototoxic treatment. Twenty participants (40 ears) who were undergoing chemoradiation therapy with cisplatin participated in this study. After the treatment, 62.5% ears were diagnosed with hearing loss according to the criteria of the American Speech-Language-Hearing Association. The reduction of sensitivity to air-conducted audible sound occurred in the high-frequency range, which is thought to be important for ultrasonic perception. Surprisingly, BCU sensitivity was significantly improved after the treatment. These results suggest that direct ultrasonic stimulation of the cochlea, without audible sound generation, induces ultrasonic perception. The improvement of BCU sensitivity following the cisplatin administration may be explained by hypersensitivity associated with outer hair cells' disorder.

3aPPb12. Chord consonance represented in the long-lasting sustained activity in rat auditory cortex. Tomoyo I. Shiramatsu and Hirokazu Takahashi (the Univ. of Tokyo, 4-6-1, Komaba, Meguro-ku, Tokyo 1538904, Japan, isoguchi@brain.imi.i.u-tokyo.ac.jp)

Many sounds in the environment have a rich sound spectrum, which is related to the qualia of the sound such as consonance or dissonance of chords. However, the neural representation of chord consonance is not fully understood. Here, we targeted long-lasting, sustained activities in auditory cortex and investigated whether band-specific power or phase locking value (PLV) represent consonance of the chord consisting of two pure tones. A microelectrode array with 96 recording sites recorded local field potentials (LFPs) in the fourth layer of the auditory cortex of anesthetized rats in response to continuous pure tones and chords. From the LFPs, band-specific powers and PLVs were calculated in 5 bands (theta, 4—8 Hz; alpha, 8—14 Hz; beta, 14—30 Hz; low gamma, 30—40 Hz; high gamma, and 60—80 Hz). Then, differences of band-specific power and PLV (Δpower and ΔPLV) of a chord with respect to those of the higher-frequency composition tone of the chord were compared between consonance and dissonance. Consequently, Δpower was significantly smaller, and ΔPLV was significantly larger for consonance than dissonance, suggesting that the long-lasting, sustained activity in the auditory cortex is modulated according to the complex sound spectrum, such as consonance of the chords.

3aPPb13. Auditory brainstem responses to anechoic and reverberant two-syllable speech sounds in elderly listeners. Haruna Fujihira (Graduate School of Design, Kyushu Univ., 3-1, Morinosato Wakamiya, Atsugi-shi 243-0198, Japan, fujihira_haruna_a7@lab.ntt.co.jp), Kimio Shiraishi, and Gerard B. Remijn (Faculty of Design, Kyushu Univ., Fukuoka, Japan)

Elderly listeners often complain of difficulty in understanding speech under reverberation. To investigate speech processing in elderly listeners' auditory pathways, some studies have focused on auditory brainstem responses to speech sounds (speech ABRs). In an earlier study, we measured speech ABRs to the speech sound /da/ in elderly listeners and reported that listeners with low word intelligibility under reverberation showed small amplitudes in their speech ABRs to a formants frequency of/da/ (Fujihira & Shiraishi, 2015). Reverberation is a phenomenon that causes speech sounds to overlap with following speech sounds. Therefore, the effects of reverberation on multiple syllables might be larger than those on one syllable. In present study, we measured the speech ABRs to the two-syllable sound /dada/ in 27 elderly listeners and investigated the difference between the speech ABRs to the first syllable and the second syllable in anechoic and reverberant conditions. The results showed that the amplitudes of speech ABRs to the first syllable decreased compared to those to the second syllable in the anechoic condition, but both amplitudes to the first syllable and the second syllable in the reverberant condition decreased almost equally.

3aPPb14. Investigating the contribution to anti-masking by the inhibitive networks in the dorsal cochlear nucleus: Modeling and simulation. Tzu-Chi Liu and Yi-Wen Liu (Elec. Eng., National Tsing Hua Univ., Rm. 719, Elec. Eng. and Comput. Sci. Bldg., No. 101, Section 2, Kuang-Fu Rd., Hsinchu 30013, Taiwan, s102061556@m102.nthu.edu.tw)

In psychoacoustics, masking happens when the perceptibility of a target sound is influenced by another sound, which is called the masker. The medial olivocochlear (MOC) reflex pathway, which receives information from the cochlear nucleus and gives feedback to the cochlea, has been said to modify the gain function of the cochlea and enhance the audibility of sounds in noise. This enhancement of audibility is referred to anti-masking. To simulate masking and anti-masking, a tone-in-noise stimulus is delivered to an integrated model comprised of a nonlinear model of cochlear mechanics and a leaky integrate-and-fire network describing the dynamics of the MOC pathway. In particular, the T-multipolar (TM) cell in the ventral cochlear nucleus is chosen to transfer signals to the MOC, and the conductance of the outer hair cell is adjusted by the MOC efferent output to simulate anti-masking. Further, the inhibition from the tuberculoventral cell in the dorsal cochlear nucleus to the TM cell is also included in the model so its influence on the auditory nerve rate-level curves can be investigated in the tone-in-noise condition.

3aPPb15. High-pass filtering obscures stimulus encoding characteristics of the auditory brainstem response: Evidence from click and tone stimuli. Alexandra R. Tabachnick (Psychol. and Brain Sci., Univ. of Delaware, McKinley Lab, Newark, PA 19716, atabach@udel.edu) and Joseph C. Toscano (Psych., Villanova Univ., Villanova, PA)

The auditory brainstem response (ABR) is an electrophysiological measure of early auditory processing. While previous work has examined ABRs to clicks, tones, speech, and music, it remains unclear how changes in acoustic properties (e.g., frequency) map onto specific changes in ABR components. This may be partly due to filtering during data processing. High-pass filtering can severely distort cortical and subcortical responses, potentially obfuscating how stimuli are encoded. To address this, we measured ABRs to a wide range of pure tones (250 to 8000 Hz) and examined how high-pass filtering affects tone- and click-evoked ABRs. In Experiment 1, various high-pass filter settings (0.1-300 Hz) were applied to click-evoked

ABRs. In Experiment 2, ABRs to brief tones across a six-step frequency continuum were collected, and the same high-pass filter settings were applied. Results indicate that excessive high-pass filtering diminishes the amplitude of ABR components, consistent with previous findings. In addition, filtering can obscure true effects of stimulus frequency. With appropriate filters, we find that the amplitude of wave V tracks stimulus frequency log-linearly, demonstrating that tonotopic organization is preserved and easily detectable early in processing. Future ABR work should minimize the use of high-pass filters when studying the encoding of acoustic information.

3aPPb16. Level-tolerant duration selectivity in the auditory cortex of the velvety free-tailed bat *Molossus molossus*. Silvio Macías (Dept. of Animal and Human Biology, Univ. of Havana, Havana, Cuba), Annette Hernández-Abad (Dept. of Animal and Human Biology, Univ. of Havana, Havana, Cuba; Dept. of Physiol., University of AB, Edmonton, AB T6G 2H7, Canada, annette3@ualberta.ca), Julio C. Hechavarría, Manfred Kössl (Institut für Zellbiologie und Neurowissenschaft, J.W. Goethe Universität Frankfurt, Frankfurt, Germany), and Emanuel C. Mora (Dept. of Animal and Human Biology, Univ. of Havana, Havana, Cuba)

It has been reported previously that in the inferior colliculus of the bat *Molossus molossus*, neuronal duration tuning is ambiguous because the tuning type of the neurons dramatically changes with the sound level. In the present study, duration tuning was examined in the auditory cortex of *M. molossus* to describe if it is as ambiguous as the collicular tuning. From a population of 174 cortical 104 (60 %) neurons did not show duration selectivity (all-pass). Around 5 % (9 units) responded preferentially to stimuli having longer durations showing long-pass duration response functions, 35 (20 %) responded to a narrow range of stimulus durations showing band-pass duration response functions, 24 (14 %) responded most strongly to short stimulus durations showing short-pass duration response functions and two neurons (1 %) responded best to two different stimulus durations showing a two-peaked duration-response function. The majority of neurons showing short- (16 out of 24) and band-pass (24 out 35) selectivity displayed “O-shaped” duration response areas. In contrast to the inferior colliculus, duration tuning in the auditory cortex of *M. molossus* appears level tolerant. That is, the type of duration selectivity and the stimulus duration eliciting the maximum response were unaffected by changing the sound level.

3aPPb17. Comparing auditory perceptual thresholds in pediatric and adult cochlear implant populations. Gabrielle O’Brien, Mishaela DiNino (Speech and Hearing Sci., Univ. of Washington, 1417 N.E. 42nd St., Box 354875, Seattle, WA 98105-6246, andronovhopf@gmail.com), J. D. Biesheuvel, Jeroen J. Briaire, Johan H. Frijns (Leiden Univ. Medical Ctr., Leiden, Netherlands), and Julie A. Bierer (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

It is well known that cochlear implant listeners show enormous variability in speech recognition performance, but the individual differences that affect outcomes are still poorly understood. The overall level and variability of psychophysical thresholds obtained with a focused electrode configuration may account for some of the variability in outcomes, as these measures might reflect the quality of the electrode-neuron interface. To date, psychophysical thresholds using direct stimulation have not been assessed in children. In this work, we compare such thresholds between pediatric and adult cochlear implant listeners. Thresholds were collected from 12 pediatric and 34 adult ears using a fast threshold sweeping method with both focused (partial quadrupolar) and broad (monopolar) electrode configurations. On average, both focused and broad thresholds were 4 dB lower in the pediatric population compared to the adult population. Intra-array variability in thresholds was found to be smaller in pediatric than adult populations for broad but not focused stimulation. Increased threshold observed for focused stimulation compared with broad, quantified as the ratio of focused to broad thresholds, did not differ significantly across populations. These results suggest that there may be differences in the quality of the electrode-neuron interfaces observed in pediatric and adult cochlear implant listeners.

3aPPb18. Interaural-time-difference discrimination as a measure of place of stimulation for cochlear-implant listeners with single-sided deafness. Olga Stakhovskaya, Gerald I. Schuchman (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., 4954 North Palmer Rd., Bldg. 19, R. 5607, Bethesda, MD 20889, olga.stakhovskaya.ctr@mail.mil), Matthew J. Goupell (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, College Park, MD), and Joshua G. Bernstein (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

Recently, some patients with single-sided deafness (SSD) have received a cochlear implant (CI), with the goal of improving their spatial-hearing abilities. Because electrode arrays typically do not reach the cochlear apex, standard frequency-to-electrode allocations are likely to create a substantial interaural cochlear place-of-stimulation mismatch, and might not optimize binaural function for SSD-CI listeners. As a first step, we evaluated whether a test of interaural-time-difference (ITD) discrimination could be used to identify the cochlear place of stimulation for individual electrodes. Six SSD-CI listeners were presented with 500-ms bursts of 100-pps electrical pulse trains on a single electrode, and bandlimited pulse trains with variable carrier frequencies in the acoustic ear. Listeners discriminated between two “static” intervals (each containing four bursts with constant ITD) and a “moving” interval (four bursts with variable ITD). For most listeners and electrodes, performance peaked at a particular acoustic carrier frequency, which was on average 0.57 octaves higher (range = -0.2-1.1 octaves) than the standard clinically allocated center frequency. These results demonstrate that an ITD-discrimination task can identify the optimal frequency allocation required for a given CI electrode to maximize binaural function for SSD-CI listeners. The opinions and assertions presented are the private views of the authors and are not to be construed as official or as necessarily reflecting the views of the Department of Defense.

3aPPb19. Psychophysical tuning curves as a measure of electrode position in cochlear implant listeners. Lindsay DeVries and Julie A. Bierer (Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, lindsdev@uw.edu)

Speech perception abilities vary widely among cochlear implant listeners. A potential source of this variability is electrode position; suboptimal placement has been associated with poorer outcomes. Insight into electrode position can be obtained via postoperative CT imaging, specifically the distance of each electrode from the modiolus. Additional information can be obtained with behavioral measures, which are sensitive to electrode position and neural integrity.Electrode-to-modiolus distance can explain some variability in behavioral thresholds; however, the relationship with psychophysical tuning curves (PTCs) has not been evaluated, which may provide a more detailed assessment of local variations in channel interaction. Four unilaterally implanted adults with the Advanced Bionics HiRes90K device participated. CT scans were obtained and image reconstructions were performed. Using a fast, sweep procedure, PTCs were collected for all available electrodes using the steered quadrupolar configuration within a forward-masking paradigm. Spread of neural excitation was quantified with the equivalent rectangular bandwidth (ERB). Preliminary data show a correlation between electrode-to-modiolus distance and the ERB of PTCs, suggesting poorer electrode placement may cause broader activation. The goal of this research is to develop a fast, non-radiologic method for estimating electrode position, which may lead to improved device programming by reducing unwanted channel interaction.

3aPPb20. Interaural spectral mismatch and binaural fusion in simulated cochlear-implant listening for single sided deafness. Jessica M. Wess (Neurosci. and Cognit. Sci. Program, Univ. of Maryland, 4954 N Palmer Rd., Bethesda, MD 20889, jwess@umd.edu), Nathan J. Spencer (Battlespace Acoust., Wright-Patterson AFB, Air Force Res. Lab., Dayton, OH), and Joshua G. Bernstein (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

Cochlear implants (CIs) can restore some spatial-hearing capabilities for individuals with single-sided deafness (SSD). However, there is a considerable variability in outcomes among SSD-CI listeners. One possible cause is spectral mismatch between the acoustic and CI ears. Vocoder simulations of

SSD-CI listening were used to investigate the effect of interaural spectral mismatch on subjective fusion for normal-hearing listeners. A virtual cocktail party was created by presenting combinations of 1-6 concurrent talkers to the left, right or both ears. After each trial, listeners were asked how many total voices were heard. If listeners were able to perceptually fuse the voices that were presented to both ears, they should be more likely to report the correct number of total talkers in the scene. If not, they should report more total talkers. Preliminary results show that listeners reported more total talkers when the vocoded speech was processed with a frequency-matched map (simulating a standard CI frequency allocation) then with a place-matched map. These data suggest that place-matched CI mapping has the potential to provide SSD-CI listeners with more opportunities for binaural fusion and improved spatial hearing. This method could be applied to SSD-CI listeners to evaluate mapping strategies and outcomes after implantation. [The opinions and assertions presented are the private views of the authors and are not to be construed as official or as necessarily reflecting the views of the Department of Defense.]

3aPPb21. Comparison of brainstem frequency-following responses associated with auditory streaming based on spectral and temporal cues. Shimpei Yamagishi (Dept. of Information Processing, Tokyo Inst. of Technol., 4259, Nagatsuta-cho, Midori-ku, Yokohama-shi, Kanagawa-ken 226-8503, Japan, yamagishi@u.ip.titech.ac.jp), Sho Otsuka, Shigeto Furukawa, and Makio Kashino (NTT Commun. Sci. Labs., NTT Corp., Kanagawa, Japan)

With the ABA streaming paradigm (van Noorden, 1975), Yamagishi *et al.* (2016) used spectrally differing A and B tones and showed that the auditory brainstem frequency-following response (FFR) to the second A tone co-varied with the perceived auditory stream. This study examined whether this effect is specific to the case in which the streaming is based on the spectral cues. We compared stream-perception-related FFRs between ABA stimuli based on spectral and temporal cues. In the *spectral* condition, the A and B tones differed in frequency (315 and 400 Hz, respectively). In the *temporal* condition, the stimulus consisted of sinusoidally amplitude-modulated 4-kHz tones that had identical long-term excitation patterns but differed in modulation frequency (60 and 170 Hz, respectively). We analyzed the spectral amplitudes of the FFRs at the stimulus frequencies for the spectral condition and those at the stimulus modulation frequencies for the temporal condition. We found that the FFR amplitudes to the second A tone generally tended to be larger in the two-stream percept than in the one-stream one for both the spectral and temporal conditions. This result implies that the mechanism responsible for the stream-perception-related FFR is common across cue domains.

3aPPb22. Sub-clinical auditory neural deficits in patients with type 1 Diabetes Mellitus. Arwa AlJasser, Kai Uus, Richard Baker, and Christopher Plack (School of Psychol. Sci., Univ. of Manchester, Ellen Wilkinson Bldg., Rm. A3.14, Manchester M13 9PL, United Kingdom, arwa.al-jasser@postgrad.manchester.ac.uk)

Diabetes mellitus (DM) is associated with a variety of sensory complications. Very little attention has been given to auditory neuropathic complications in DM. The aim of this study was to determine whether type 1 DM affects neural coding of the rapid temporal fluctuations of sounds, and how any deficits may impact on real-world listening tasks. Participants were 30 young normal-hearing type 1 DM patients, and 30 age-, sex-, and audiogram-matched healthy controls. Tests included non-invasive electrophysiological measures of auditory nerve and brainstem function using the click-evoked auditory brainstem response (ABR), and of brainstem neural temporal coding using the sustained frequency-following response (FFR), as well as behavioural tests of temporal coding (interaural phase difference, IPD, discrimination and the frequency difference limen, FDL) and tests of speech perception in noise. There were no significant differences between DM patients and controls in the ABR. However, the DM group showed significantly lower FFR responses, higher IPD and FDL thresholds, as well as worse speech-in-noise performance. The results suggest that type 1 DM is associated with degraded neural temporal coding in the brainstem in the absence of an elevation in audiometric threshold, and that this deficit may impact on real-world hearing ability.

3aPPb23. Binaural sensitivity in children with bilateral cochlear implants and in normal hearing children. Ruth Litovsky and Erica E. Bennett (Commun. Sci. & Disord., Univ. of Wisconsin-Madison, 1500 Highland Ave., Waisman Ctr. Rm. 521, Madison, WI 53705, litovsky@waisman.wisc.edu)

Children with bilateral cochlear implants (CIs), and normal hearing (NH) children were studied. One goal was to understand if sensitivity to interaural time difference (ITD) and interaural level difference (ILD) is attained at young ages with acoustic hearing. A second goal was to determine if children who hear through electrical stimulation of the auditory nerve are also sensitive to ITD and ILD cues. The bilaterally implanted children received stimulation through synchronized research processors, which is unlike the clinical speech processors that they use every day, which lack interaural synchronization. Sensitivity to ILDs was adult-like in NH children, and all children with CIs were able to use ILDs. Sensitivity to ITDs was, on average, adult-like in NH children, but many of the CI users were unable to use ITDs to determine if a sound was presented to the right or left. This contrasts with prior findings that children with CIs can detect the presence of a stimulus with a binaural cue, regardless of whether they could discriminate ITDs or not. Results underscore the importance of task selection in evaluating binaural sensitivity. Data are interpreted in the context of auditory plasticity and peripheral integrity for the development of binaural coding.

3aPPb24. Dynamic representation of acoustic features in neural responses to continuous speech. Bahar Khalighinejad, Guilherme Da Silva, and Nima Mesgarani (Elec. Eng., Columbia Univ., 500 W. 120th St., Mudd 1310, New York, NY 10027, bk2556@columbia.edu)

Humans are unique in their ability to communicate using spoken language. However, it remains unclear how distinct acoustic features of speech sounds are represented in the auditory pathway over time. In this study, we applied a novel analysis technique to electroencephalography (EEG) signals as subjects listened to continuous speech and characterize the neural representation of acoustic features and the progression of responses over time. We averaged the time-aligned neural responses to phoneme instances and calculated a phoneme-related potential (PRP). We show that phonemes in continuous speech evoke multiple observable responses which are clearly separated in time. These recurrent responses have different scalp distributions, and occur as early as 50 ms, and as late as 400 ms after the phoneme onset. We show that the responses explicitly represent the acoustic distinctions of phonemes, and that linguistic and non-linguistic information appear at different time intervals. Finally, we show a joint encoding of phonetic and speaker information, where the neural representation of speakers is dependent on phoneme category. This study provides evidence for a dynamic neural transformation of low-level speech features and form an empirical framework to study the representational changes in learning, attention, and speech disorders.

Session 3aSA

Structural Acoustics and Vibration: Dynamics of Ribbed Structures

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Invited Papers

8:30

3aSA1. Acoustic scattering from finite bilaminar composite cylindrical shells with doubly periodic stiffeners—3-D solution. Sabih I. Hayek (Eng Sci. and Mech., Penn State, 953 McCormick Ave, State College, PA 16801-6530, sihesm@engr.psu.edu) and Jeffrey E. Boisvert (NAVSEA Div Newport, Newport, RI)

The acoustic scattering from an insonified finite bilaminar cylindrical shell is analyzed. The shell is reinforced by a two sets of rib-stiffeners to form a doubly periodic stiffened shell. The two cylindrical shell laminates are perfectly bonded having the same lateral dimension but have different radii and material properties The two sets of rib-stiffeners have rectangular cross-sections and are perfectly bonded to the inside of the inner shell. One set of stiffeners is heavy and the other set is light, so that a finite number of light stiffeners are set within every pair of heavy stiffeners. The bilaminar stiffened shell is analyzed using the exact theory of three-dimensional elasticity. The stiffeners are analyzed as elastic rings with extensional, torsional, and in-plane and out-of-plane flexural rigidities. The scattered acoustic farfield is evaluated for various incident plane wave frequencies. Shells with different combinations of light to heavy stiffener geometries are analyzed. A uniform steel stiffened shell submerged in water and stiffened with doubly periodic stiffeners was initially analyzed. A second shell made up of an outer elastomer shell bonded to an inner stiffened steel shell was also analyzed. [Work supported by NAVSEA Division Newport under ONR Summer Faculty Program.]

8:50

3aSA2. Reduction of transmitted power on a plane by using mode pair cancellation. Toru Yamazaki (Mech. Eng., Kanagawa Univ., 3-27-1 Rokkakubashi, Kanagawa-ku, Yokohama 221-8686, Japan, toru@kanagawa-u.ac.jp)

Reduction of structure-borne sound is an important issue for industry mechanical products. This paper presents a new idea for controlling structural intensity (SI) on panel structure derived from a cancellation of SI due to mode pairs of multiple roots. At first the modal formulation of SI on a flat plate is summarized. Next we discuss the change of SI on a flat plate by attaching a reinforcement beam cross the plate in one direction. It is shown that this attachment give multiple natural frequencies, and their mode pairs and the pairs of modal components of SI are in same phase in the area of half of the plate and in anti-phase in that of the other half. Then the modes and modal components of SI can be said to be canceled by each modal pairs in the area of half on the plate. Based on this cancellation due to modal pairs, a new method for reducing transmitted power to a given area on the plate has been derived and proposed by attaching a beam on the plate for providing partial symmetry on the plate. It is demonstrated that the partial attachment of a beam on the plate based on the proposed method can reduce the transmitted power to the given area by the cancellation due to the pairs of SI modal components.

9:10

3aSA3. Influence of stiffeners and internal structures on the vibroacoustic behavior of submerged cylindrical shells under random excitations. Valentin Meyer (DCNS Res., 199 Ave. Pierre-Gilles de Gennes, Ollioules 83500, France, valentin.meyer@dcnsgroup.com), Laurent Maxit (LVA, INSA Lyon, Lyon, France), Christian Audoly, and Ygaäl Renou (DCNS Res., Ollioules, France)

The vibroacoustic behavior of structures excited by random pressure fields such as turbulent boundary layers or diffuse sound fields is of great interest for industrial applications. Many works have been carried out for periodically stiffened plates. In particular, the influence of Bloch-Floquet waves on the panel radiation has been highlighted. However, few studies have investigated more complex structures under random excitations. The present work studies the influence of internal structures on the vibro-acoustic behavior of submerged cylindrical shells. The geometric complexity is successively increased by including periodic, non-periodic stiffeners and various internal frames. The numerical prediction is based on the combination of two methods developed by the authors. The first one is the wavenumber-point (k,M) reciprocity technique. This method estimates the response of the system at point M from the shell velocity in the wavenumber space under a point excitation at M. The velocity field is estimated with the second method, called the Condensed

3a WED. AM

Transfer Function method. It is a substructuring approach which couples a semi-analytical model of a submerged cylindrical shell with Finite Element models of axisymmetric and non-axisymmetric frames. Numerical results are analyzed to evaluate the influence of the stiffeners and the internal structures on the shell radiation.

9:30

3aSA4. Exhaustive optimization of rib-stiffened, layered plate structures for acoustic response. Heather M. Reed (Weidlinger Assoc., Inc., New York, NY) and Jeffrey Cipolla (Weidlinger Assoc., Inc, 1825 K St NW, Ste. 350, Washington, DC 20006, cipolla@wai.com)

A previously reported structural-acoustic frequency-domain formulation for layered, ribbed structures is used here as the basis for an approach to optimize these systems. We will review a previously reported singular perturbation approach to resolve convergence difficulties in both planar and cylindrical configurations, and discuss verification and validation. The significant successes of topological structural optimization for strength, weight, and efficiency are more challenging to repeat in coupled wave-bearing systems with many modes present. This is due, at least, to the difficulties of defining appropriate cost and regularization functions applicable across frequency ranges of interest and across the orders of magnitude of response characteristic of vibroacoustic systems. We discuss overcoming these difficulties by evading them entirely: the layered ribbed structural-acoustic model we use is sufficiently fast that sophisticated optimization algorithms are not required. To understand the dependence of critical solution performance metrics on the input parameters, we simply sample the design space exhaustively and develop graphical representations of the relative optimality of solutions. By using sensitivity analysis (effected through Fisher Information) and uncertainty quantification (UQ), optimal layered designs are obtained and balanced by performance robustness. In this talk, we will present examples of applying the approach to realistic systems of interest.

9:50–10:05 Break

10:05

3aSA5. Active control of bending waves propagating in an orthotropic rectangular panel. Hiroyuki Iwamoto (Seikei Univ., 3-3-1 kichijoji-kitamachi, Musashino, Tokyo 180-8633, Japan, hiroyuki-iwamoto@st.seikei.ac.jp) and Nobuo Tanaka (Tokyo Metropolitan Univ., Tokyo, Japan)

Due to the recent advances in microprocessors, active control of a flexible structure has become a realizable method for suppressing the vibration. Conventional modal-based method encounters difficulties in controlling a distributed parameter structure, since such a structure has an infinite number of vibration modes. To overcome this problem, active wave control has been studied in recent years. This paper presents an active wave control of an orthotropic rectangular panel by expanding the conventional active wave control method for a two-dimensional structural case. First, a transfer matrix method for a rectangular panel is introduced to describe the wave dynamics of the structure. This is followed by the derivation of feedforward control laws for absorbing reflected waves or eliminating transmitted waves. In the proposed method, the control laws are including the modal actuation scheme for uncontrolled direction. Then, from a viewpoint of numerical analyses, basic properties of the proposed method are verified. It is found that the reflected wave absorbing control enables the inactivation of all vibration modes in the controlled direction and the transmitted wave eliminating control enables the generation of an almost vibration-free state.

10:25

3aSA6. Prediction of reflection and transmission by an elastic barrier with periodic structural discontinuities forced by oblique acoustic waves. Donald B. Bliss, Mauricio Villa, and Linda P. Franzoni (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Box 90300, Durham, NC 27708-0300, dbb@duke.edu)

An analysis method is developed for acoustic reflection and transmission from an infinite fluid-loaded flexible barrier with spatially periodic discontinuities. The barrier has acoustic fluids on both sides, which can be dissimilar. The structure is excited by an oblique-wave incident acoustic field. The fully-coupled structural/acoustic problem is treated by the method of Analytical-Numerical Matching (ANM). The ANM framework separates the problem into a global numerical solution and local analytical solutions. ANM handles rapid spatial around the structural discontinuities, improving the accuracy and convergence rate of reflected and transmitted pressures. Furthermore, the ANM approach offers a way to handle the mathematical difficulties associated with coincidence frequencies. The periodic spatial discontinuities create variations from simple specular-like directivity with multiple reflection and transmission angles, the effect being most pronounced at structural resonances. The periodic discontinuities can be thought of as redirecting a portion of the structural energy into resonant substructures having wavenumbers different from the oblique wave forcing, reradiating reflection and transmission fields with different directivity patterns. Discrete frequency and broadband results are presented. The goal is to develop efficient first-principles methods for structural-acoustic reflection and transmission between coupled acoustic spaces and into surrounding media.

10:45

3aSA7. Sound radiation from a fluid-loaded and elastically coated infinite plate with attached periodically located inhomogeneities. Yanni Zhang, Jie Pan, and Hai Huang (Zhejiang Univ., 38 Zheda Rd., Hangzhou 310027, China, yanni_zhang@zju.edu.cn)

An analytical model is developed for evaluating the sound radiation characteristics of a fluid-loaded and elastically coated infinite plate attached with periodically located inhomogeneities. The vibration and sound responses of such a periodic structure under a local excitation are obtained through wavenumber transformation method. The radiation characteristics of the plate are investigated via comparing with those of a plate with a single inhomogeneity and one without inhomogeneities. Three interesting features of the sound radiation are observed, which are related to plate's global resonant modes, local resonances of the trapped modes formed around each inhomogeneity, and the band-pass characteristics of the periodicity, respectively. These features are explained by the variation in the surface supersonic wavenumber components and the waveguide properties of the fluid-loaded composite plate and the coherent resonance of all inhomogeneities.

11:05

3aSA8. How the sound adjustment process of cymbals affects their vibration and sound radiation characteristics. Fumiyasu Kuratani, Tatsuya Yoshida (Mech. Eng., Univ. of Fukui, 3-9-1 Bunkyo, Fukui 910-8507, Japan, kuratani@mech.u-fukui.ac.jp), Toshio Koide (Koide Works, Ltd. (Koide Cymbals), Osaka, Japan), Taiji Mizuta (Osaka Alloying Works, Co., Ltd., Fukui, Japan), and Kozo Osamura (Res. Inst. for Appl. Sci., Kyoto, Japan)

Cymbals are percussion instruments. They vibrate and radiate sounds when hit with a drumstick or when used in pairs and the sounds depend on the vibration characteristics. Cymbals are made through spin forming, hammering and lathing. The spin forming creates the domed shape of the cymbal, determining its basic vibration characteristics. The hammering and lathing produce specific sound adjustments by changing the vibration characteristics. In this study, we focus on how the hammering affects the vibration and sound radiation characteristics. The hammering produces many shallow dents over the cymbal's surface, generating residual stresses in it. These residual stresses change the vibration characteristics. We perform finite element analysis of the hammered cymbal to obtain its vibration and sound radiation characteristics. In the analysis, we use thermal stress analysis to reproduce the stress distribution and then with this stress distribution we perform vibration analysis. The results show that the effect of thermal load (i.e., hammering) depends on mode: an increase or decrease in the natural frequency. The difference between the modes changes the peak frequencies and their amplitudes in the frequency response. As a result, the deflection shapes and their sound radiation efficiencies at the peak frequencies are changed.

Contributed Paper

11:25

3aSA9. Finite-difference time-domain analysis of the vibration characteristics of building structures using a dimension-reduced model. Takumi Asakura (Tokyo Univ. of Sci., 2641 Yamazaki, Noda, Chiba 278-8510, Japan, t_asakura@rs.tus.ac.jp), Masahiro Toyoda (Kansai Univ., Osaka, Japan), and Tohru Miyajima (Inst. of Technol., Shimizu Corp., Tokyo, Japan)

In order to accurately predict the vibration characteristics of buildings, wave-based numerical methods are effective from the viewpoint of the modeling accuracy of the physical mechanism and the detailed geometries of the simulated field. However, because of the performance of current PCs, the prediction of real-scale problems remains difficult. In order to address such problems, we herein propose a vibration simulation method for a beam-plate structure using a dimension-reduced modeling method. The target structure is modeled as a composite structure consisting of two-dimensional plate elements and one-dimensional beam elements, which are coupled based on the implicit finite-difference approximation scheme. By applying such a low-dimensional element, a faster simulation that requires less memory, as compared with a three-dimensional discretization scheme, is made available. To validate the method, the vibration characteristics obtained by the proposed scheme are compared to the measured results for model-scale and full-scale structure. The comparison of the measurement and simulation results suggest that the proposed method can be used to accurately simulate a multilayered building structure.

Session 3aSC

Speech Communication: Double-Weak Theory of Speech Production and Perception: A Session in Honor of Terrance Nearey

Michael Kiefte, Cochair
Human Communication Disorders, Dalhousie University, 1256 Barrington St., Halifax, NS B3J 1Y6, Canada

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Design Information Sciences, Wakayama University, 930 Sakaedani, 930, Wakayama 640-8510, Japan

Chair's Introduction—8:00

Invited Papers

8:05

3aSC1. Incongruence in second language vowel perception and production. Ron I. Thomson (Appl. Linguist, Brock Univ., 1812 Sir Isaac Brock Way, St. Catharines, ON L2V 5E6, Canada, rthomson@brocku.ca) and Murray J. Munro (Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

Nearey's work on double-weak theory has brought to the fore the conceptualization of native-language (L1) speech perception and production as autonomous subsystems. In second-language (L2) phonetics research, the perception-production relationship is also a central concern, although the facilitating and hampering effects of a previously learned phonological inventory must be taken into account (Flege, 2003). Some current views of L2 phonetic learning assume that a reorganization of perceptual knowledge normally comes first, and that production eventually falls into line with perceptual representations (Huensch, 2013; Thomson, 2011, 2013). However, the available data point to a number of complexities in the relationship that have yet to be accounted for. In the present report, we synthesize new data on vowel perception and production by English learners from a variety of L1 backgrounds. Among the key findings are that (1) perceptual training on vowels in particular phonetic contexts leads to very limited transfer of learning to new phonetic contexts, (2) perceptual accuracy only weakly predicts production accuracy, especially for cross-linguistically marked vowels, (3) word familiarity and frequency correlate with vowel production accuracy, and (4) elicitation techniques influence production accuracy. These issues are discussed in terms of the problems they pose for models of phonetic learning.

8:25

3aSC2. Investigating the sliding template model of vowel perception. Santiago Barreda (Linguist, UC Davis, Dept. of Linguist, Edmonton, AB T6G 2E7, Canada, sbarreda@ucdavis.edu)

According to the sliding-template model of vowel perception, vowel quality is specified by the frequencies of the first three formants, and within-category variability in formant-patterns is primarily according to a single scaling-parameter [Nearey, J. Acoust. Soc. Am. 85, 2088-2113, 1989]. In this view, speaker normalization in vowel perception centers on the estimation of the spectral-scaling parameter for a speaker. This process may include consideration of "indirect" evidence such as fundamental frequency (f0), or information about speaker size or sex. This model also suggests the potential integration of vowel perception and speaker-size estimation via shared use of the estimated scaling parameter, which is related to speaker vocal-tract length and height. An experiment is presented where listeners were asked to listen to vowel stimuli whose formant patterns were potentially ambiguous between the /æ/ of a larger speaker and the /ʌ/ of a smaller speaker. Listeners were asked to make a vowel-category judgment, and to estimate the height of the apparent speaker in feet and inches. Results are consistent with predictions made by the sliding template model: apparent speaker size was predictive of perceived vowel quality independently of the acoustic characteristics of a sound, and f0 appears to affect vowel quality primarily indirectly.

8:45

3aSC3. Formants in speech perception. Michael Kiefte (Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, mkiefte@dal.ca)

Formants, or vocal-tract resonances, have played a dominant role in the study of both speech production and perception, particularly with vowels. They form the basis of descriptions of speech in phonetics, speech pathology, speaker verification, sociolinguistics, language acquisition, as well as in many other fields. In contrast, work in engineering applications of speech processing—specifically automatic speech recognition—typically ignores formants in favor of acoustic properties that are significantly easier to extract but which make no assumptions regarding the nature of speech acoustics. This talk describes studies that explore how listeners process formant-like information in speech and how this evidence might relate to speech perception.

9:05

3aSC4. Unusual public voices. Peter F. Assmann (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, MS GR 41, Box 830688, Richardson, TX 75075, assmann@utdallas.edu)

Terry Nearey has a long-standing interest in vocal properties that carry phonetic information and simultaneously contribute to voice quality. He has drawn attention to the perceptual importance of the relationship between time-averaged fundamental frequency (f0) and formant frequencies and shown how combinations of these properties conspire to produce unique and highly identifiable voices, such as that of the late television cook Julia Child, who had atypically low formants but f0 in the expected range; and the cartoon figure Popeye (played by Jack Mercer) who had relatively high formant frequencies but a normal f0 range. This talk will present examples from Terry's collection along with acoustic analyses and a discussion of their perceptual implications.

9:25

3aSC5. Making decisions about modeling decision-making when analyzing speech perception data. Noah H. Silbert (Commun. Sci. and Disord., Univ. of Cincinnati, French East 344, 3202 Eden Ave., Cincinnati, OH 45267, silbernh@ucmail.uc.edu)

Speech perception researchers are typically interested in, well, perception. But perception is not directly observable. An experimental participant hears (and maybe sees) a stimulus, but the researcher only observes the listener's response (e.g., which button is pushed, how long it takes to push the button). Crucially, a listener's response doesn't directly, or solely, reflect perceptual processing; listeners' responses also reflect what we might call response selection or decision-making. While there is a sizable sub-field in psychology devoted to the study of judgment and decision making, these issues are of more peripheral concern for speech perception researchers. Nonetheless, I will argue that it is important to take response selection into account when analyzing perception data. Terry Nearey is one of a small number of speech researchers who has consistently taken decision-making seriously when analyzing speech perception data. I will discuss a number of mathematical approaches to simultaneously modeling perception and response selection and how changes in modeled perception and decision-making predict distinct observable patterns in data. The importance of decision-making in the study of perception will be discussed and illustrated via a number of Terry Nearey's studies as well as my own.

9:45–10:00 Break

10:00

3aSC6. The perception of spontaneous speech: "Lo how complex". Benjamin V. Tucker (Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca)

While most of the speech that listeners encounter is produced during spontaneous conversation, researchers actually know very little about this type of speech. What we do now is that there is tremendous variability and complexity in spontaneous speech that is not found in laboratory styles of speech. These differences compound and present new challenges for speech researchers, and in this case speech perception. It is possible that the changes found in spontaneous speech are systematic; however, it is also possible that these changes increase the noise in the speech signal. The vast majority of recent research on spontaneous speech has focused on word recognition and has not focused on aspects of perception and how spontaneous speech impacts theories of speech perception. In this presentation, I provide several examples of the variability and complexity found in spontaneous speech. I explore some of the challenges spontaneous speech creates for models of speech perception. I also discuss some perception data on spontaneous speech which sheds some light on future directions in research on speech perception.

10:20

3aSC7. What does sinewave speech tell us about phonetic perception? James Hillenbrand (Western Michigan Univ., 1903 W Michigan Ave., Kalamazoo, MI 49008, james.hillenbrand@wmich.edu)

It has been 35 years since the striking demonstration that intelligible speech can be created by mixing sinusoids that follow the formants of natural speech [Remez *et al*. (1981) Science **212**, 947-950]. A good deal has been written about what sinewave speech (SWS) might reveal about the mechanisms underlying phonetic perception. However, most of the experimental work on SWS has been carried out using well-formed sentences whose intelligibility depends on many factors other than phonetic identification. Results will be reported for experiments designed to measure SWS intelligibility explicitly at the phonetic level. For example, listeners were asked to identify the vowel in SW /hVd/ syllables. Intelligibility was far above chance (55%) but dramatically lower than that of the original utterances. Further tests showed: (1) intelligibility increased substantially when the SW syllable was preceded by a brief SW carrier phrase (CP); (2) tests that alternated trials with and without the SW CP showed that CP enhancement is a real-time effect; (3) SW vowels can be classified using a template-matching algorithm trained on *naturally spoken*vowels at rates exceeding those of human listeners. Preliminary results will be reported for consonant recognition using SW nonsense syllables. [Work supported by NIH.]

10:40

3aSC8. A double-Nearey theory of vowel normalization: Approaching consensus. D. H. Whalen (Haskins Labs., 300 George St. Ste. 900, New Haven, CT 06511, whalen@haskins.yale.edu)

Terry Nearey has provided substantial data and insights into the realm of speech perception throughout his career. A major concern of his is to reconcile seemingly contradictory results. Here, his double-weak theory of perception and his probabilistic sliding template model (PSTM) are examined in relation to vowel normalization. Various algorithms achieve good results, but they must all confront the contradiction in behavior: We normalize the speech of others but we can also tell when they use a (slightly) different vowel than we do. Thus, listeners are (as in double-weak theory) fairly accommodating of different vocal tracts, but, contrarily, they can hear differences as well. PSTM may account for this difference, if deviations from theory-predicted vowel location are just those that speakers identity as being different; this remains to be tested. The various sources of vowel information (lower formants, F0, higher formants, spectral tilt)

interact in complex ways, as shown in Nearey's work and others'. No theory is comprehensive yet, as acknowledged by Nearey himself. He has also made the observation that our theories of speech perception ought to perform at least as well as automatic speech recognition systems. Although we have not reached that stage, Nearey's work brings us closer.

11:00

3aSC9. Toward a triple-weak theory of phonology/phonetics. Robert Port (Linguist, Cognit. Sci., Indiana Univ., 5975 S. Handy Rd., Bloomington, IN 47401, port@indiana.edu)

Almost 20 years ago, Nearey proposed the Double-Weak theory of speech production and perception. He pointed to three dominant theories of speech production and perception. All begin by assuming the memory representations of language consist of **phonemes** made from segmental distinctive features. He called the linguistic approach "double-strong" since it assumed each phoneme has a simple set of **articulatory correlates** as well as **acoustic correlates**. The other two theories assume either that the articulatory definitions are simple and strong while the acoustics is messy. Or that acoustics plays the strong role while articulation is messy. However, Nearey's double-weak theory assumed no simple relation of either articulation or acoustics to phonemes was possible. In contrast, my theory agrees with Nearey but further argues there is no fixed, invariant set of discrete, non-overlapping phonemes for spelling words (Port, *Lang Sci*, 2010). This presentation will also discuss recent attempts to salvage the phoneme noting they include so many caveats and revisions that they basically cede the issue.

11:20

3aSC10. Vowel dynamics and sentence processing. Diane Kewley-Port (Dept. of Speech and Hearing Sci., Indiana Univ., 5975 S. Handy Rd., Bloomington, IN 47401, kewley@indiana.edu) and Daniel Fogerty (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, Columbia, SC)

Throughout Terrance Nearey's influential academic career, research on the acoustic properties, models and theories of vowels has been a major focus. Studies by Nearey and his colleagues demonstrated the importance of spectral change (VISC) in vowel identification, even for monophthongal vowels. This presentation describes our research on vowel dynamics inspired by VISC research. Starting with syllables, the auditory system can discriminate F1 and F2 formants transition differences very well. Further results using high-fidelity resynthesis with sentences revealed that formant discrimination thresholds are 50% smaller than differences observed for natural syllables. These results support that human perception is very sensitive to spectral information in dynamically changing vowels in sentences. A second line of research examined the role of vowel information in sentences using noise interruption paradigms. Based on traditional segment boundaries, vowels had a two-to-one perceptual advantage over consonants for recognizing words in sentences. Shifting boundaries to favor consonantal transitions revealed a large advantage for even small proportions of vowel information. Recent research has enhanced understanding of the interdependence between vowel dynamics and suprasegmental contours of F0 and amplitude for accurate speech recognition. Thus, dynamic information in vowels, both VISC and suprasegmental, is essential for sentence intelligibility.

11:40

3aSC11. Double weak theory as a framework for discovery. Terrance M. Nearey (Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 0A2, Canada, tnearey@ualberta.ca)

A weak theory is not very attractive, a double-weak one even less so. The double-weak theory of speech perception serves mainly as a cover story for postponing rumination on the deeper principles that ultimately underlie human speech communication. Instead, it adopts minimal assumptions about the relation among symbols, sounds, and gestures that provides a rationale for immediately pursuing some hunches about empirical generalizations, or primitive "laws." It is hoped that some of these may be akin to Kepler's laws of planetary motion or eighteenth century "gas laws" that may serve as grist for later, more deeply explanatory theories. The active search for such empirical laws in speech perception starts from existing hypotheses about the general structure of surface phonological representations (e.g., features, phonemes, syllables) and proceeds by constructing detailed statistical models that accord with the fine structure of human responses in parametrically controlled speech perception experiments. Some examples and possible extensions of existing models are presented.

Session 3aSP

Signal Processing in Acoustics: Signal Processing in Nonlinear Acoustics

Brian E. Anderson, Cochair
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Yoshikazu Ohara, Cochair
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Invited Papers

8:00

3aSP1. Comparison of various methods of nonlinear signatures extraction. Alexander Sutin (Maritime Security Ctr., Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

Nonlinear Acoustic NDE (NA NDE) methods are very sensitive to defect presence and currently number on publications on this matter exceed several thousand. Despite intensive research of NA NDE, these methods are still do not widely used in practice and the goal of this presentation is to give review of various methods of nonlinear signatures extraction and to show advantages on some of them in the comparison with the others. The major part of the NA NDE was applied for characterization the tested part as whole. These methods include: High Harmonic Generation, Nonlinear Resonant Ultrasound Spectroscopy (NRUS), Amplitude Dependent Internal Friction, Nonlinear Wave Modulation Spectroscopy (NWMS) or Vibro Acoustic Modulation (VAM), Slow Dynamics Diagnostics (SDD), Nonlinear Reverberation Spectroscopy (NRS) or Nonlinear Impact Resonance Acoustic Spectroscopy (NRAS), Nonlinear Coda Wave Interferometry, Bi Spectral and High Order Statistic Analysis, Subharmonic and Ultraharmonic Methods. The methods that can provide nonlinear imaging and damage localization include: Nonlinear Time Reversal Acoustics (NTRA) or TR NEWS, Nonlinear Harmonic Imaging, Nonlinear Acoustic Tomography, Pulse Modification of Vibro Acoustic Modulation (VAM), Nonlinear Guided Wave Imaging and Tomography, Subharmonic and Ultraharmonic imaging, Nonlinear Structural Intensity.

8:20

3aSP2. Nonlinear Coda Wave Interferometry: Detecting, quantifying, and locating damage in complex solids. Vincent Tournat (LAUM, CNRS, Université du Maine, Av. O. Messiaen, Le Mans 72085, France, vincent.tournat@univ-lemans.fr), Odile Abraham (IFSTTAR, Nantes, France), Yuxiang Zhang (Université Joseph Fourier, Grenoble, France), Jean-Baptiste Legland, Benoit Hilloulin, Olivier Durand (IFSTTAR, Nantes, France), Stéphane Letourneur, Emmanuel Brasseur, and Mathieu Chekroun (LAUM, CNRS, Université du Maine, Le Mans, France)

In this talk, we report results on the nonlinear interactions of ultrasonic coda waves with lower pump waves, in reverberating or multiple scattering mesoscopic solid media. Using the method of coda wave interferometry (CWI), we analyze the effect of mixing a coda wave with an additional lower frequency pump wave. The extracted CWI parameters, known to be highly sensitive to small geometric or elastic modifications of the tested medium, are shown to be pump-amplitude dependent and to capture finely the results of the nonlinear interactions. Although nonlinear self-action effects with coda waves have been reported in unconsolidated granular media, they are difficult to implement in cracked solids or concrete. Instead, the reported nonlinear CWI class of methods (NCWI) shows robustness, a high sensitivity, and has been applied successfully to various complex media and structures. We show through several examples on « model » media (cracked glass plates) and on concrete structures, that NCWI can be useful for the nondestructive evaluation of complex solids that are strongly scattering at usual probing frequencies. Preliminary results and prospects in nonlinear elastic properties imaging and quantitative evaluation with NCWI are discussed.

8:40

3aSP3. Evaluation of crack parameters by a nonlinear frequency-mixing laser ultrasonics method. Sylvain Mezil (Faculty of Eng., Div. of Appl. Phys., Hokkaido Univ., N13W8, Kita-Ku, Sapporo 060-8628, Japan, sylvain.mezil@eng.hokudai.ac.jp), Nikolay Chigarev, Vincent Tournat, and Vitalyi Gusev (LAUM, Université du Maine, Le Mans Cedex 9, France)

Nonlinear acoustic methods are commonly used in crack detection because of their high sensitivity in comparison to linear ones. However, the dependence of the nonlinearities on the crack state can not only localize it but also provides information on its characteristics. We present a laser ultrasonic method, based on nonlinear frequency-mixing, to locally evaluate several crack parameters, including some, like the local crack elasticity, which are assessed uniquely by the present technique. Two laser beams, independently intensity

3a WED. AM

modulated at two cyclic frequencies ω_H and $\omega_L(\omega_H>>\omega_L)$, excite the sample. For a large sinusoidal thermo-elastic stress generated by the low-frequency modulated laser beam, the crack oscillates between closed and open states. In the presence of this variation, nonlinear frequency-mixing ultrasonic components at frequencies $\omega_H \pm n\omega_L$ (n is an integer) are detected. By modifying the intensity of the laser beam modulated at ω_L, we can influence the time spent in opened and closed state over a period ($2\pi/\omega_L$). The developed theoretical model demonstrates the dependence of the nonlinear components at $\omega_H \pm n\omega_L$ on the time spent by the crack in each state. Comparison between theoretical and experimental results offers a way to characterize some local crack properties, including its width and effective rigidity.

9:00

3aSP4. Quantitative evaluation of strength degradation by using linear-nonlinear ultrasonic techniques. Kyung-Young Jhang (School of Mech. Eng., Hanyang Univ., 204 Eng. Ctr. Annex, 222 Wangsimni-ro, Seongdong-gu, Seoul 04736, South Korea, kyjhang@hanyang.ac.kr), Jongbeom Kim, and Ju-ho Lee (Dept. of Mech. Convergence Eng., Hanyang Univ., Seoul, South Korea)

The ultrasonic nonlinear parameter, β, has been known as effective for the evaluation of material degradation, and its correlation with material degradations such as thermal aging, fatigue, creep, and plastic deformation has been reported. However, most studies were limited to the relative measurements that is effective only for the relative comparison and not able to evaluate the degradation in quantitative. To overcome this limit, this study proposes a new algorithm which is able to evaluate the quantitative strength degradation. The proposed method mainly consists of four steps: 1) Measure the linear elastic modulus (E) from the longitudinal and shear wave velocities by using pulse echo method. 2) Measure the absolute ultrasonic nonlinear parameter, β, by using piezoelectric detection method. 3) Construct the stress-strain curve by substituting the measured linear elastic modulus (E) and absolute ultrasonic nonlinear parameter β to the nonlinear stress-strain equation. 4) Estimate the 0.01% offset yield strain and yield strength. The yield strengths estimated by the proposed technique in the heat-treated Al6061-T6 and SA508 showed good agreement with the yield strength obtained by tensile test. Consequently, the proposed method makes the quantitative evaluation of strength degradation using ultrasonic measurement possible. [This research was supported by the National Research Foundation of Korea (NRF) grant funded by the Korean government (NRF-2013M2A2A9043241).]

9:20

3aSP5. Retreiving global nonclassical nonlinearity parameters from local measurements. Martin Lott (Waves and Imaging, CNRS-LMA, Marseille, France), Marcel Remillieux (Geophys., Los Alamos National Lab., Los Alamos, NM), Pierre-Yves Le Bas, Timothy J. Ulrich (Detonator Phys., Los Alamos National Lab., Los Alamos National Lab., Geophys. Group, MS D446, Los Alamos, NM 87545, tju@lanl.gov), Vincent Garnier, and Cedric Payan (Waves and Imaging, CNRS-LMA, Marseille, France)

We demonstrate the equivalence between local and global measures of nonclassical nonlinear elasticity in a slender resonant bar of Berea sandstone. Nonlinear effects are first measured globally using Nonlinear Resonance Ultrasound Spectroscopy (NRUS), which monitors the relative shift of the resonance frequency as a function of the maximum dynamic strain in the sample. Subsequently, nonlinear effects are measured locally at various positions along the sample using Dynamic Acousto-Elasticity Testing (DAET). After processing the DAET signals and correcting for three-dimensional strain effects, it is shown that by numerically integrating these corrected data along the length of the sample the NRUS global measures are retrieved almost exactly.

9:40–10:00 Break

10:00

3aSP6. Nonlinear ultrasonic phased array for closed crack imaging. Yoshikazu Ohara, Kazushi Yamanaka, and Tsuyoshi Mihara (Dept. of Mater. Processing, Tohoku Univ., 6-6-02 Aoba, Aramaki-aza, Aoba-ku, Sendai 980-8579, Japan, ohara@material.tohoku.ac.jp)

In nondestructive ultrasonic inspection, closed cracks can lead to the underestimation or overlook, resulting in the catastrophic accidents. To solve this problem, nonlinear ultrasonics has been expected as a most promising approach. In nonlinear ultrasonics, by irradiating large amplitude, the nonlinear interaction between closed cracks and ultrasonics generates nonlinear components. Among them, we found that subharmonics has a higher selectivity than the others including higher harmonics. We also found that the subharmonics has higher temporal resolution, providing the high spatial resolution in ultrasonic imaging. Based on these findings, we have developed a novel imaging method, subharmonic phased array for crack evaluation (SPACE), based on the subharmonic generation at closed cracks due to large-amplitude short burst waves input and phased array imaging with frequency filtering. As a result of applying SPACE to closed-crack specimens, SPACE visualized the closed part of cracks, which were not observed in the conventional methods. It was substantiated that SPACE is useful in accurately measuring closed-crack depths.

10:20

3aSP7. How to improve detectability of crack size in nonlinear sub-harmonic imaging. Choon-Su Park, Seunghyun Cho, Dae-Chul Seo (Ctr. for Safety Measurements, Korea Res. Inst. of Standards and Sci., Gajeong-ro 267, Bld. # 206 / Rm. # 206, Daejeon 34113, South Korea, choonsu.park@kriss.re.kr), and Jun-Woo Kim (Safety Qualification Technol. Ctr., KHNP Central Res. Inst., Daejeon, South Korea)

Crack growth is regarded as an important monitoring parameter for structural degradation. To precisely estimate crack size, therefore, has been highly demanded to monitor structural healthiness and to assess residual life of various structures from small components to huge plants. Conventional non-destructive inspections have been successfully detected cracks, but some deficiencies for accuracy still remain. Nonlinear sub-harmonic generation was well proved to detect closed cracks that could cause underestimation of crack size. Moreover, sub-harmonic phased array (PA) imaging was proposed to visualize where the closed cracks are. The sub-harmonic PA images, however, has lower spatial resolution than the fundamental frequency PA images due to its longer wavelength, which often prevents from clearly observing closed cracks. Many of signal processing techniques have been developed to improve spatial resolution of

images for more than half a century, and some resolution enhancing techniques such as deconvolutions and an eigen-analysis based technique are employed to make the resolution of sub-harmonic imaging better. Point spread function issues for deconvolution have been also investigated theoretically and experimentally. In addition, some experiments with CT specimens have been done to prove closed-crack localization by improving spatial resolution of sub-harmonic imaging.

10:40

3aSP8. Extraction of nonlinear elastic material parameters from single-impact nonlinear ring-down spectroscopy. Parisa Shokouhi, Jiang Jin, and Jacques Riviere (Civil and Environ. Eng., Penn State, 215 Sackett Bldg., University Park, PA 16802, parisa@engr.psu.edu)

Volumetric microcracking is one of the early symptoms of distress in cementitious materials caused by excessive mechanical stress, chemical attacks, and environmental influences. The microcracks widen, coalesce, and develop into larger cracks with the progress of damage. Visible macro-cracks indicate severe damage that often cannot be mitigated. As such, detection of damage at the early stages of development is essential for designing optimal preventive maintenance programs for concrete structures. Nonlinear acoustics-based nondestructive testing techniques have shown great promise in identification of microscopic cracks in diverse materials including concrete. Impact-based alternatives of conventional techniques are gaining popularity for concrete testing mainly due to their field transportability. In this study, we focus on impact-based nonlinear resonance acoustic spectroscopy (INRAS). We compare the results from multi-impact INRAS, where several impacts of increasing intensities are applied, to those from single impact ring-down spectroscopy where only one impact of large intensity is used. Hilbert Huang Transform (HHT) is used to obtain the time-dependent frequency content of the single-impact ring-down. We propose several models for fitting the entire ring-down in order to extract nonlinear elastic material parameters. We demonstrate that our proposed approach for analyzing single-impact INRAS data yields material parameters compatible with those obtained from conventional testing and analysis.

Contributed Papers

11:00

3aSP9. Nonlinear scattering of crossed focused ultrasonic beams in a constricted flow for the detection of a thrombosis. Emily S. Kilen, Theodore R. Johnson, Daniel J. Fisher, and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, korman@usna.edu)

Experiments performed at the USNA Acoustics Lab use turbulent flow to generate nonlinear scattering of mutually perpendicular crossed beams ($f_1 = 1.9$ MHz, $f_2 = 2.1$ MHz, 15 cm focal lengths) at the combination sum frequency component ($f_+ = 4$ MHz). The object is to simulate blood flow through a constriction, and measure the nonlinear scattering downstream with a receiving circular plane array transducer (4 MHz) that rotates azimuthally in the plane of the transmitted beam axes about the intersection. Angular dependent Doppler shift (about f_+), spectral broadening, skewness and kurtosis measurements vs. angle help predict the characteristics of the turbulence. The setup involves a 4 cm diameter polyethylene (0.5 mm thick) tube cut in two sections (22 cm upstream (a), 38 cm downstream (b)). The sections are mated with a special block union allowing one to insert individual thin orifice plates of different diameters. The far section ends are open but are connected through opposite side ports on (a) and likewise on (b) to in-phase and out of phase synchronized bellows pumps. The apparatus when submerged in a water tank generates pulsating turbulent flow. Nonlinear scattering is measured vs. orifice size. [See S. M. Mock (2013) and D. J. Fisher (2014), USNA Capstones.]

11:15

3aSP10. Characterizing nonlinear systems with memory while combating and reducing the curse of dimensionality using new volterra expansion technique. Albert H. Nuttall (NUWCDIVNPT (retired), Old Lyme, CT), Derke Hughes, Richard A. Katz, and Robert M. Koch (NUWCDIVNPT, 1176 Howell St., Newport, RI 02841, derke.hughes@verizon.net)

A generalized model for characterizing nonlinear systems was originally proposed by Italian mathematician and physicist Vito Volterra (1860-1940). A further development by American mathematician and MIT Professor Norbert Wiener (1894-1964) was published in 1958. After direct involvement with Norbert Wiener publication, Albert H. Nuttall has recently made new inroads along with his coauthors in applying the Wiener-Volterra model. A general description of a nonlinear system to the third order is termed the Nuttall-Wiener-Volterra model (NWV) after its co-founders. In this formulation, two measurement waveforms on the system are required in order to characterize a specified nonlinear system under consideration: an excitation input, x(t) (the transmitted signal) and a response output, z(t) (the received signal). Given these two measurement waveforms for a given system, a kernel response, h= [h0,h1,h2,h3] between the two measurement points, is computed via a least squares approach that optimizes modeled kernel values by performing a best fit between measured response z(t) and a modeled response y(t). New procedures developed by A. Nuttall are invoked to significantly diminish the exponential growth of the computed number of kernel coefficients with respect to third order and higher orders to combat and reasonably reduce the curse of dimensionality.

11:30

3aSP11. Superdirective non-linear beamforming with deep neural network. Mitsunori Mizumachi (Dept. of Elec. Eng. and Electronics, Kyushu Inst. of Technol., 1-1 Sensui-cho, Tobata-ku, Kitakyushu 805-8440, Japan, mizumach@ecs.kyutech.ac.jp) and Maya Origuchi (Dept. of Elec. Eng. and Electronics, Kyushu Inst. of Technol., Kitakyushu, Fukuoka, Japan)

Beamforming has been one of important issues in acoustic signal processing, since it can achieve signal enhancement and sound source localization. In general, traditional beamformers are designed by an analytical approach or an adaptive approach. It is, however, difficult to properly optimize the beamformers under the complicated acoustical scene. An alternative non-linear beamforming can be substituted for the linear beamforming. In this study, a flexible framework for optimizing the beamformer is introduced based on a deep neural network. Capturing acoustic signals using a microphone array is regarded as spatial sampling, so that annoying grating lobes appear in beam-pattern when the relationship between the wavelength and the microphone spacing does not satisfy the sampling theorem. The proposed method achieves sub-band beamforming using the non-uniform microphone array with eight nesting microphones, which are carefully designed not to cause spatial aliasing. Feasibility of the proposed method has been confirmed by computer simulation. The proposed non-linear beamformer could successfully achieve superdirectivity compared with conventional beamformers.

11:45

3aSP12. Defect detection in a concrete bridge by a non-destructive technique. Masato Abe, Toyota Fujioka, and Yoshifumi Nagata (Comput. and Information Sci., Iwate Univ., 4-3-5 Ueda, Morioka 020-8551, Japan, abe@cis.iwate-u.ac.jp)

Post-tensioned prestressed concrete is a method for overcoming concrete's natural weakness in tension, and it is used widely to produce bridges with a longer span. Because of poor workmanship or quality control during

construction, however, sometimes the ducts containing the prestressing tendons are not fully filled, leaving voids in the grout where the steel is not protected from corrosion. Therefore, we propose a method to detect such the voids using multiple sensors and an impulse hammer. The sensors pick up the vibration wave caused by the impulse hammer. The experimentally picked-up vibration waves are compared with those simulated by the Finite-Difference Time Domain (FDTD) method, and it is found that these two waves quite similar. It is also found from some experiments for the bridge pier that we can detect (1) the presence or absence of a void and (2) the void position.

WEDNESDAY MORNING, 30 NOVEMBER 2016 NAUTILUS, 8:15 A.M. TO 10:05 A.M.

Session 3aUWa

Underwater Acoustics, Noise and Acoustical Oceanography: Transmission Through the Air-Water Interface

Peter H. Dahl, Cochair
Applied Physics Laboratory, University of Washington, Mechanical Engineering, 1013 NE 40th St., Seattle, WA 98105

David R. Dall'Osto, Cochair
Acoustics, Applied Physics Laboratory at University of Washington, 1013 N 40th St., Seattle, WA 98105

Invited Papers

8:15

3aUWa1. Complex acoustic intensity transmission through the air-water interface. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Mech. Eng., and Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Sound transmission through the air-water interface occurs when there is a non-zero normal component of active and reactive acoustic intensity. Transmission occurs for both active intensity (pressure and particle velocity in phase) and reactive intensity (pressure and particle velocity out of phase), and depending on the proximity of the source to the surface these can convert from one form to the other. Intensity transmission through the interface is governed Chapman's Law—which differs slightly from Snell's law but is consistent with the requirement of phase continuity along the surface. Properties of intensity transmission through the air-water interface are examined using three examples: (1) Marine pile driving, where sound generated by the in-air portion of the pile is injected into the water-column, and sound generated underwater leaks into the air. (2) Transmission of far-field active and reactive intensity generated by aircraft, and how Chapman's law or Snell's law describe Doppler shifts. (3) The ability of a fish-catching bat to locate fish underwater with sonar, and how transmitted active intensity resonates with the swim-bladder of the fish generating a near-field that radiates into the air as active intensity. [Research supported by ONR.]

8:35

3aUWa2. The role of diffraction in sound transmission through air-water interfaces. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu)

Ray-theoretical modeling and elementary considerations that are based on plane-wave reflection coefficients predict very weak sound transmission through air-water interfaces because of a large contrast in the acoustic impedances of air and water. With ray-type geometric contributions to transmitted acoustic energy being suppressed by the disparities in the sound speeds and densities, wave diffraction often plays the dominant role in coupling acoustic fields in air and water. In particular, acoustic diffraction leads to the phenomenon of anomalous transparency of air-water interfaces for low-frequency sound. Almost all the energy that is radiated by shallow compact underwater sound sources is transmitted into air [O. A. Godin, Phys. Rev. Lett. **97**, 164301 (2006)]. Other scenarios, in which diffraction plays a critical role in sound transmission through air-water interfaces, include excitation of Scholte-type surface waves in water by sonic booms and propagation of Lamb waves along the ocean surface. The paper will review the theory and experimental evidence of the anomalous transparency of air-water interfaces for finite-size underwater sound sources and for coupling of acoustic fields in the ocean and atmosphere by Lamb and Scholte-type surface waves.

8:55

3aUWa3. Compiling the air situation picture from a submerged submarine. Brian G. Ferguson and Gary C. Speechley (Maritime Div., DSTG, PO Box 44, Pyrmont, NSW 2009, Australia, Brian.Ferguson@dsto.defence.gov.au)

The transmission of sound through the air-water interface produces an underwater sound field that can have as many as four separate contributions from an airborne acoustic source. These contributions are identified as: direct refraction, one or more seafloor reflections, the evanescent wave, and sound scattered from a rough sea surface. The relative importance of each contribution depends on the horizontal distance of the source from the receiver, the water depth, the depth of the receiver in relation to the wavelength of the signal radiated by the source, and the roughness of the sea surface. This paper demonstrates how a submerged submarine towing an array of hydrophones is able to detect, classify, localize and track maritime patrol aircraft and helicopters, whilst at depth, by exploiting the direct and indirect sound propagation paths of the aircraft's radiated noise (acoustic signature). As the airborne source is in relative motion with respect to the array, the received signal at each hydrophone is Doppler-shifted in frequency. It is shown that the variation with angle of arrival of the observed Doppler-shifted propeller blade passing frequency is in close agreement with that predicted for the direct refraction path. Similar observations are also made for the seafloor reflected propagation paths. Early warning/long range detection of threat aircraft is enabled by one or more reflections from the sea floor.

9:15

3aUWa4. Light aircraft sound for underwater acoustics experiments. Michael J. Buckingham (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

A propeller-driven light aircraft produces sound in the form of a sequence of harmonics from the propeller and from the reciprocating engine. The lowest frequency, from a two-bladed propeller, is around 80 Hz with detectable harmonics extending up to about 1 kHz. When flying over the shallow ocean at low level, some of the aircraft sound penetrates the air-sea interface to undergo multiple reflections between the seabed and the sea surface as it propagates through the channel. On approach to a sub-surface receiver station, the frequency of a given harmonic is Doppler upshifted, whereas on departure it is downshifted. Since these Doppler shifts depend not only on the speed of the aircraft but also on the geo-acoustic properties of the seabed, they provide the basis of a rapid and efficient inversion technique for surveying the seabed. To develop the technique, shallow-water experiments have been performed using various types of light aircraft as the sound source and a single hydrophone as the receiver. It was found that, with the aid of the grain-shearing theory of wave propagation in unconsolidated sediments, it is possible to recover the sediment geo-acoustic parameters from the Doppler-shifted aircraft harmonics. [Research supported by ONR.]

Contributed Papers

9:35

3aUWa5. A geoacoustic inversion technique using the low-frequency sound from the main rotor of a Robinson R44 helicopter. Dieter A. Bevans, Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, dbevans@ucsd.edu), and Paul Hursky (Heat, Light, & Sound Res. Inc., San Diego, CA)

A series of underwater acoustic experiments utilizing a Robinson R44 helicopter and an underwater receiver station has been conducted in shallow (16.5 m) water. The receiver station consisted of an 11-element nested hydrophone array with a 12 m aperture configured as a horizontal line (HLA) 0.5 m above the seabed. An in-air microphone was located immediately above the surface. The main rotor blades of the helicopter produce low-frequency harmonics, the fundamental frequency being ~13 Hz. The tail rotor produces a sequence of harmonics approximately six times higher in frequency. The first experiment characterized the underwater sound signature of the helicopter with altitude and range. Using analytical and numerical 3-layer (atmosphere-ocean-sediment) acoustic propagation models a sediment geoacoustic inversion technique has been developed. This technique, requiring only knowledge of the relative location of the sensors and sound source (helicopter), uses the cross-correlation between HLA sensor pairs to produce the estimated time delay of the head wave. The results from the simulations and the latest experiment are presented. [Research supported by ONR, SMART(DOD), NAVAIR, and SIO.]

9:50

3aUWa6. Model validation of rough surface scattering at a water/air interface using a hybrid parabolic equation model. Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Rm. 114, Monterey, CA 93943, kbsmith@nps.edu), Mustafa Aslan (Turkish Naval Acad., Istanbul, Turkey), and Geoffrey Moss (Naval Undersea Warfare Ctr. Div. Newport, Newport, RI)

Traditionally, ocean acoustic propagation models assume the sea surface can be treated as an idealized pressure release boundary. For flat surfaces, this can easily be accomplished through a variety of modeling techniques. Rough surfaces, however, introduce additional complexities in numerical models. For propagation models based on the parabolic equation that utilize split-step Fourier (SSF) algorithms, previous work has involved field transformational techniques to treat the rough surface displacements. Such techniques assume small angle scattering at the interface, which may not produce adequate numerical accuracy. An alternative approach is to model the physical water/air interface, and allow the higher order propagator functions of the parabolic approximation to more accurately model the rough surface scatter. However, the introduction of such large interface discontinuities have been known to introduce phase errors in SSF-based models. In this work, a previously developed hybrid split-step Fourier/finite-difference approach is implemented at the water/air interface. Results are compared with standard SSF smoothing approaches, as well as the pressure release field transformational technique, for simple rough surfaces. A finite element model is utilized to provide a benchmark solution and comparisons are made for both standard Dirichlet and explicit mixed media treatments of the air-water interface. Tradeoffs between accuracy and stability are discussed, as well as transmission across the water/air interface.

Session 3aUWb

Underwater Acoustics: Topics in Underwater Acoustics (Poster Session)

Aubrey L. Espana, Chair
Acoustics Dept., Applied Physics Lab. - Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow all contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and authors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

3aUWb1. Seabed target discrimination using multistatic acoustic scattering data. Erin M. Fischell and Henrik Schmidt (Mech. Eng., MIT, 77 Massachusetts Ave., 5-204, Cambridge, MA 02139, emf43@mit.edu)

One application for autonomous underwater vehicles (AUVs) is detecting and classifying hazardous objects on the seabed. We have been studying an alternative acoustic approach to this problem in which an acoustic source insonifies seabed targets while receiving AUVs discriminate targets based on features sensed in the resulting 3D scattered fields. The receiving AUVs have entirely passive sensing payloads, and therefore lower power draw and cost. When both the acoustic source and receiver are mobile, multistatic scattering data are collected. Using the OASES-SCATT scattering simulator, we studied how multistatic scattering data collected by AUV-based receivers around targets insonified by AUV-based sources might be used for sphere and cylinder target characterization in terms of target shape, composition, and size. The impact of target geometry on multistatic scattering fields is explored, and a target discrimination approach developed in which the acoustic source and receiver to circle the target with the same radial speed, collecting multistatic scattering data at constant bistatic angles. The frequency components of the multistatic scattering data at different bistatic angles are used to form probabilistic and machine learning models for target characteristics. New data are then classified using these models. [Work supported by ONR and Battelle.]

3aUWb2. A spectral approach to geoacoustic inversion from sparse impulsive sources. Thomas W. Yudichak, Steven A. Stotts, Robert A. Koch, Dan G. Jacobellis, and Daniil Ruditskiy (Appl. Res. Laboratories, The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, yudi@arlut.utexas.edu)

A common approach to geoacoustic inversion is to match the temporal structure displayed by modeled and measured acoustic data produced at a small number of frequencies by range sampling source emissions over a broad interval between a source and a receiver. An alternative approach is to match spectral structure in modeled and measured acoustic data for a small number of source-receiver separations from a fine frequency sample over a broad bandwidth. The latter approach is demonstrated with Shallow Water '06 experiment data in which the acoustic sources were imploding light bulbs. Geoacoustic parameter values obtained by this approach are compared to values obtained from previous inversions, and the limits of validity of the approach are discussed.

3aUWb3. Reciprocity calibration of underwater acoustic transducer in a reverberation pool. Dajing Shang, Qi Li, and Jundong Sun (Underwater Acoust. Eng. Dept., Harbin Eng. Univ., Nangang District Nantong St., No.145,Harbin City, Heilongjiang Province 150001, China, shangdajing@hrbeu.edu.cn)

A new underwater acoustic transducer reciprocity calibration method was put forward in this paper. This calibration was finished by the space-averaging in the reverberation pool, and the influence of normal mode interference was eliminated, so the reverberation field acoustic parameters were obtained in the reverberation pool. The calibration can be set up in a small reverberation pool. In the case of the same frequency range acoustic transducer calibration, the reverberation pool for reverberation method reciprocity calibration can be smaller than the free-field pool for free-field reciprocity calibration. The reverberation field reciprocity constant was determined by measuring the reverberation time, the single-frequency sine signal was emission by the sound source higher than the cutoff frequency, the transmitting transducer and receiving transducer was space-averaged in the area of the reverberation, and the corresponding electrical parameters were obtained, so the receiving sensitivity for the reciprocity transducer and the testing hydrophone was calculated. The testing and the results of the reverberation calibration are presented in this paper. The error between the hydrophone calibration by the reverberation method and the factory free field calibration was less than 1.4 dB, and the A kind of measuring uncertainty is less than 0.13 dB. This method can be applied to all kinds of hydrophones and transducer calibration.

3aUWb4. Modeling and simulation technique for development of multibeam echo sounder. Dong Hwan Jung, Jea Soo Kim, and Gi Hoon Byun (Ocean Sci and Technol., Korea Maritime and Ocean Univ., 727 Taejong-ro, Yeongdo-Gu, 253, Busan 49112, South Korea, ehdghkss104@naver.com)

Multibeam echo sounder (MBES) is commonly used for rapid sea floor mapping. We present a time-domain integrated system simulation technique for MBES development. The simulation and modeling (M&S) modules consist of four parts: sensor array signal transmission, propagation and backscattering modeling in the ocean environment, beamforming of the received signals, and image processing. Also, the simulation employs a ray-theory-based algorithm to correct the reconstructed bathymetry, which has errors due to the refraction caused by the vertical sound velocity profile. The developed M&S technique enables design parameter verification and system parameter optimization for MBES. The framework of this technique can also be potentially used to characterize the seabed properties. Finally, typical sea floor images are presented and discussed.

3aUWb5. Prediction of array gain in directional noise field. Jisung Park, Yonghwa Choi, Jeasoo Kim (Ocean Eng., Korea Maritime and Ocean Univ., 727 Taejong-ro, Yeongdo-Gu, Busan 49112, South Korea, pjs840627@gmail.com), Sungho Cho (Maritime Security Res. Ctr., Korea Inst. of Ocean Sci. & Technol., Ansan, South Korea), and Jungsoo Park (Agency for Defense Development, Jinhae, South Korea)

The Array Gain(AG) is a metric to assess the performance of an array and is dependent on the configuration of array, frequency, as well as on the directionality of noise. In this study, AG is calculated based on the spatial coherence between sensor elements in directional noise environment for a given array shape. The estimated AG is then compared with AG derived from the sea going data based on the signal to noise ratio. The results are presented and discussed.

3aUWb6. Research for the position detection of the object within the tube liquid using sound wave. Seijun Muto, Katsumi Fukuda (National Inst. of Technol., Tokyo College, School, Hachioji, Kunugidacyo 1220-2, Tokyo, Japan, seijunsu47@yahoo.co.jp), and Yoshihiro Nishimura (National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Ibaraki, Japan)

The purpose of detecting using sound waves the position of an object in the tube filled with liquid. Experiments were carried out taking into account the frequency and oscillation of the process of the sound. As a result, good results for the position detection of the object to obtain a new knowledge.

3aUWb7. The noise of rock n'roll: Incidental noise characterization of underwater rock placement. Rute Portugal, Sei-Him Cheong (Gardline Environ., Gardline Geosurvey, Ltd., Gardline Environ., Endeavour House, Admiralty Rd., Great Yarmouth NR30 3NG, United Kingdom, rute.portugal@gardline.com), James Brocklehurst (Royal Boskalis Westminster N.V., Papendrecht, Netherlands), and Breanna Evans (Gardline Environ., Gardline Geosurvey Ltd., Great Yarmouth, United Kingdom)

Underwater noise is a growing concern to conservation and stock management efforts to which supra-national organizations (e.g., OSPAR or the European Union) and governments (e.g., USA) are beginning to respond by building catalogues of the noise introduced in the marine environment by human activity. Rock placement is a construction activity for which there is scarcely any data available. In order to fill the knowledge gap, opportunistic recordings were taken while the Gardline Mk 3 hydrophone array was deployed for Passive Acoustic Monitoring and mitigation for marine mammals. The recordings were analysed for their spectral and temporal characteristics, a correlation analysis between the amount of rock placed and the intensity of sound produced was made and the suitability of the hydrophone array for the collection of this type of data was assessed.

3aUWb8. Sound exposure level and energy spectral density of underwater explosions in shallow water over a coral substrate off the southern coast of O'ahu, Hawai'i. Alexander G. Soloway (Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, soloway@u.washington.edu), Peter H. Dahl (Mech. Eng. and Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Lee H. Shannon (Naval Facilities Eng. and Expeditionary Warfare Ctr., Pearl Harbor, HI)

This work presents the sound exposure levels (SEL) and energy spectral densities (ESD) from underwater explosions measured in shallow water (10-18 m) at distances of 500 to 1500 m at the Pu'uloa Underwater Detonation Range off the southern coast of Oahu. Nine explosive charges, with TNT-equivalent weights of 2.2 to 8.6 kg, were detonated on a seabed characterized by a thin sand layer over limestone. The ESD of the measurements are characterized by high propagation loss in the frequency ranges 50 to 350 Hz. Previous studies have shown that this is a common characteristic of this environment with the upper and lower frequencies directly related to the geoacoustic properties of the seabed and the waveguide geometry. Unlike measurements collected in sandy environments, where SEL agreed with empirical predictions, the SEL for these measurements differ by up to 30 dB. To understand the mechanisms responsible for these lower than expected levels, a geoacoustic model for the seabed is developed using these frequencies and utilized in broadband modelling simulations. The SEL will also be calculated for these simulations and compared to the measured levels. [Research supported by Pacific Fleet with partial support from the Office of Naval Research.]

3aUWb9. Point measurements of ambient biological noise before, during, and after multiple underwater detonation events over coral substrate off the southern coast of Oahu, Hawai. Dara M. Farrell (Appl. Phys. Lab., Mech. Eng., Univ. of Washington, Henderson Hall, Seattle, WA 98105, daraf@uw.edu), Lee H. Shannon (Marine Resources Assessment Diving Services, Naval Facilities Eng. and Expeditionary Warfare Ctr., Pearl Harbor, Hawaii), Peter H. Dahl (Mech. Eng., Univ. of Washington, Appl. Phys. Lab, Seattle, WA), and David R. Dall'Osto (Appl. Phys. Lab, Seattle, WA)

Measurements were taken of background noise at the U.S. Navy's Pu'uloa Underwater Detonation Range off the south coast of O'ahu during a U.S. Navy underwater explosive training exercise in shallow water (10-18 m) where distance from the detonation site, charge size, and explosive composition were controlled. The autonomous recording unit was deployed approximately 1500 m from the detonation location. The combined data from the paired low (-220 dB re 1 V /μPa) and high (-170 dB re 1 V /μPa) sensitivity hydrophones gives an approximately 100 dB dynamic range view of each event. Each trial consisted of three replicate events per charge weight (2.2, 4.5, and 8.6 kg) for a total of nine explosive events over the coral substrate. An analysis of the background noise environment immediately before and after each replicate is presented using metrics such as spectral probability density. Results are considered in the context of typical coral reef noise including discussion of snapping shrimp noise (one of the dominant features of the soundscape) before and after each event. [Work supported by U.S. Pacific Fleet.]

3aUWb10. Stochasticism in noise generated by an array of marine hydrokinetic devices. Erin C. Hafla (Civil Eng., Montana State Univ., 205 Cobleigh Hall, Bozeman, MT 59717-3900, erinhafla@gmail.com), Erick Johnson (Mech. Eng., Montana State Univ., Bozeman, MT), and Jesse Roberts (Energy & Climate, Sandia National Labs., Albuquerque, NM)

Marine hydrokinetic (MHK) devices generate electricity from the motion of tidal and ocean currents and ocean waves and provide another source of renewable energy. Additionally, MHK devices are also a new source of anthropogenic noise in the marine ecosystem and must meet regulatory guidelines that mandate a maximum amount of noise that may be generated. In the absence of measured levels from in-situ deployments, a model for predicting the propagation of sound from an array of MHK sources in a real environment needs to be established. A 3D finite-difference, time-domain solution to the governing velocity-pressure equations is used, which permits a finite number of complex sources and spatially varying sound speeds, bathymetry, and bed composition. However, deterministic solutions to these types of problems cannot capture uncertainties in the source profiles that may result from operational changes. This work presents the broadband sound pressure levels from an array of MHK sources as the amplitude and frequency from each source are allowed to vary. This Monte Carlo approach demonstrates that the idealized, deterministic solution, vastly underestimates the compounding uncertainty on the final sound field and that these small variations in the source profile cannot be ignored.

3aUWb11. Acoustic characterization of a wave energy converter. Brian L. Polagye, Paul Murphy (Mech. Eng., Univ. of Washington, Box 352600, Seattle, WA 98195-2600, bpolagye@u.washington.edu), Keith Bethune, Patrick Cross, and Luis Vega (Hawai'i Natural Energy Inst., Univ. of Hawai'i, Manoa, HI)

As progress toward the commercial deployment of wave energy converters accelerates, it is important to ensure that these renewable energy systems do not have unintended, adverse environmental consequences. While the sound from wave energy converters is unlikely to cause acoustic injury to marine animals, it may affect their behavior. Here, we present measurements from a point-absorber wave energy converter at the U.S. Navy Wave Energy Test Site in Kaneohe Bay, HI. Measurements of wave converter sound are obtained for a range of sea states using a combination of free-drifting near-surface measurements and stationary bottom packages. The relative effectiveness of these systems are contrasted and the unique challenges associated with acoustic measurements at energetic sites discussed. For example, fixed measurements are found to be substantially contaminated by flow-noise (non-propagating sound) during long-period ocean

swell, while free-drifting measurements require significant post-processing to avoid convolving flow-noise or self-noise with wave converter sound. Preliminary results of parabolic equation modeling is also presented and used to interpret spatially distributed measurements.

3aUWb12. Acoustic ground truthing of airgun noise in Chatham Rise, New Zealand. Sei-Him Cheong (Marine Wildlife, Gardline Geosurvey, Endeavour House, Admiralty Rd., Great Yarmouth, Norfolk NR30 3NG, United Kingdom, sei-him.cheong@gardline.com)

Underwater noise is a growing concern to conservation of the marine life. Noise generated by seismic survey in particular is recognised as a significant and pervasive pollutant, with the potential of impacting the marine ecosystem. Between the 1st February and 21st March 2016, a geophysical research survey was conducted in Chatham Rise, New Zealand, to collect seismo-acoustic data using a Sercel Sentinel seismic streamer in order to ground truth the underwater noise impact assessment, conducted according to the Department of Conservation (New Zealand) 2013 Code of Conduct. Data were analyzed in order to determine the received sound level at a distance up to 3 km from the seismic source array. This paper establishes the method to predict the mitigation impact radii from seismic data in order to validate the results obtained based on predictive noise modelling using Gardline 360M model. The study was also aimed to provide confidence to the capability of predictive modeling for estimating the acoustic impact zone of a seismic sound source. Data showed that seabed reflection can fluctuate significantly according to sediment topography and other environmental influences; however, a very consistent trend can be obtained from direct propagation to confidently establish mitigation radii. Results show that the employment of a seismic streamer for the establishment of effective mitigation radii is technically feasible and may be used as a tool to ground truth predictive modelling as part of a marine mammal mitigation plan.

WEDNESDAY MORNING, 30 NOVEMBER 2016 — NAUTILUS, 10:30 A.M. TO 11:45 A.M.

Session 3aUWc

Underwater Acoustics: Inversion, Beam-Forming, and Calibration I

Steven A. Stotts, Chair

Environmental Sciences Laboratory, Applied Research Labs/The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78759

Contributed Papers

10:30

3aUWc1. Impact of array tilting on source-range estimation based on the array/waveguide invariant. Chomgun Cho and Hee-Chun Song (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, chomgun@ucsd.edu)

Recently, the array/waveguide invariant was proposed for robust range estimation of a broadband source in a waveguide environment using a short-aperture vertical array (VA). The approach involves conventional plane-wave beamforming and exploits the separation of arrivals in beam angle and travel time. The beam angle estimate, however, can be sensitive to array tilting. In this paper, we investigate the impact of array tilting on source-range estimation based on the array/waveguide invariant. Analysis of experimental data indicates that even a small tilt angle ($<2°$) of a 1.2-m long VA can result in a relative range error of 20% or more for a source (9-17 kHz) at 3-km range in ~100-m deep water. Inversely, the array tilt angle can be estimated for a known source range.

10:45

3aUWc2. Direction-of-arrival estimation using a three-dimensional cross array equipped underwater glider. Yong-Min Jiang (Res. Dept., NATO-STO-Ctr. for Maritime Res. & Experimentation, Viale San Bartolomeo 400, La Spezia 19126, Italy, yong-min.jiang@cmre.nato.int)

With the development of autonomous platform and sensor technologies, unmanned underwater vehicles have been more and more involved in maritime intelligent surveillance and reconnaissance missions. The NATO—STO—CMRE has been exploring the capabilities of gliders for monitoring underwater environment by means of passive acoustic sensing. During the CMRE GLISTEN'15 sea trial, an eight-element, three dimensional cross array equipped Webb Slocum glider was deployed to study its capability of providing the direction of arrival (DOA) estimation of the signal of interest. Continuous wave pulses at multiple frequencies (300—1000 Hz) were transmitted by an acoustic source. The 3D array equipped glider was programmed to glider along a helix curve in the vicinity of the acoustic source. The DOA capability of the glider using the acoustic data collected by the acoustic payload, along with the glider pitch-roll-heading, and position information is evaluated in this study. [*Work funded by NATO-Allied Command Transformation.*]

11:00

3aUWc3. Near-field localization of under water noise source based on matrix spatial filter with vector sensor array processing. Wang Xueyan (College of Underwater Acoust. Eng.,Harbin Eng. Univ., Harbin 150001,China, Harbin, Hei Longjiang, China), Shi Shengguo, and Shi Jie (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin 150001 Harbin, Shuisheng Dept.,College of Underwater Acoust. Lab., Harbin Eng. University, Harbin 150001, China, Harbin, China, shishengguo@hrbeu.edu.cn)

To detect the weak sources of interest area which are obscured by strong sources, a new algorithm of near-field localization of underwater noise source based on matrix filter is proposed. This algorithm use matrix filter as a pretreatment to filter array data of vector sensor, optimized design method is used to design the matrix filter, and then formulate the problem into a second-order cone programming model to calculate the matrix equation, which can strictly control the attenuation of stop band, and it can avoid increment of noise power by restrain the norm of matrix filter. With this filter matrix, a new covariance matrix is produced for the following beam-forming processing with vector sensor array. Compared with the previous work, this algorithm can accurately locate the weak sources of interest that are obscured by strong sources; in addition, higher location accuracy and spatial resolution,

better spatial anti-aliasing performance, as well as stronger capability to distinguish the starboard and larboard benefit from using vector sensor array. The simulation and real data show that this algorithm can precisely locate the weak source of interest among some strong sources.

11:15

3aUWc4. A sequential filtering algorithm for range estimation of a moving ship based on striation geometric features. Qunyan Ren, Li Ma (Key Lab. of Underwater Acoust. Environment, Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, renqunyan@mail.ioa.ac.cn), Shengchun Piao (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ. Harbin, Harbin, China), Shengming Guo (Key Lab. of Underwater Acoust. Environment, Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and Tianjun Liao (State Key Lab. of Complex System Simulation, Beijing, China)

There exist regular striation pattern in the broad-band noise field excited by a moving ship in shallow waters, whose time (range) frequency structure can be interpreted by the waveguide invariant theory. The striation characteristics have been exploited for underwater acoustic inversion problems, e.g., sediment geoacoustic parameter characterization. In this paper, the geometrical features of specific striations are observed for continuously ship ranging by a sequential algorithm. Comparing to matched-field processing techniques, this approach also has no requirements prior knowledge of environmental properties and burden of forward sound field calculation, which is similar to virtual receiver approach. The technique is tested on the striation fragments extracted from the ship noise data collected in Dalian 2008, and the outputs are in high agreement with *in-situ* GPS measurements. The results from independent runs with different initial values also suggest that this approach is robust to initial conditions in the sense of all runs can converge to the true values rapidly.

11:30

3aUWc5. Acoustic source localization in ocean waveguides using principles of Riemannian geometry. Steven Finette and Peter Mignerey (Acoust. Div., Naval Res. Lab.,Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

Source localization in underwater acoustics entails a comparison between acoustic fields involving some measure of correlation, looking for similarity between the acoustic field propagated from the true source location and replica fields propagated from different locations in the waveguide. The uniqueness of the deterministic Green function between pairs of source-receiver positions forms the basis for the solution to this inverse problem. We consider a novel approach to source localization based on non-Euclidean geometry, where the “distance” between cross-spectral density matrices (CSDMs) is used to estimate the spatial location of the source. The traditional Euclidean distance is not necessarily appropriate because CSDMs are not arbitrary points in space; rather, they form a manifold constrained by the facts that CSDMs are both Hermitian and positive definite. These properties naturally lead to the interpretation of geodesic distance between CSDMs as a measure of similarity between acoustic fields with this minimum distance, parametrized by replica source location, establishing an estimate of the source position. We discuss the underlying concepts and present simulation results for a waveguide with internal wave-induced ocean variability. Several Riemannian metrics are considered and compared to more traditional approaches to matched-field localization. [Work supported by the Office of Naval Research.]

WEDNESDAY MORNING, 29 NOVEMBER 2016 CORAL FOYER, 9:00 A.M. TO 12:00 NOON

Exhibit

The instrument and equipment exhibit is located near the registration area in the Coral Foyer.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

Exhibit hours are Monday, 28 November, 5:30 p.m. to 7:00 p.m., Tuesday, 29 November, 9:00 a.m. to 5:00 p.m., and Wednesday, 30 November, 9:00 a.m. to 12:00 noon.

Coffee breaks on Tuesday and Wednesday mornings will be held in the exhibit area as well as an afternoon break on Tuesday.

The following companies have registered to participate in the exhibit at the time of this publication:

AIP Publishing: publishing.aip.org/

American Institute of Physics: https://www.aip.org/

Aqua Sonic, Inc.: aqua-sonic.com

Echoview Software: www.echoview.com/

Head acoustics GmbH: www.head-acoustics.de/eng/

Mason Industries: www.mason-industries.com/masonind/

Ocean Sonics Ltd.: oceansonics.com/

ODEON A/S: www.odeon.dk/

PAC International: www.pac-intl.com/

RION Co., Ltd: www.rion.co.jp/english/

Sensidyne: www.sensidyne.com/

Springer: www.springer.com/us/

Teledyne RESON Inc.: www.teledyne-reson.com/

Session 3pAAa

Architectural Acoustics and Signal Processing in Acoustics: Advanced Analysis, Simulation, and Auralization in Room Acoustics II

Michael Vorlaender, Cochair
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Tetsuya Sakuma, Cochair
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Toshiki Hanyu, Cochair
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Chair's Introduction—1:00

Contributed Papers

1:05

3pAAa1. A synthesized boundary condition for wave-based room acoustics simulations using ensemble averaged impedance measured in-situ. Toru Otsuru, Reiji Tomiku (Oita Univ., 700 Dannoharu, Oita, Oita 870-1192, Japan, otsuru@oita-u.ac.jp), Noriko Okamoto (The Univ. of Kitakyushu, Kitakyushu, Japan), Sakura Saigo, and Saki Yamauchi (Oita Univ., Oita, Japan)

The importance of wave-based room acoustics simulation has been increasing in various fields of researches and design-stages. The problem remained, however, is how to model the rooms' absorptive boundary conditions. It is known that traditional absorption indices like sound absorption coefficient and normal impedance are not always suit for such a fine method like finite element method. Then, the concept and measurement technique of ensemble averaged impedance were presented. In our previous studies, a series of in-situ measurement of ensemble averaged impedances were successfully performed to result excellent reproducibility and repeatability. Herein, the outline of the concept of ensemble averaged impedance including the measurement technique is summarized, first. Second, several example results of practical measurements are given to exhibit that their uncertainties stay within the range for wave-based simulations to keep their resulting uncertainties less than just noticeable difference. Then, a mathematical-physical model of synthesized boundary condition is given using the ensemble averaged impedance in the sound field analysis by finite element method. Finally, the results of sound field simulations of realistic rooms are shown to examine the plausibility of the model.

1:20

3pAAa2. Absorption characteristics of micro-perforated panels using scale modeling and microflown impedance gun. Stephen Dance, Simon Brown, and Carl Ruegger (Urban Eng., London South Bank Univ., Borough Rd., London SE1 0AA, United Kingdom, dances@lsbu.ac.uk)

A theoretical and experimental comparison of the absorption characteristics of micro-perforated panels was undertaken. The micro-perforated panels were designed using the Sheffield University Java based website, constructed from 0.3 m by 0.3 m of acrylic, the thousands of holes were laser cut. The panels were large enough to be tested using the microflown impedance gun as well as a new scale model reverberation chamber based at London South Bank University. The 1:10 scale model reverberation chamber was constructed from acrylic materials and used a 3-D printed dodecahedron sound source. Data acquisition was through a 1/4" microphone connected through a Nexus preamplifier to a 24 bit 192 kHz sound card. MATLAB code was used to acquire and process the signal in accordance to ISO 354 including air compensation for high frequencies. Results will be presented for theoretical, scale, and impedance based measurement of the absorption coefficients for the micro-perforated panels.

1:35

3pAAa3. A hybrid impedance measurement method to inversely determine the impedance of finite porous absorber material samples. Rob Opdam, Mark Müller-Giebeler, and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, rob.opdam@akustik.rwth-aachen.de)

The determination of complex angle-dependent reflection factors is not common practice. Mostly, because existing methods are complicated and very time-consuming. This work presents a method that only needs a single sound pressure measurement of a finite porous absorber piece, placed in a semi-anechoic chamber, as input data. The complex pressure data obtained of the measurement are compared with that of a finite element method (FEM) simulation, which is based on the same geometric dimensions of the material sample as in the measurement, but with an arbitrary porous material. With a non-linear fitting algorithm, the simulated complex pressure data are adjusted to fit the measured data iteratively by changing the absorber model parameters (porosity, flow resistance, etc.) in the FEM simulation. When the simulated and measured results are identical within a defined threshold, the characteristic absorber properties are found and with that the angle-dependent complex reflection factors are determined. Applying this inverse approach allows also for a correction of the edge effects of the finite material sample, such that the found impedance is the impedance as measured on an infinite extended material sample and it also determines the individual material properties as porosity and flow resistivity of a porous material.

1:50

3pAAa4. Finite-difference time-domain analysis in porous materials by using Z-transform theory and infinite impulse response filter design. Jing Zhao, Hyojin Lee, and Shinichi Sakamoto (Inst. of Industrial Sci., Univ. of Tokyo, Japan, Tokyo 153-8505, Japan, zhaojing@iis.u-tokyo.ac.jp)

Porous materials are wildly used to control noise in various places. When simulating sound field containing porous materials by Finite-difference time-domain (FDTD) method, it is necessary to develop FDTD formulations in the porous materials. When the frame of the porous material is motionless, the porous material can be replaced on the macroscopic scale by an equivalent fluid. Based on the equivalent fluid model, FDTD formulations in the porous material are developed. The method combines IIR (Infinite impulse response) filter design and Z transform theory. The effective bulk modulus and the effective density of the porous material are frequency dependant, which is designed as IIR filter. Wave equations with IIR filter modeled parameters are solved in Z domain to avoid troublesome convolution integrals in the time domain. At the Z domain, new parameters are assumed to simplify the wave equations. Finally, wave equations and assumed parameters are returned back to the time domain. The accuracy of the proposed method is verified with the measurement. As two application examples, porous material coved with permeable membrane or non-permeable membrane are simulated and compared with the theory.

2:05

3pAAa5. Signal processing method for compensating air absorption and phase change in room impulse response. Shinichiro Koyanagi (Takenaka R&D Inst., 1-5-1, Ohtsuka, Chiba, Inzai-shi 270-1395, Japan, koyanagi.shinichirou@takenaka.co.jp)

It is often required to remove an influence of air absorption effect from a measured impulse response (e.g., absorption coefficient measurements by a reverberation room method). In particular, a room acoustic test in a scale model is greatly influenced by the air absorption and most of the attenuation of the observed impulse response is caused by air absorption, because the test measurement must be carried out in a high-frequency region including ultrasonic band in order to cover a wide frequency audible range in the real scale. Therefore, a high-accuracy compensation technique is needed so as to correctly evaluate the scale model responses. So far, two methods have been proposed, which utilize a subband division or a time-varying filter. In this study, we propose a new signal processing technique to remove the air-absorption effect by using a propagation distance distribution function obtained from the impulse response and a propagation constant. Some practical application results are presented and show that our proposed method can be performed with high accuracy and overcome some disadvantages of the existing methods.

2:20

3pAAa6. Design of a compact spherical loudspeaker array for simulating accurate instrument directivities for concert hall measurement and auralization. Matthew T. Neal, Molly R. Smallcomb, David A. Dick, and Michelle C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mtn5048@psu.edu)

An omnidirectional sound source is often used as a repeatable source to measure impulse responses in concert halls to obtain room acoustics metrics. The limitation of using an omnidirectional source is that real sound sources, such as musical instruments, do not exhibit omnidirectional radiation patterns. For an orchestra, many instruments with unique directional radiation patterns are distributed across the stage. To achieve more realistic auralizations, a 20-element compact spherical loudspeaker array was constructed to simulate the frequency-dependent radiation properties of instruments. This loudspeaker array is capable of achieving a truncated third-order spherical harmonics representation of an instrument's radiation pattern. Once the source is moved to different instrument locations around stage, frequency-dependent weighting factors for each driver are controlled to match the source's radiation pattern to that instrument. Full three-dimensional impulse response measurements of the source were made to characterize the source's directional performance. From these measurements, frequency-dependent gains were applied to determine the accuracy and frequency range of the directional reproduction, and the results for various instruments will be presented. The implementation of this source into an existing measurement setup for future measurements of U.S. and European concert halls will also be discussed. [Work supported by NSF Award 1302741.]

2:35

3pAAa7. Musical instrument directivity database for simulation and auralization. Gottfried Behler (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstraße 5, Aachen D-52074, Germany, gkb@akustik.rwth-aachen.de), Noam R. Shabtai (Inst. of Tech. Acoust., RWTH Aachen Univ., Beer Sheva, Israel), Stefan Weinzierl (Audio Commun. Group, Tech. Univ. Berlin, Berlin, Germany), and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany)

A study of 41 modern and historical orchestral musical instruments was made recently. All instruments were played in an anechoic chamber and recorded simultaneously with an array of 32 microphones place on a sphere of 4.2 m diameter. The musicians were placed in the centre of the sphere and instructed to play chromatic scales covering the whole range of each instrument at two levels—very soft and very loud. From these recordings the directivity for steady tones has been derived. The radiation pattern is represented in the 4-th order spherical harmonics domain, using 25 coefficients and 31 third-octave frequency bands. For 7 instruments, acoustic source centering is applied in order to align the acoustic center of the sound source to the physical center of the microphone array using the center-of-mass approach [Ben-Hagai *et al*., JASA 2011] and using the phase symmetry approach [Shabtai and Vorländer, JASA 2015]. Analysis of the database is found in [Shabtai *et al*. JASA 2016 (submitted)]. From this analysis, a database has been created that will be available for scientific use soon. This database is recommended to be used in virtual reality applications to improve the sense of realism perceived by the user.

2:50–3:05 Break

3:05

3pAAa8. A new metric to predict listener envelopment based on spherical microphone array measurements and ambisonic reproductions. David A. Dick and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dad325@psu.edu)

The objective of this work is to propose a new metric for listener envelopment (LEV), the sense of being immersed in a sound field. Current LEV metrics, for example, Late Lateral Energy Level (L_J), are based only the late sound field. However, recent studies indicate that the late component of the impulse response (IR) alone is inadequate for predicting LEV. For this study, room IR measurements were obtained with a 32-element spherical microphone array in a total of nine halls ranging from 400-2500 seats. A subset of these IRs were selected for a subjective listening test and processed for 3rd order Ambisonic reproduction over a 30-loudspeaker array. Two sets of modified IRs were also generated for the listening test by combining parts of IRs from different halls and by bandpass filtering to select certain frequency ranges, respectively. The ambisonic IRs were convolved with anechoic orchestral recordings and subjects rated the perceived LEV for each stimuli. The IRs were also analyzed objectively via plane wave decomposition (PWD). Energy from the PWD as a function of time, frequency, and arrival direction was correlated with the subjective LEV ratings to develop the new metric. [Work was supported by NSF Award 1302741.]

3:20

3pAAa9. Subjective evaluation of sound diffusion in a surround hall. Hyung Suk Jang (Dept. of Architectural Eng., Hanyang Univ., Seoul 133-791, South Korea, janghyungs@gmail.com), Hansol Lim, and Jin Y. Jeon (Dept. of Architectural Eng., Hanyang Univ., Seongdong-gu, Seoul, South Korea)

The effects of diffusing elements on sound field diffuseness were investigated by subjective evaluations using binaural impulse responses (BIRs) in seating areas in a surround hall. The room acoustical parameters as well as the number of peaks (Np) and local maxima within -20 dB of the amplitude

of direct sound were investigated to evaluate the diffuseness in the halls. For the listening test, stimuli were selected to explore the combination of acoustic parameters and degree of diffusion. The music signals were convolved with the measured BIRs in different listening positions. Based on the listening test, it was found that positions close to the diffusers with a high scattering coefficient were preferred by the listeners.

3:35

3pAAa10. The influence of hearing and seeing on aesthetic judgments of performance spaces. Hans-Joachim Maempel and Michael Horn (Staatliches Institut für Musikforschung, Tiergartenstraße 1, Berlin 10785, Germany, horn@sim.spk-berlin.de)

The perception of rooms involves different modalities, particularly hearing and seeing. Fundamental questions of audio-visual room perception have, however, not been answered yet. We investigated to what extent aesthetic judgments on music and speech performances in different spaces are based on hearing, seeing, and their interaction. Meeting methodological criteria such as optoacoustic commensurability and optoacoustic dissociation, and applying a data-based, three-dimensional, high-resolution optoacoustic simulation providing nearly all perceptually relevant physical cues allowed for a valid proportionate quantification of these shares for the first time. BRIRs and panoramic stereoscopic images were acquired in six rooms. Recordings of both a music and a speech performance were put in the rooms by applying dynamic binaural synthesis and chroma-key compositing. The scenes were played back by the use of a linearized extraaural headset and a semi-panoramic stereoscopic projection system. Test participants were asked to rate the pleasantness, powerfulness, and excitingness of the performance, their spatial presence in the virtual scene, and the matching of the acoustical and the optical room. Statistical analyses indicate among others, that the aesthetic judgments mainly relied on acoustic information, whereas the presence and the matching judgment was mainly influenced by the interaction of optical and acoustical room properties.

3:50

3pAAa11. Study of the perception of warmth in concert halls and correlation with room acoustics metrics. Kristina M. Sorensen and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, kms674@psu.edu)

Challenges in concert hall acoustics research occur in bridging the domain between subjective attributes and objective metrics. Several studies have shown that the primary attributes associated with concert hall acoustic preference ratings are strength, clarity, and timbre. The purpose of this study was to investigate the correlation of warmth, which is closely associated with timbre, with the objective metric of bass ratio (BR), which is a measure of the ratio of acoustic information (such as reverberation time or strength) at low-frequencies to that at mid-frequencies. The stimuli for this study were simulated binaural room impulse responses (BRIRs) obtained in ODEON and convolved with a concise orchestral anechoic motif. The BRIRs were simulated using models of five European halls. A model of the Vienna Musikverein with multiple absorption settings was also used to achieve different values of BR within a single hall geometry. The stimuli were presented to participants over headphones and they were asked to rate warmth, low- and high-frequency reverberance on continuous five-point scales. Participants rated each of the attributes in separate sets to avoid potential bias in the ratings. A repeated measures analysis of variance was conducted to examine the relationship between the subjective ratings and BR.

4:05

3pAAa12. A fundamental study of evaluating sound fields in performance spaces based on equal tempered scale. Akiho Matsuo (Dept. of Architecture, Graduate School of Sci. and Technol., Nihon Univ., 7-24-1, Narashinodai, Funabashi, Chiba 274-8501, Japan, motu.kepgad@gmail.com), Toshiki Hanyu, and Kazuma Hoshi (Dept. of Architecture and Living Design, Junior College, Nihon Univ., Funabashi, Chiba, Japan)

Generally, the reverberation times, energy decay curves, directional characteristics, and so on are analyzed in 1 octave bands or 1/3 octave bands for evaluating acoustic conditions in performance spaces. The center frequencies such as 1000 Hz are defined by engineering implication. However, these center frequencies are not always adequate for evaluating performance spaces because music is basically composed of musical scale like the equal tempered scale. Therefore, it is thought that it is reasonable to use center frequencies based on the equal tempered scale in order to evaluate performance spaces. Based on this idea, we calculated reverberation times and energy decay curves for 1/12 octave bands at which center frequencies are set by using the equal tempered scale. Directional characteristic of reflected sounds is an important factor for evaluating spatial impressions in performance spaces. Therefore, we measured the directional characteristics based on sound intensity in the 1/12 octave bands. Moreover, we focused on chord which is a combination of several musical tones, in other words, several 1/12 octave bands. We calculated energy decay curves and directional characteristics focusing on the chord. As a result, characteristic tendencies were obtained on the energy decay curves and the directional characteristics.

4:20

3pAAa13. The use and mis-use of reactive systems in the design of performance halls. Richard Talaske and Scott Hamilton (TALASKE | Sound Thinking, 1033 South Blvd., Oak Park, IL 60302, rick@talaske.com)

While the "surface of infinite impedance" or "totally absorptive material" makes our acoustic theories understandable and our analytical calculation methods useful, more often than not such conditions are not met in built performance facilities. Since the materiality of a performance hall strongly influences the quality of sound within a space, the physical reality of the reactive impedance of building elements must be managed to achieve quality sound. This paper discusses the practical application of various room acoustic design options, including wall and ceiling material choices, and offers comment on how various reactive systems within performance halls can detract from a room and/or reintroduce sound energy into a space. Discussion will occur regarding how such conditions can be either favorable or detrimental to the aural environment. Examples will be offered which suggest a reactive room surface can offer favorable room acoustic results for impulsive sound events, and unfavorable results with sustained sound events. Observations and subsequent testing suggest that the ability to discern the impact of reactive systems within a hall depends upon the test methods implemented during acoustic measurement sessions.

Session 3pAAb

Architectural Acoustics and Signal Processing in Acoustics: Advanced Analysis, Simulation, and Auralization in Room Acoustics III (Poster Session)

Michael Vorlaender, Cochair
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Toshiki Hanyu, Cochair
Department of Architecture and Living Design, Junior College, Nihon University, 7-24-1, Narashinodai, Funabashi 274-8501, Japan

All posters will be on display from 4:30 p.m. to 5:30 p.m.

Contributed Papers

3pAAb1. Evaluating acoustical features of 3D sound rendering systems by using a visualizing tool of sound intensities. Masataka Nakahara (Onfuture Ltd. / SONA Corp., 2-19-9 Yayoi-cho, Nakano-ku, SONA Corp., Tokyo 164-0013, Japan, nakahara@onftr.com), Takashi Mikami (SONA Corp., Tokyo, Japan), and Akira Omoto (Kyushu Univ. / Onfuture Ltd., Fukuoka, Japan)

The authors have developed a simple measurement/analyzing tool which visualizes 3D acoustical properties of sound by using sound intensity information. The tool, VSV (Virtual Source Visualizer), has already being used for practical acoustic design works, and the authors have also begun to use it for evaluating acoustical features of 3D sound rendering systems. On the audio industries, various 3D sound reproduction systems have already been commercialized. Therefore, further skill is now required to design acoustic specifications of their production studios. From this point of view, development of a simple method to obtain relationship between loudspeaker positions and source locations rendered by a 3D reproduction system would be helpful. This research introduces analyzing flows of the measurement tool briefly, and shows measured results of physical and/or phantom sound locations detected from sound intensity information in some different 3D reproduction environments including commercial audio strategies such as Dolby Atmos and 22.2ch, and a wave theory based Boundary Surface Control system, BoSC. Verifying differences between target source positions and measured ones, the authors suggest the feasibility of the evaluation method of a rendered 3D sound field by a simple measurement method.

3pAAb2. Evaluating the auralization of a small room in a virtual sound environment using objective room acoustic measures. Axel Ahrens, Márton Marschall, and Torsten Dau (Dept. of Elec. Eng., Hearing Systems group, Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, aahr@elektro.dtu.dk)

To study human auditory perception in realistic environments, loudspeaker-based reproduction techniques have recently become state-of-the-art. To evaluate the accuracy of a simulation-based room auralization of a small room, objective measures, such as early-decay-time (EDT), reverberation time, clarity, interaural cross-correlation (IACC), and the speech transmission index were measured in an IEC listening room for 28 source-receiver combinations. The room was then modeled in the room acoustics software ODEON, and the same objective measures were also evaluated for the auralized version of the room. The auralizations were generated using the loudspeaker-based room auralization toolbox (LoRA; Favrot and Buchholz, 2010) and reproduced in a 64-channel loudspeaker array, set up in an anechoic chamber. Differences between the objective measures evaluated in the real and the virtual room were within about twice the just-noticeable differences for most measures, and were comparable to the median results of the study by Favrot and Buchholz, who did not consider the contribution of a real reproduction system. However, the EDT showed considerably higher errors than the other measures, even though medians were similar in the real and auralized room. This suggests that fine details in the early part of the room impulse response may be difficult to reproduce accurately.

Session 3pAAc

Architectural Acoustics: Assorted Topics in Architectural Acoustics (Poster Session)

Damian Doria, Chair
Stages Consultants LLC, 75 Feather Ln, Guilford, CT 06437-4907

All posters will be on display from 4:10 p.m. to 5:30 p.m. To allow all contributors in this session to see other posters, authors of odd-numbered papers will be at their posters from 4:10 p.m. to 4:50 p.m. and authors of even-numbered papers will be at their posters from 4:50 p.m. to 5:30 p.m.

Contributed Papers

3pAAc1. Characterization of the acoustic quality of classrooms in a music education facility. Erasmo F. Vergara (UFSC, Florianópolis, SC, Brazil), Stephan Paul (UFSC, rua Joao Colin, 2700, Joinville 97080-035, Brazil, stephan.paul@ufsc.br), Eric Brandão (Undergraduate program Acoust. Eng., UFSM, Santa Maria, RS, Brazil), and Fernanda Marros (Graduate Program Civil Eng., UFSM, Santa Maria, Brazil)

Classrooms of a music education facility were characterized by the user by means of a questionnaire study and by acoustic impulse response measurements and subsequent calculation of reverberation time, definition, and other room acoustics parameters. The questionnaire study enabled to understand the preferences of students and music teachers regarding the acoustic quality of a total of twenty rooms for music practice and teaching. It was observed that preference for a certain room also depends on the instrument mostly played by the user. Three study rooms and three collective classrooms were the most cited ones and these have been evaluated according to ISO 3382. Opinions of the musicians showed to be coherent with the measurement data as the rooms considered to be dry had reverberation times around 0.3 seconds, and the rooms considered to be reverberant had reverberation times around 1.5 seconds. The six classrooms that were characterized as clear and well defined rooms, showed values for Clarity around 1 dB for live rooms and 14 to 22 dB for dry rooms. Definition remained above 43% for all rooms with Central Time below 24 ms for clarity rooms and less reverberant, and above 98 ms for the live rooms.

3pAAc2. Autonomous optimal location of articulation materials using robotics. Masashi Uehara, Shuhei Kawai, Shigeki Okawa (Chiba Inst. of Technol., 2-17-1 Tsudanuma, Narashino-shi, Chiba-ken 275-0016, Japan, s1126017jn@s.chibakoudai.jp), and Manabu Fukushima (Nippon Bunri Univ., Oita-shi, Oita-ken, Japan)

We developed an application that determines optimal places of a robot equipped with articulation materials to improve sound environment. Lots of acoustic devices that incorporated architectural acoustics technology have been developed (e.g., articulation material, sound-absorbing material, etc.). In many cases, those materials are placed at positions that are effective by professional people. Our purpose is to make an autonomous robot to decide the material's position instead of professionals. We chose "Aural Sonic," the articulation materials to be mounted on the robot. "Aural Sonic" is one of the acoustic devices that is expected effects to improve the hearing. This robot can change the position and angle little by little, to measure the impulse response each time. The application extracts the acoustic features corresponding to the speech clarity from the measured impulse response and calculates the evaluation value. The robot, which gives the position with high evaluation value, could be used for the articulation materials more effectively. We introduce the system that finds the optimal location of the articulation material with the robot.

3pAAc3. Autonomous mobile robot to improve sound environment for speech conversation. Shuhei Kawai, Masashi Uehara, Shigeki Okawa (Chiba Inst. of Technol., 2-17-1 Tsudanuma, Narashino-shi, Chiba-ken 275-0016, Japan, s1326035cq@s.chibakoudai.jp), and Manabu Fukushima (Nippon Bunri Univ., Oita-shi, Oita-ken, Japan)

This study pursues to improve sound environment for speech conversation by moving an articulation panel by a robot. We tried to move the articulation panel to behind of the listeners by a small robot to keep fixed distance from the listener. An optimal location of the articulation panel depends on the sound sources such as speech conversation or music appreciation. It is usually decided based on the sense of users and experts. If we use a room for the purpose of speech conversation, the articulation panel is placed behind the listeners. However, moving the articulation panel optimally is almost impossible in terms of economical and physical conditions. In recent years, robot technology for transportation is in progress widely. We attempt resolving these problem by robotics. The articulation panel is framed by wood to have absorption and reflection surfaces, and the robot has omni-wheels for ultra-pivot turn performance. It can turn the surface on the sound source by the rotation of the robot. The robot has four-wheel drive because it should have large stable range as to the purpose to carry objects. Several sensors are used such as an ultrasonic sensor and an infrared sensor to have robustness. As a result, we approached to improvement of sound environment by robotics.

3pAAc4. Perceptually plausible acoustics simulation of single and coupled rooms. Torben Wendt, Steven van de Par, and Stephan D. Ewert (Acoust. Group, Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Carl-von-Ossietzky-Straße 9-11, Oldenburg 26111, Germany, torben.wendt@uni-oldenburg.de)

The fast and perceptually plausible room acoustics simulator [RAZR, see Wendt *et al.*, JAES **62**, 11 (2014)] has been optimized and extended to simulate the acoustics of connected rooms. RAZR synthesizes binaural room impulse responses (BRIRs) with high computational efficiency using a hybrid approach: early reflections are calculated as image sources for a shoebox-room approximation up to a low order, and the later reverberation is generated by a binaurally extended feedback delay network (FDN). For the extension toward two coupled rooms, validity, visibility, and diffraction of the image sources in the two rooms were taken into account, as well as different reverberation properties and the opening angle of the door to the neighbor room seen from a specific receiver position. The suggested method was evaluated by comparing measured and synthesized BRIRs for several rooms differing in size and reverberation time. For room acoustical parameters such as early decay time, late decay time, definition, and clarity a good agreement in terms of the Pearson correlation coefficient (0.86—0.99) was achieved between measured and simulated BRIRs. Subjective listening tests showed a good agreement of ratings of perceptual attributes (e.g., tone-color, reverberation time, envelopment, and naturalness) for the measured and synthesized BRIRs.

3pAAc5. Impact noise insulation performance of commonly used materials when excited by different impact noise sources. Andriele d. Panosso (UCEFF, Rua Benjamin Constant, Santa Maria, Rio Grande do Sul 97050022, Brazil, andrielep@gmail.com) and Stephan Paul (Universidade Federal de Santa Catarina, Joinville, Santa Catarina, Brazil)

This study aimed to verify the performance of different materials used as impact noise insulation when excited by different impact sources. The method applied consisted of several acoustic measurements, following the standards' instructions and using different impact sources, such as the tapping machine and an impact tire. When using the tapping machine results indicate rather big differences in noise insulation performance for the different materials, showing a range from 50 to 76 dB for the Weighted Standardized Impact Sound Pressure Level while measurements with the tire showed a range of 72 to 82 dB for the Average Maximum Impact Pressure Level, meaning very similar noise insulation for all materials.

WEDNESDAY AFTERNOON, 30 NOVEMBER 2016 CORAL 2, 1:00 P.M. TO 2:30 P.M.

Session 3pABa

Animal Bioacoustics: Session in Honor of Whitlow Au II

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

Marc Lammers, Cochair
Hawaii Institute of Marine Biology, 46-007 Lilipuna Rd., Kaneohe, HI 96744

Tomonari Akamatsu, Cochair
Fisheries Research Agency, 7620-7, Hasaki, Kamisu, Ibaraki 314-0408, Japan

Contributed Papers

1:00

3pABa1. Interaction of gain-control mechanisms in the sonar of odontocetes. Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru) and Paul E. Nachtigall (Hawaii Inst. of Marine Bilogy, Kaneohe, HI)

The sonar of odontocetes processes echo signals within a wide range of echo levels. The level of echoes varies by tens of dB depending on the the target strength, the distance to the target, and the sound absorption by the media. The sonar of odontocetes has several mechanisms to compensate for the echo-level variation (gain control): (i) variation of emitted sonar pulse levels (the longer the distance to the target, the higher the level of the emitted pulse), (ii) short-term variation of hearing sensitivity based on forward masking of the echo by the preceding self-heard emitted pulse and subsequent release from the masking, and (iii) long-term active control of hearing sensitivity. These three mechanisms not just add to one another but act coordinately and form a united system of gain control. In particular, an increase in the sonar pulse level prolongs the distance of action of the forward-masking mechanisms. Active variation of hearing sensitivity makes the range of action of the forward-masking mechanism fitting the particular target distance and strength. These interactions of the gain-control mechanisms makes the auditory system capable of effectively analysis of echoes within a wide range of target strengths and distances.

1:15

3pABa2. Beam patterns of the dolphin demonstrate task-dependent dynamics. Dorian S. Houser, Lois Talmadge (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), Josefin Starkhammar (Lund Univ., Lund, Sweden), and Partick Moore (National Marine Mammal Foundation, San Diego, CA)

The echolocation beam of the bottlenose dolphin was first carefully described by Au and colleagues (1978)["Propagation of Atlantic bottlenose dolphin echolocation signals," J. Acoust. Soc. Am. **64**, 411-422] using various hydrophone array configurations and targets located in front of the dolphin and along its longitudinal axis. Measured beams were described as vertically elevated with mean vertical and horizontal 3-dB beamwidths of ~10°. The experimental paradigm was later augmented with denser hydrophone arrays, greater spatial coverage of the acoustic field, and the training of target detection with targets presented to the left or right of the dolphin's longitudinal axis. Utilizing two dolphins, beam steering capabilities and beamwidth control were demonstrated. The two dolphins steered the axis of the echolocation beam up to 18° and 28° in the horizontal plane. Horizontal beamwidths were bimodally distributed in one dolphin, with peaks at 16° and 26°. The other dolphin had a unimodal distribution of beamwidths (peak at ~25°), likely reflecting a learned strategy for the off-axis detection task. The use of dense hydrophone arrays and challenging echolocation tasks continue to build on the work of Au and colleagues, who provided a paradigm that has greatly contributed to our understanding of dolphin biosonar.

1:30

3pABa3. Using "phantom" echoes to study dolphin biosonar. James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Jason Mulsow, Brian K. Branstetter, Dorian S. Houser, and Patrick W. Moore (National Marine Mammal Foundation, San Diego, CA)

Biosonar studies have greatly benefited from the use of electronic, or "phantom," echoes. In this paradigm, amplitude and timing information are extracted from an emitted biosonar pulse, then a delayed signal is broadcast to the animal to appear as an echo from a more distant target. Phantom echoes provide unique capabilities for studying biosonar, since they allow echo features such as amplitude and delay to be independently manipulated. In 1987, Whit Au and colleagues described a phantom echo system for use with marine mammals [Au *et al.* (1987). "Phantom electronic target for dolphin sonar research," J. Acoust. Soc. Am. **82**, 711-713]. In this system,

delayed broadcasts of one or more replicas of the dolphin click were triggered by dolphin click emissions. A major improvement in phantom echo systems was later described by Aubauer and Au (1998) ["Phantom echo generation: A new technique for investigating dolphin echolocation," J. Acoust. Soc. Am. **104**, 1165-1170], who simulated the impulse response of a physical target, rather than broadcasting a stereotyped waveform. In this talk, phantom echo system concepts are briefly presented and several recent applications of phantom echo systems—inspired by the work of Aubauer and Au—are discussed. [Work supported by ONR.]

1:45

3pABa4. Information-seeking in an echolocating dolphin. Heidi E. Harley (Psych., , Div. of Social Sci., New College of Florida, 5800 Bay Shore Rd., Sarasota, FL 34243, harley@ncf.edu), Wendi Fellner (The Seas, Disney's Epcot, Lake Buena Vista, FL), Candice Frances (Psych., New College of Florida, Sarasota, FL), Amber Thomas, Barbara Losch, and David Feuerbach (The Seas, Disney's Epcot, Lake Buena Vista, FL)

Dolphins gain information through echolocation, a publicly accessible sensory system in which dolphins produce clicks to investigate objects. We measured information-seeking behavior by counting clicks that a blindfolded dolphin directed toward the sample object in a three-alternative matching task with object sets that varied in discriminability: Indiscriminable (performance accuracy M = 33%) vs. Easy (performance accuracy M>90%). The dolphin produced a similar number of clicks when first investigating each set type. Across multiple sessions, however, the dolphin emitted fewer clicks only when investigating indiscriminable (vs. easy) sets. Reduced echoic investigation with indiscriminable, but not easy, object sets was not due to overall motivation: the differential relationship between click number and object set discriminability was maintained when difficult and easy trials were interleaved and when objects from originally difficult sets were grouped with more discriminable objects. Further analyses of the dolphin's click production towards the choice alternatives also revealed that the dolphin produced more clicks towards the chosen (vs. unchosen) alternatives when solving the task. Overall, these data suggest that dolphins calibrate the effort they invest in information seeking in accordance with the information content available in their immediate environment.

2:00–2:30 Panel Discussion

WEDNESDAY AFTERNOON, 30 NOVEMBER 2016 — CORAL 2, 2:45 P.M. TO 5:45 P.M.

Session 3pABb

Animal Bioacoustics: On the Bleeding Edge of Animal Bioacoustics Technology

Selene Fregosi, Chair

Oregon State University, Hatfield Marine Science Center, 2030 SE Marine Science Drive, Newport, OR 97365

Contributed Papers

2:45

3pABb1. Acoustic methods of pest detection in agricultural shipments. Alexander Sutin, Timothy Flynn, Nikolay Sedunov, Hady Salloum, Alexander Sedunov (Maritime Security Ctr., Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu), and David Masters (Sci. and Technol. Directorate, Dept. of Homeland Security, Washington, DC)

Stevens Institute of Technology, in cooperation with the DHS Science and Technology Directorate and the U.S. Customs and Border Protection, has been investigating engineering solutions to augment the current inspection process at ports of entry in an effort to minimize the threat posed by invasive species. Stevens has built several sensitive acoustic systems for detection of tiny acoustic/vibrational signals produced by moving insects and tested them in a Laredo, TX port of entry. The system for detection of insects in vegetables and herbs uses a soundproofed case where boxes filled with vegetables or herbs were placed during test. For wood boring insects, sensitive custom-made accelerometers were built. Special algorithms for detection of events connected with insect movement and separating them from the ambient noise were developed. Tests conducted demonstrated reliable detection of *Copitarsia* larva in a large box of vegetables. Smaller insects similar to Khapra beetles were clearly detected in a relatively small volume of grains (2-4 lbs). [This work was supported by DHS's S&T Directorate.]

3:00

3pABb2. Experimental audible sonar to model echolocation by the blind. Roman B. Kuc (Elec. Eng., Yale, 15 Prospect St., 511 Becton, New Haven, CT 06511, roman.kuc@yale.edu) and Victor B. Kuc (Comput. Sci., Iona College, New Rochelle, NY)

Some blind humans echolocate by emitting palatal clicks and processing echoes, even when there is temporal emission/echo overlap (EEO). Our previous work indicates that high frequencies in the emission that travel directly to the ears are attenuated and, when high frequencies are heard, they come from target echoes. Binaural processing of these high frequency components in the EEO signals provide target range and bearing. Our experiments with 3D-printed parabolic pinnae and a speaker emitter indicate that while the high frequency components are important for object classification, the low frequencies in the emission provide more robust bearing localization because pinna diffraction effects are reduced. Classifying targets with EEO signals requires analysis in the power spectral domain because of the phase insensitivity of hearing. The power spectrum Fourier inverse estimates the autocorrelation function of the target reflector sequence. Robust autocorrelation estimates occur when the emission and echo contain comparable energies. Our experiments demonstrate that the echo energy varies with target type (e.g., planar or cylindrical) and the optimum range for classification depends on the target and itself forms a target classification feature.

3:15

3pABb3. Simultaneous recordings of marine mammal calls by a glider, float, and cabled hydrophone array. Selene Fregosi (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Hatfield Marine Sci. Ctr., 2030 SE Marine Sci. Dr., Newport, OR 97365, selene.fregosi@oregonstate.edu), Holger Klinck (BioAcoust. Res. Program, Cornell Lab. of Ornithology, Cornell Univ., Ithaca, NY), Haru Matsumoto, Alex Turpin (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR), Stephen W. Martin, Brian M. Matsuyama (National Marine Mammal Foundation, San Diego, CA), Tyler A. Helble, E. Elizabeth Henderson (Navy Marine Mammal Program, SPAWAR Systems Ctr. Pacific, San Diego, CA), David J. Moretti, Ronald P. Morrissey (Naval Undersea Warfare Ctr., Newport, RI), and David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR)

Recent advances in passive acoustic monitoring (PAM) technologies have led to development of mobile autonomous platforms for recording marine mammals. These instruments may allow greater spatial and temporal sampling than traditional towed or bottom moored systems. However, comparison of recording abilities of these instruments to traditional methods has yet to be performed. We deployed two types of commercially available platforms at the Southern California Offshore Range (SCORE) complex in late December 2015 through early January 2016. The QUEphone, based on the APEX float (Teledyne Webb Research, Falmouth, MA, USA), is a buoyancy driven device capable of descending to 2000 m where it drifts horizontally with the currents. The Seaglider (Kongsberg Underwater Technology, Lynwood, WA, USA) is also buoyancy driven, but dives repeatedly up to 1000 m following a flight path controlled via satellite. We deployed one glider and two floats, each equipped with identical acoustic sensors developed by Oregon State University, sampling at 125 kHz. Each instrument recorded 250-300 hours of data over 13 days. Marine mammal detections included beaked whales, Risso's dolphins, and fin, blue, and humpback whales. We utilized known marine mammal locations derived from the SCORE hydrophone array to compare PAM capabilities of these novel mobile platforms.

3:30

3pABb4. IRAP: An integrated, real-time, autonomous passive acoustic monitoring system for beaked whale detection, localization, and tracking. Vince Premus, Philip Abbot, Charles Gedney, Russ Christman, Mark Helfrick, Richard Campbell, and Kim Douglas (OASIS, Inc, 5 Militia Dr., Lexington, MA 02421, primus@oasislex.com)

The integration and demonstration of passive sonar hydrophone arrays into autonomous undersea platforms for acoustic marine mammal monitoring has witnessed significant advances in the last few years due to the availability and scalability of low-cost, low-power commercial technology for acoustic remote sensing. In this paper, we will review the at-sea performance of a nine-element Mills Cross high frequency hydrophone array (HFA) integrated into an autonomous undersea vehicle (AUV) for the detection, classification, localization, and tracking (DCLT) of beaked whales during a deployment conducted off leeward Kauai in February, 2016. Using passive sonar equation analysis, we will review the design rationale of the HFA, and quantify system performance using standard signal processing metrics such as measured array gain, tracking accuracy, and detection range on a calibrated high-frequency source transmitting replica beaked whale click trains at a *de minimus* source level. A candidate autonomous real-time passive sonar processing architecture that combines an adaptive beamformer/matched filter front-end with a false alarm mitigation back-end to balance detection sensitivity, classifier robustness, and processing throughput will also be presented. We conclude with a cost/power/persistence trade-off for some commercially available autonomous platforms. [This work was supported by NAVFAC Living Marine Resources Program.]

3:45

3pABb5. Automatic detection and classification of toothed whale echolocation clicks in diverse long term recordings. Scott Lindeneau, Yun Trinh (Comput. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182, slindeneau@gmail.com), Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA), and Marie Roch (Comput. Sci., San Diego State Univ., San Diego, CA)

We present results of classification of toothed whale echolocation clicks to species for 4 TB of recordings in the development data of 7th Intl. DCLDE Workshop. The data span multiple seasons, years, locations, and instruments. Five species were acoustically identified by analysts, and a sixth category was assigned to echolocation clicks that could not be identified to species by analysts. Methods were developed to identify periods of echolocation activity taking into account sporadic false positives that occur throughout the data, and to reduce false positives from both anthropogenic and biologic sources. Dense echolocation activity presented particular challenges for noise removal and long term recordings permitted the targeting of regions for noise estimation outside of echolocation encounters. Extracted features consisted of noise normalized cepstral characterizations of spectra, inter-click interval, and -3 dB bandwidth. These features were classified with a Gaussian mixture model (GMM). A robust scoring method was introduced to reduce outlier influence by using a per-click voting scheme within each encounter as opposed to joint likelihood scores. The error rate across 300 Monte Carlo trials on the development data set was 34.5%. When encounters from the unspecified toothed whales were removed, the error rate dropped to 15.5%.

4:00–4:15 Break

4:15

3pABb6. Passive acoustic localization of North Atlantic Right Whales using a modified near-field Bartlett beamformer. Dean L. Hawthorne and Daniel P. Salisbury (Cornell Lab. of Ornithology, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, dh27@cornell.edu)

Owing to the principle of least commitment, near-field beamformers can produce passive acoustic source localizations which are robust to noise and interference. The power output of a traditional Bartlett near-field beamformer may be formulated in terms of the pairwise cross correlation functions of the sensor signals. We present a modified Bartlett processor wherein the ordinary cross correlation is replaced by generalized cross correlation, generalized in both time and frequency. As an example of the utility of the technique, we present results from a six element pentagonal marine array with a baseline of approximately 15 km deployed off the coast of Virginia. Localization of synthetic FM sweeps from known source locations within the array yield deviations from the true source location as measured by GPS of less than 100 meters. Localization of actual North Atlantic Right Whale calls provide good alignment of the unknown channel arrivals and acceptably small error ellipses.

4:30

3pABb7. Prototype of a low-cost towed array with data-logging and short-range telemetry. Eduardo Romero-Vivas, Fernando D. Von Borstel, Francisco Villa Medina (CIBNOR, Av Instituto Politecnico Nacional 195, Playa Palo de Santa Rita Sur, La Paz, Baja California Sur 23096, Mexico, evivas@cibnor.mx), Omar A. Bustamante (Acoust. Lab, ESIME, IPN, IMA, D.F, Mexico), Sergio Beristain (Acoust. Lab, ESIME, IPN, IMA, Mexico City, Mexico), and Joaquin Gutiérrez-Jagüey (CIBNOR, La Paz, Mexico)

Passive Acoustic Monitoring studies have been benefited from open source software initiatives such as PAMGuard and ISHMAEL, but their widespread use is often limited by availability of low cost hardware. This work presents the design of a towed hydrophone array that can be built with off the shelf components. The system comprises a Raspberry Pi computer to log GPS, three-axis accelerometer, magnetometer, gyroscope, and audio from the hydrophone array (20 Hz-44 kHz). Data can be analyzed in PAMGuard, either in real time, through a short range (10 m) RF link, or retrieved later from the data logger. The array has been tested in La Paz bay Mexico, where semi-resident populations of long-beaked common dolphin

(*Delphinus capensis*) and bottlenose dolphin (*Tursiops truncatus*) have been reported. Results and performance of the array in four modalities are presented; as a drift array, as a secondary array deployed from a small vessel, as an array towed by a Kayak, and as an array towed by an Autonomous Surface Vehicle. This array provides a useful tool for fieldwork, and since it is built form off the shelf components on a low budget, it could also be used for educational purposes, such as in PAMGuard Courses.

4:45

3pABb8. Acoustic cam on a remote operated vehicle for coral reef soundscape recording. Eduardo Romero-Vivas, Joaquín Gutiérrez-Jagüey (CIBNOR, La Paz, Baja California Sur, Mexico), Omar A. Bustamante (Acoust. Lab, ESIME, IPN, IMA, Austral 77 Col Atlanta, Cuahutitlan Izcalli, Mexico City 54740, Mexico, omar.p@hotmail.com), Francisco Villa Médina (CIBNOR, La Paz, Mexico), Sergio Beristain (Acoust. Lab, ESIME, IPN, IMA, Mexico City, Mexico), and Fernando D. Von Borstel (CIBNOR, La Paz, Mexico)

Although the broadband high level sounds of snapping shrimps dominate the soundscape of a coral reef, the spatial, spectral, and temporal variations due to biotic and abiotic factors are indicators of the health of these vulnerable environments; for instance, reefs become louder during new moons of the wet season, when many larval organisms settle, or tends to go quiet as animals abandon them when conditions become less favorable. Moreover, the soundscape is able to provide information on the reef benthic composition and has been identified as a possible driver of reef population dynamics. To study this particular environment, a Remote Operated Vehicle (openROV) is being used to visually identify the biota associated with a shallow coral reef near a major port in La Paz, Mexico. An acoustic linear array has been incorporated, providing directional sound recording capabilities to capture better space variability of the coral soundscape. By combining visual and acoustical clues in an acoustic cam, it is also feasible to associate fish species and acoustic signals. A preliminary assessment of the design for monitoring a shallow coral reef is presented, along with a comparison with traditional diving surveys.

5:00

3pABb9. A system for acoustic detection, classification, and localization of terrestrial animals in remote locations. Dean L. Hawthorne (Lab. of Ornithology, Cornell Univ., Cornell Lab of Ornithology, 159 Sapsucker Woods Rd., Ithaca, NY 14850, dh27@cornell.edu), William Horn (Coherent Tech. Services, Inc., Lexington Park, MD), and Danny C. Reinke (Environ. Management, U.S. Air Force, Edwards Air Force Base, CA)

Acoustic monitoring of endangered terrestrial animals often involves deploying observation equipment to remote locations which may be difficult or hazardous for human observers to access. We present a system designed for deployment to such locations which is capable of detecting, classifying, and localizing a wide range of vocalizations. The battery/solar powered system can continuously record data for two weeks while sending real-time Wi-Fi detection alerts. Normally, three systems, each composed of a sensor array consisting of a 16 element microphone array, an ancillary processor box, plus ancillary equipment as needed, are deployed allowing detection geolocation. The computer system uses a computationally efficient beamforming algorithm to achieve 12 dB of gain and locate detections in both azimuth and elevation. Vocalizations are classified using a template matching algorithm in the spectrogram image domain. Localization is achieved by crossing the far-field bearings obtained from the three beamformers. Output is integrated into Cornell's Raven Pro software package. Experimental trials with endangered Mojave Ground Squirrel vocalizations have resulted in successful detection and bearing estimation. [Work sponsored by Air Force SBIR AF112-193 currently in Phase II.]

5:15

3pABb10. A generic system for the automatic extraction of narrowband signals in underwater audio. Shyam Madhusudhana, Alexander Gavrilov, and Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Phys. Bldg. 301 Rm. 136A, Curtin University – Marine Sci., Kent St., Bentley, WA 6102, Australia, s.madhusudhana@postgrad.curtin.edu.au)

Narrowband acoustic signals occur prominently in underwater environments and are commonly used to monitor or track the sources producing them. Disparate *ad hoc* systems have previously been developed for the detection or recognition of several forms of narrowband signals produced by specific sources. We present a generic system, based on post-processing of spectrograms, for the automatic extraction of time-frequency contours of narrowband signals produced by any source—biological, anthropogenic, or geophysical. A two-phase approach is proposed where the first phase is based on an image-processing technique for detecting intensity ridges and the second phase is a Bayesian filtering approach for tracing the trajectory of detected ridge apices. The choice of algorithm parameters and conditionals are backed with theoretical motivations and are geared to result in a generic (non-targeted) system. In comparison to an existing method, using publicly available pre-annotated recordings containing four species of dolphins, our system offered improvements of 21% and 15% in precision and recall rates, respectively. A streaming-mode implementation in Matlab processed inputs with an average real-time factor of 0.10 on a modest desktop computer, showing that the system is well-suited for both offline and in-situapplications.

5:30

3pABb11. A novel approach to measurement of underwater sound levels in a dangerous tidal fjord using a miniature self-contained acoustic recorder. William C. Burgess (Greeneridge Sci., Inc., 6060 Graham Hill Rd. Stop F, Felton, CA 95018, burgess@greeneridge.com) and Tamara L. McGuire (LGL Alaska Res. Assoc., Inc., Anchorage, AK)

Recent plans for highway construction along the shore of Turnagain Arm, Alaska, raised concerns about acoustic impact on the critically endangered Cook Inlet subpopulation of beluga whales (*Delphinapterus leucas*). Evaluation of possible impact required estimates of normal background sound levels. However, as a shallow fjord with profound (9-m mean) tides, hazardous currents, tidal bores, and quicksand-like tidal flats, Turnagain Arm has historically defied most vessel operations and thwarted vessel-based sound measurements. In August 2014, the first documented acoustic measurements in Turnagain Arm of which we are aware took place using a novel technique based on a miniature, self-contained acoustic recorder developed as a marine-mammal tag (Acousonde^{TM}, Greeneridge Sciences, Inc.). The recorder's small size allowed it to be suspended from a motorized buoy and towed up to 80 m from shore. Measurements took place off a point known as Windy Corner at high and low tide as well as near maximum flood and ebb current. Data showed Turnagain Arm to be quieter than expected, with broadband (40 Hz to 9.3 kHz) sound pressure levels as low as 74 dB re 1 μPa and below a 125-dB provisional regulatory assumption. [Work supported by Alaska Department of Transportation and Public Facilities.]

Session 3pAO

Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Ocean Acoustic Tomography: Active and Passive, Theory, and Experiment III

Bruce Howe, Cochair
Ocean and Resources Engineering, University of Hawaii, 2540 Dole Street, Holmes Hall 402, Honolulu, HI 96822

Arata Kaneko, Cochair
Graduate School of Engineering, Hiroshima University, 1-4-1 Kagamiyama, Higashi-Hiroshima 739-8527, Japan

Hiroyuki Hachiya, Cochair
Tokyo Institute of Technology, 2-12-1 S5-17, Ookayama, Meguro-ku, Tokyo 152-8550, Japan

Contributed Papers

1:00

3pAO1. Application of coastal acoustic tomography to Lake Biwa, Japan. John C. Wells, Yasuaki Aota, Guillaume Auger (Civil Eng., Ritsumeikan Univ., Noji Higashi 1-1-1, Kusatsu, Shiga 525-8577, Japan, jwells@se.ritsumei.ac.jp), Arata Kaneko, and Noriaki Goda (Hiroshima Univ., East Hiroshima, Japan)

We will provide an update on continuing development of coastal acoustic tomography (CAT) in Lake Biwa, Japan, which will provide monitoring data for assimilation into a prototype “nowcast system” for the flow and temperature fields in the lake. Employing 5 kHz transducers, we have achieved a number of successful CAT tests since August 2014 in the North Basin, which has a mean depth of 41 meters. These were performed under both unstratified and stratified conditions, at transmission distances up to 10 km, and included a continuous 2-month deployment in homogenized winter conditions. Due to weaker mixing than coastal seas, stratification-induced refraction in lakes is stronger than normally encountered in CAT. We believe these to be the first successful tests of acoustic tomography ever reported for a lake. Received signals were remarkably different between the nearshore station with bottom depth about 9 m, and offshore of Takeshima island with bottom depths around 35 m. Current efforts aim to improve SNR for reception offshore of Takeshima island.

1:15

3pAO2. The application of coastal acoustic tomography to a large experimental wave/current basin. Guangming Li, David Ingram (Inst. of Energy Systems,School of Eng., Univ. of Edinburgh, Faraday Bldg., King’s Buildings,Colin Maclaurin Rd., Edinburgh, Scotland EH9 3DW, United Kingdom, G.Li@ed.ac.uk), Arata Kaneko, Noriaki Gohda (Graduate School of Eng., Hiroshima Univ., Hiroshima, Japan), and Nick Polydorides (Inst. of Digital Communications, School of Eng., Univ. of Edinburgh, Edinburgh, United Kingdom)

This article describes the use of acoustic tomography to characterize the flow pattern in the Flowave basin. Flowave is a unique circular wave/current basin, 2 m deep and 25 m in diameter. Current can be created in any direction at speed of up to 1.2 $ms.^{-1}$. Two coastal acoustic tomography (CAT) stations were used operating at 50 kHz with a M- sequence signal. By combining three different pairs of station positions and seven current directions a network of seven CAT stations has been emulated. Throughout the test, a current speed of 0.8 ms^{-1} was used. Using the inverse Fourier method, a velocity profile has been obtained. During the experiment, a Nortek vectrino velocimetry was placed between the stations. This paper will present the comparison between the vectrino and CAT measurement.

1:30

3pAO3. 3D assimilation of Hiroshima Bay acoustic tomography data into a Princeton Ocean Circulation Model. Minmo Chen, Arata Kaneko (Graduate School of Eng., Hiroshima Univ., 1-4-1 Kagamiyama, Higashi-Hiroshima, Hiroshima 739-8527, Japan, d153155@hiroshima-u.ac.jp), Chuangzheng Zhang (State Key Lab. of Satellite Ocean Environment Dynam., Second Inst. of Oceanogr., Hangzhou, Zhejiang, China), and Ju Lin (College of Information Sci. & Eng., Ocean Univ. of China, Qingdao, Shandong, China)

A baroclinic data assimilation, based on the ensemble Kalman filter (EnKF) method, with coastal acoustic tomography (CAT) data is first applied to Princeton Ocean Model (POM) to elucidate a 3D structure of coastal upwelling which occurred in Hiroshima Bay due to a northerly wind from a typhoon. The CAT experiment with four acoustic stations was performed from Sept. 11 to Sept. 25, 2013, in the northern part of Hiroshima Bay, Japan. Reciprocal data were acquired at three stations and along three transmission lines with a triangular array and travel time were identified every 10 min for the first and second arrival rays which travel at different depth ranges. As for the EnKF method, the Kalman gain was determined to minimize the observed and simulated travel time differences and summations. The model error covariance was given by adding pseudorandom noises (zero mean value and one variance) with an appropriate multiplication factor to the POM result. The observation error covariance was estimated from the variation of observed travel times. A 3D structure of coastal upwelling which was difficult to be simulated by the POM was reconstructed by data assimilation and compared with the CTD result.

1:45

3pAO4. Feasibility of geoacoustic tomography in shallow water. Altan Turgut, Jeffrey A. Schindall, and Steven L. Means (Naval Res. Lab, Acoust. Div., Code 7160, Washington, DC 20375, altan.turgut@nrl.navy.mil)

In shallow water, spatial and temporal variability of the water column often restricts accurate estimations of bottom properties from long-range acoustic propagation data. Two recent shallow-water experiments, conducted at a sandy site on the New Jersey Shelf and a muddy site on the New England Shelf, showed 5-10 dB acoustic transmission-loss differences between two sites. The New Jersey Shelf experiment was conducted during May 2015 and The New England Shelf experiment was conducted during November 2015, representing summer and winter propagation conditions, respectively. Acoustic modal analysis and pulse-decay tomography methods were used to map the geoacoustic variability of each site covering a 40 km by 40 km area. Both numerical simulations and the New England Shelf

experimental results showed that geoacustic variability can be measured by geoacoustic tomography. Tomographic results from the New Jersey Shelf sites showed less geoacoustic variability that may be due to the presence of a harder bottom. Effects of the water-column variability on the geoacoustic tomography results are also discussed for summer and winter environments. [Work supported by the ONR.]

2:00

3pAO5. Current field tracking based on spatiotemporal evolution model with distributed networked underwater sensor system. Wen Xu, Ying Zhang, Huifang Chen, and T. C. Yang (Zhejiang Univ., No. 38, Zheda Rd., Hangzhou 310027, China, wxu@zju.edu.cn)

Distributed networked underwater sensor (DNUS) system can provide ocean measurements over a wide area with a large number of sensors. This paper studies the estimation of the two-dimensional ocean current field with acoustic travel time difference data observed by a DNUS system. Considering that the current field is correlated in time and space, we present a statistical-based tomography method based on Kalman filter to reconstruct and track the ocean current field. A spatiotemporal autoregressive (AR) model is used to describe the evolution of the current field. In the spatiotemporal AR model, the observation region is divided into sub-triangle grids. The sub-triangles are partitioned into clusters by distance and each cluster is assigned with one AR coefficient. The AR coefficients are updated adaptively with the past estimated current velocities. The proposed method is verified with the synthetic observational data generated by a barotropic ocean model and the tomography experiment conducted near Zhoushan Island, China. Compared with the regular time-independent distributed processing method, the proposed ocean current field tracking achieves a lower root-mean-square error. In addition, the proposed method is robust to the measurement link failure and burst errors commonly occurred in a DNUS system.

2:15

3pAO6. Rip current tomography based on reciprocal sound transmissions using reflective waves. Iwao Nakano (Marine Acoust. Res. Lab. of Japan, 5-12-63-101 Yashio, Shinagawa-Ku, Tokyo 140-0003, Japan, nakano-family@s7.spaaqs.ne.jp), Hiroshi Ishida (Oshima College, National Inst. of Technol., Japan, Suooshima-cho, Japan), and Ichiro Deguchi (Osaka Univ., Japan, Suita, Japan)

In order to monitor the occurrence of rip current in beaches, an acoustic measurement system for rip currents based on passive reciprocal sound transmissions using reflective waves was designed and developed. The offshore current velocity V is given by $V = C_0^2/L*(t_{21}-t_{12})/2$ where C_0 is sound speed, L is the distance between the transducers, t_{21} is travel time at onshore transmission, and t_{12} is travel time at offshore transmission. The typical components of the measurement system consisted of one modified dual-frequency sounder, two broad-band acoustic transducers, six sound reflectors, and a data acquisition network. The frequency, source level, and transmission interval of transmitted signal was 50 kHz, 1 kW (max), and 2 s, respectively. The typical monitoring area is 30 m wide and 90 m long, and the depth is 1 m. This system simply required the precise position measurement of two transducers and sound reflectors. The sound speed was determined by the direct travel times between the transducers. When the field tests were conducted at Uradome Beach in Tottori Prefecture of Japan in September of 2014, the phases of sound signals were so stable to allow the automatic peak and/or phase tracking. The obtained and reconstructed velocity data were consistent with those by EM current meters.

2:30

3pAO7. Multipath propagation of sound in a shallow tidal channel and its implications on tomographic current measurements. Mahdi Razaz, Len Zedel (Physical Oceanogr., Memorial Univ. of NF, 500 Prince Philip Dr., St. John's, NF A1B 3X7, Canada, mrazaz@mun.ca), Alex Hay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada), and Kiyosi Kawanisi (Civil and Environ. Eng., Hiroshima Univ., Higashi-Hiroshima, Hiroshima, Japan)

A pilot observational experiment with Fluvial Acoustic Tomography (FAT) system was conducted in the Grand Passage, Nova Scotia, Canada, in 2014, to assess the capabilities of FAT in continuous monitoring of transport in a tidal channel. To implement the tomographic measurements, two broadband FAT transceivers operated at 7 kHz central frequency were positioned in both sides of the channel emitting a pulse every 30 s for 4 days. Three coherent arrivals were identified in the acoustic receptions. This paper investigates the influence of physical characteristics of water and current shear on multipath propagation of sound in well-mixed conditions of the channel. At the end, we report the comparison between FAT and reference velocity data collected by a moving-boat ADCP.

2:45

3pAO8. Reliable acoustic path tomography at Aloha Cable Observatory. Vincent J. Varamo and Bruce Howe (Ocean and Resources Eng., Univ. of Hawaii - Manoa, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822, varamo@hawaii.edu)

Using a mobile ship platform (R/V *Kilo Moana*) with an acoustic source transmitting to a fixed bottom hydrophone at the ALOHA Cabled Observatory (ACO), we are investigating the feasibility of "RAP tomography" (Reliable Acoustic Path). This will allow the spatial mapping of the path integrated sound speed (temperature) over a 60 km diameter "teacup" volume of the ocean. This can be considered an extension of an inverted echosounder (from a vertical to near horizontal path) combined with the precise positioning and timing of seafloor geodesy, where noise for the latter, sound speed, is our signal. As a first step, transmissions were sent from a 4x4 array of transducers at 3.5 kHz when the vessel approached and departed from ACO. The measured travel times were compared with estimated travel times based on CTD measurements. As the next step, we use new instrumentation to ensure exact time of acoustic transmission and to test user-generated signals (LFM sweeps and M-sequences) to improve the time of arrival resolution. Also, to increase the amount of shallow angle energy for the longer ranges, only one transducer with a broader beamwidth will be used. Results from an August 2016 experiment will be presented. [Work supported by ONR.]

Session 3pBAa

Biomedical Acoustics: Quantitative Ultrasound II: From Micro to In-vivo

Jonathan Mamou, Cochair
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Invited Paper

1:00

3pBAa1. Mutual interpretation between B-mode image and cross-sectional acoustic microscopy. Naohiro Hozumi, Wei-Chean Tan, Sachiko Yoshida (Toyohashi Univ. of Technol., 1-1 Hibarigaoka, Tenpaku, Toyohashi, Aichi 441-8580, Japan, hozumi@icceed.tut.ac.jp), Yuki Ogura (Shiseido, Yokohama, Japan), Kazuto Kobayashi (Honda Electronics, Toyohashi, Japan), and Tadashi Yamaguchi (Chiba Univ., Chiba, Japan)

Acoustic microscopy is a two-dimensional profile of an acoustic parameter such as sound speed and characteristic acoustic impedance. It may be useful to understand how practically important B-mode image looks depending on cross-sectional spatial distribution of acoustic properties. In the presentation, it will be exhibited that B-mode echo-graph can be briefly reproduced from acoustic impedance profile observed by acoustic microscope by a similar process to the time domain reflectometry (TDR), with some simple assumptions. As an example, the acoustic impedance profile of a rat cerebellar tissue was acquired by acoustic impedance microscope that was developed in our lab. The B-mode image reproduced by calculation was compared with an actually acquired B-mode image. A good agreement between these two images was seen. On the other hand, a B-mode profile may be interpreted into cross-sectional distribution of acoustic profile. As it is attributed to inverse problem, several assumptions may be required. In the calculation, strait acoustic beam and no attenuation were assumed. In addition, the B-mode profile with RF waveforms was assumed to be available. In the presentation, it will be exhibited that cross-sectional acoustic impedance profile of a shallow region of human skin can be reproduced from non-invasive B-mode scan.

Contributed Papers

1:20

3pBAa2. Repeatability and reproducibility of ultrasonic attenuation coefficient and backscatter coefficient measures in adults with non-alcoholic fatty liver disease. Aiguo Han (Univ. of Illinois, Urbana, IL), Michael P. Andre, Rohit Loomba, Claude B. Sirlin (Dept. of Radiology, Univ. of California at San Diego, 408 Dickinson St., San Diego, CA 92103, mandre@ucsd.edu), John W. Erdman, Jr., and W. D. O'Brien, Jr. (Univ. of Illinois, Urbana, IL)

The attenuation coefficient (AC) and backscatter coefficient (BSC) are two promising quantitative ultrasound biomarkers, yet their repeatability and reproducibility (R&R) have never been assessed in human subjects. This study assesses R&R of the AC and BSC in 30 adults with suspected non-alcoholic fatty liver disease (NAFLD), a wide liver fat content range, age range of 28-71, using a Siemens S3000 scanner. Each subject had two consecutive scans between which the subject was repositioned. In each trial, the 4C1 and 6C1HD transducers were used for multiple B-mode/RF acquisitions of the right lobe of the liver. Then, a reference phantom was scanned using the same transducer and machine settings. Two sonographers each scanned 15 patients independently. Two fields of interest (FOIs; with and without visible blood vessels) were drawn for each image. AC and BSC were estimated within the FOIs. AC and log-transformed BSC mean values within 2.6-3.0 MHz were used for the R&R analysis based on an unweighted sums of squares ANOVA approach. The variabilities of inter-trial, inter-image, inter-FOI, and inter-transducer conditions were found to be lower than the inter-patient variability. The inter-transducer variability was the lowest among all analyzed variables. [Support: NIH R01DK106419.]

1:35

3pBAa3. A method for estimating the 3-D structure function from 2-D histology sections. Aiguo Han (BioAcoust. Res. Lab., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801, han51@illinois.edu)

Understanding the fundamental scattering mechanism(s) in tissues is critical to the success of Quantitative Ultrasound (QUS) imaging. From a discrete scatterer point of view, the ultrasound scattered power depends not only on the properties of the individual scatterers (modeled by the form factor) but also on the spatial correlation among the scatterers (modeled by the structure function). Previously, the structure function was obtained from the ultrasound backscatter data for dense cell pellet biophantoms. This study introduces a method for obtaining the 3-D structure function from the digitized hematoxylin & eosin (H&E) stained histology images. The method starts with calculating the 2-D pair correlation function using the scatterer positions determined from a histology image. Assuming isotropic media with spherical scatterers, the 3-D pair correlation function is then estimated by numerically solving a stereological function that relates the 3-D pair correlation function with the 2-D pair correlation function. The 3-D structure function is finally calculated through a Fourier-Bessel transformation of the 3-D pair correlation function. The proposed method was validated through computer simulations and cell pellet biophantom experiments. Using this method to estimate 3-D structure functions may lead to improved understanding of acoustic scattering in biological media. [Support : R01CA111289 and R01DK106419.]

1:50

3pBAa4. Quantitative ultrasound and the pancreas. Rita J. Miller, Aiguo Han, John W. Erdman, Jr., Matthew A. Wallig, and William D. O'Brien, Jr. (Univ. of Illinois, BioAcoust. Res. Lab., 405 N Mathews Ave., Urbana, IL 61801, rjmille@illinois.edu)

Pancreatitis is inflammation of the pancreas. Cerulein-induced pancreatitis in animal models can cause a significant increase in pancreatic enzymes in blood, as well as interstitial edema and inflammatory cell infiltration in the pancreas. This degree of pancreatitis is mild; all animals survive the induction of pancreatitis and resolves within ~6 days after induction, making this an excellent model to study the attenuation coefficient (AC: dB/cm-MHz) and backscatter coefficient (BSC: 1/cm-sr) of the pancreas temporally. The edematous stroma and shrinkage and dedifferentiation of acinar cells, has certain similarities with the morphology in some forms of pancreatic carcinoma. The pancreas' AC and BSC (20-50 MHz) were estimated *in vivo* and ex vivo at baseline (before cerulein injections) and at 24 h, 48 h and 72 h after cerulein injections in Sprague-Dawley rats (N = 24). AC and BSC showed the same trends whereby (relative to baseline values) the 24-h AC decreased ~0.3 dB/cm-MHz and BSC decreased ~1-2 dB with AC and BSC progressively approaching the baseline values at 48 h and 72 h, suggesting that as a function of time, the AC and BSC are reflecting a change in the pancreas back to baseline conditions. [Support: NIH Grant R37EB002641.]

2:05

3pBAa5. Multi-parameter analysis of bladder mechanical properties using ultrasound bladder vibrometry. Matthew T. Huber (Dept. of Phys., Rhodes College, 2000 North Parkway, Rhodes Box 2644, Memphis, TN 38112, hubmt-18@rhodes.edu), Aparna Signh, Mathew Cheong (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN), Matthew Urban (Dept. of Radiology, Mayo Clinic College of Medicine, Rochester, MN), Mahdi Bayat, and Mostafa Fatemi (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN)

Bladder wall mechanical properties are important indicators of bladder compliance. This study compares the effectiveness of the Kelvin-Voigt, Maxwell, and fractional Kelvin-Voigt rheological models in capturing the mechanical properties of normal (compliant) and aberrant (non-compliant) ex-vivo pig bladder walls. Bladders were filled to different volumes, varying their thickness and internal pressure, and excited by the acoustic radiation force. Pulse-echo ultrasound imaged anti-symmetrical Lamb waves traveling along the bladder wall. Two-dimensional Fourier analysis generated velocity-frequency dispersion curves for the Lamb wave propagation. Fitting these experimental curves against theoretical curves from the models yielded values for bladder elasticity and viscosity. Pressure, measured simultaneously, was used as a point of comparison with elasticity. Our findings indicate an increasing trend in both elasticity for the rheological models and recorded pressure with increasing volume. This rate was observed to be significantly higher in aberrant bladders than intact bladders for the Maxwell and fractional Kelvin-Voigt models (p-value of 0.0082). Additionally, all three models showed a similar strong correlation (greater than 0.9) between pressure and elasticity. These findings provide a foundation for selecting an appropriate bladder wall rheology model which can potentially be used in evaluation of human bladder mechanical properties. [Work supported by NIH grant DK99231.]

2:20

3pBAa6. Comparison of quantitative ultrasound parameters from two approaches on *in vivo* human fatty liver. Pauline Muleki-Seya, Aiguo Han (Univ. of Illinois, BioAcoust. Res. Lab., 405 N Mathews Ave., Urbana, IL 61801, muleki@illinois.edu), Michael P. Andre, Rohit Loomba, Claude B. Sirlin (Univ. of California at San Diego, San Diego, CA), John W. Erdman, Jr., and William D. O'Brien, Jr. (Univ. of Illinois, Urbana, IL)

Two different quantitative ultrasound (QUS) processing strategies are evaluated and compared. The first method estimates the Lizzi-Feleppa (LF) parameters from the linear fit of the normalized power spectrum, and yields the slope, the intercept, and the midband. The QUS approach relies on the attenuation and BSC versus frequency, and yields the mean attenuation and BSC. By using the spherical Gaussian model, the effective scatterer diameter (ESD) and acoustic concentration (EAC) are estimated from the BSC. The goal of this study is to compare the LF and QUS parameters to determine how they are correlated and to verify their correlation with ESD and EAC. To do this, RF data were acquired using a Siemens S3000 from 20 patient livers (suspected of Non-Alcoholic Fatty Liver Diseases (NAFLD)) and from these spectra, the LF and QUS parameters were estimated. The liver fat fraction for each patient was estimated using MRI, and ranged from 0.03 to 0.3. The results show that LF slope is correlated to ESD ($R^2=0.75$), LF midband to EAC ($R^2=0.70$), and LF intercept to ESD and EAC. A very high correlation between LF midband and the mean BSC is observed ($R^2=1.00$), along with the fat fraction ($R^2=0.63$). [Support: NIH R01DK106419.]

2:35

3pBAa7. Design of transmission-mode measurements for estimating ultrasound attenuation and nonlinearity in liver. Wayne Kreider, Christopher Hunter (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Vera A. Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Brian MacConaghy, and Yak-Nam Wang (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Previously, it has been shown that ultrasound measurements of sound speed and nonlinearity can be used to quantify the fatty and non-fatty components of liver tissue. In addition, it has been proposed that ultrasound attenuation measurements can be used to distinguish fatty components comprising either sub-micron lipid droplets (microsteatosis) or much larger cell-sized droplets (macrosteatosis). To perform all of these measurements, a caliper device is being developed based on a transmission-mode approach. Design challenges are posed by competing requirements: sub-megahertz frequencies for optimal detection of microsteatosis and nonlinearity estimation based on waveform distortion over relatively short distances. To meet these challenges, a design is proposed in which ultrasound at 667 kHz is generated by a 50 mm piezoceramic disk and all measurements are made in the plane-wave regime. Given geometrical and physical constraints, all analysis was performed in the time domain. The nonlinearity coefficient was accurately estimated from a quasi-linear approximation to the lossless Burgers equation for pressure amplitudes less than 1 MPa and propagation distances from 15 to 50 mm. An analogous time-domain approach was implemented for determining attenuation at the fundamental frequency and several odd harmonics. [Work supported by NIH grants EB017857, EB007643, EB016118, and DK104854.]

2:50–3:05 Break

3:05

3pBAa8. Characterization of the lung parenchyma using ultrasound multiple scattering. Kaustav Mohanty (Dept. of Mech. and Aerosp. Eng., College of Eng., North Carolina State Univ., 3147 B, 911 Oval Dr., EB-3, Raleigh, NC 27606, kmohant@ncsu.edu), John Blackwell, Thomas Egan (Div. of Cardiothoracic Surgery, Univ. of North Carolina, Chapel Hill, NC), and Marie Muller (Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

We use multiple scattering of ultrasound waves to characterize the lung micro-architecture in order to differentiate between a healthy lung and a lung suffering from Alveolar Interstitial Lung Diseases. The experimental setup consists of a linear transducer array with an 8 MHz central frequency placed in direct contact of the lung to be assessed. The diffusion constant D and scattering mean free path L* of the lung parenchyma are estimated by separating the incoherent and the coherent intensities in the near field. 2D FDTD numerical simulations were carried out on rabbit histology images with varying degrees of lung collapse. Phantom experiments were conducted in melamine sponges to study the variations in D and L* with varying air volume fraction. Significant correlations were observed between air volume fraction and L* in simulation ($r = -0.9542$, $p<0.0117$) and sponge phantom experiments ($r = -0.9932$, $p<0.0068$). Finally, *in vivo* measurements were conducted in healthy and edematous rat lungs. In the control rat lung, L* was found equal to 83 μm (+/-14.9), whereas in the edematous lung, it was found equal to 260 μm (+/-27). These results are extremely promising for the assessment of lung pathologies using ultrasound.

3:20

3pBAa9. *In-vivo* quantitative analysis of the angiogenic microvasculature in tumor-bearing rats using multiple scattering: A preliminary study. Aditya Joshi (Mech. & Aerosp. Eng., North Carolina State Univ., 2500 Avent Ferry Rd., Apt. No. 203, Raleigh, NC 27606, aajoshi4@ncsu.edu), Sarah Shelton, Virginie Papadopoulou, Brooks Lindsey, Paul Dayton (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC), and Marie Muller (Mech. & Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

We propose a method to quantify the vascular density in vascular networks using contrast-enhanced multiple scattering. We measured the diffusion constant D and transport mean free path L* from the time evolution of the incoherent intensity in a rat model of cancer. An 8 MHz linear transducer array was used to record the backscattered signals from subcutaneous fibrosarcoma tumors and control tissue. The coherent and incoherent contributions to the backscattered intensity were separated, and the growth rate of the incoherent contribution was measured, giving the D and L*, knowing the effective speed of sound. By translating the linear array along the tumor, mapping of L* was achieved. Tumors were implanted in the right flank of four rats, and the contralateral side served as control. Acoustic angiography and measurements of the incoherent intensity were performed. The mean L* values in control and tumor tissue were significantly different (105.27 +/-30.96 micron and 41.28+/- 14.23 micron, respectively, $p = 8.4033 \times 10^{-49}$). The mean distance between vessels was estimated from acoustic angiography images using Monte-Carlo simulations, and was in agreement with the experimentally calculated values of L* ($r = 0.9507$, $p = 1.4957 \times 10^{-9}$).

3:35

3pBAa10. Numerical evaluation of time-domain Green's functions for space-fractional wave equations. Xiaofeng Zhao and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48824, mcgough@egr.msu.edu)

Ultrasound attenuation in soft tissue follows a power law as a function of the ultrasound frequency. Several different models for the power law attenuation of medical ultrasound have been developed using fractional calculus, where each contains one or more time- or space-fractional derivatives. For certain time-fractional models, exact and approximate time-domain Green's functions have been derived and evaluated numerically, but a similar analysis has not yet been performed on the space-fractional models. To address this deficiency, time-domain Green's functions are numerically calculated here for two space-fractional models, namely the Chen-Holm and Treeby-Cox space-fractional wave equations. Numerical results are computed for both of these in breast and liver with power law exponents of 1.5 and 1.139, respectively. The results show that these two space-fractional wave equations are causal everywhere. Away from the origin, the time-domain Green's function for the dispersive Treeby-Cox space-fractional wave equation is very similar to the time-domain Green's functions calculated for the corresponding time-fractional wave equations, but the time-domain Green's function for the nondispersive Chen-Holm space-fractional wave equation is quite different. To highlight the similarities and differences between these, time-domain Green's functions are compared and evaluated at different distances for both breast and liver.

3:50

3pBAa11. Fatty-acid species identification by quantitative high-frequency acoustic impedance measurements. Kazuyo Ito (Graduate School of Eng., Chiba Univ., 1-33 Yayoicho, Inage-ku, Chiba, Chiba 2638522, Japan, k_ito@chiba-u.jp), Kenji Yoshida (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Chiba, Japan), Hitoshi Maruyama (Graduate School of Medicine, Chiba Univ., Chiba, Chiba, Japan), Jonathan Mamou (Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), and Tadashi Yamaguchi (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Chiba, Japan)

Accurate discrimination of non-alcoholic steatohepatitis (NASH) from simple steatosis is a critical issue in current clinical practice because NASH can progress to cirrhosis or even liver cancer. Some studies reported that the free fatty acids (FFA) composition of the liver changes with the progression of NASH. Therefore, an ultrasound-based diagnostic tool could be based on measurements of the acoustical properties of FFAs. This study investigated how FFAs influence the acoustical properties of liver cells at a microscopic level using scanning acoustic microscopy (SAM) with four pathological types of mouse livers, five kinds of FFAs in solution, and five kinds of FFAs in cultured cells. A SAM system based on an 80-MHz center-frequency transducer was employed. The acoustic impedance was computed from the echo amplitude and the pressure-reflection coefficient. One-way ANOVA showed statistically significant differences ($p < 0.01$) in acoustic impedance among pathological types. NASH was found to have the lowest acoustic impedance. Results obtained in FFA solutions and FFA-containing cultured cells also demonstrated significant differences in acoustic-impedance values ($p < 0.01$) for all paired tests except one (palmitate and linoleate). Therefore, these initial results suggest that NASH identification and characterization using *in-vivo* quantitative ultrasound methods may be possible.

4:05

3pBAa12. 500-MHz scanning acoustic microscopy of cancerous human lymph nodes. Jonathan Mamou, Daniel Rohrbach (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@rri-usa.org), Emi Saegusa-Beecorft, and Junji Machi (General Surgery, Univ. of Hawaii and Kuakini Medical Ctr., Honolulu, HI)

Scanning acoustic microscopy (SAM) at 500 MHz permits forming 2D maps of the speed of sound (SOS), attenuation (A) and acoustic impedance (Z) of tissue microstructure with a spatial resolution of 4 μm. Maps from cancerous lymph nodes (LNs) can improve understanding of ultrasound scattering at 25 MHz and thereby can improve the specificity and sensitivity of quantitative ultrasound (QUS) methods for clinically significant micrometastases. SAM data were acquired from ten fixed LNs from colorectal-cancer patients. Deparaffinized, 6-μm thick sections were scanned using a custom-built SAM equipped with a 500-MHz transducer having a 264-MHz bandwidth and a 4-μm lateral beamwidth. Reflected signals were digitized at 2.5 GHz and maps of Z, A, and SOS were generated. Histological thin sections adjacent to the scanned samples were stained using hematoxylin and eosin, and metastatic regions were demarcated on the photomicrograph. The 4-μm spatial resolution permitted distinguishing metastases from other tissue types. The average Z for metastases was 1.66 ± 0.06 Mrayl, which was significantly larger than the average Z of surrounding tissue. Spatial variations in the Z maps will translate into new ultrasound-scattering form factors, which are expected to improve QUS approaches for detecting metastases in LNs. [NIH Grant R21EB016117.]

4:20

3pBAa13. High-frequency ultrasound study of angiogenesis in zebra fish embryos. Dolly A. Sanjinez, Michaelle A. Cadet (Biology, Utah Valley Univ., 496 N 400 E, Orem, UT 84097, dolly.sanjinez@gmail.com), Garrett Wagner (Comput. Eng., Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Cancer and other prevalent diseases such as heart disease, rheumatoid arthritis, stroke, and diabetic retinopathy are directly associated with either an increase or decrease in levels of angiogenesis in the body. High-frequency (HF) ultrasound (10-100 MHz) is particularly sensitive to small vascular structures that are close in size to the ultrasound wavelength (15-150 μm). The ability to rapidly determine the degree of vascularization in the development of small animals *in vivo* would provide a useful characterization tool for the study of a variety of diseases. The objective of this study was to determine if HF ultrasonic measurements could identify microscopic vessel formation that occurs during the development of zebra fish embryos within 1-3 days post-fertilization (dpf). Casper and P53 Nacre zebra fish embryos were provided by the Stewart Lab at the Huntsman Cancer Institute (Salt Lake City, Utah). The embryos were placed in 12-well cell culture plates, embedded in agarose and egg water, and immobilized with tricaine. Embryos were positioned on their lateral side and tested 1-2 dpf using 50-MHz pulse-echo measurements. The results show that the number of peaks in the ultrasonic spectra correlate with embryo development and may provide a quantitative assessment method for angiogenesis.

4:35

3pBAa14. A quantitative imaging biomarker alliance. Timothy J. Hall (Medical Phys., Univ. of Wisconsin, 1005 WIMR, 1111 Highland Ave., Madison, WI 53705, tjhall@wisc.edu), Brian S. Garra (Radiology, Washington DC VA Medical Ctr., Washington, DC, DC), Paul L. Carson (Radiology, Univ. of Michigan, Ann Arbor, MI), Andy Milkowski (Ultrasound Div., Siemens Healthcare, Issaquah, WA), J B. Fowlkes, Oliver Kripfgans (Radiology, Univ. of Michigan, Ann Arbor, MI), Richard G. Barr (Southwoods Imaging Ctr., Boardman, OH), and Mike Averkiou (Biomedical Eng., Univ. of Washington, Seattle, WA)

The Radiological Society of North America (RSNA) created the Quantitative Imaging Biomarker Alliance (QIBA) to convert "imaging systems" with subjective image interpretation to "measurement systems" with objective measurands having actionable numerical values. The goal is to create QIBA protocols and profiles that specify methods for data acquisition, analysis, and interpretation. QIBA profiles also provide specific claims of what can be accomplished by following the QIBA protocol. The intent is to validate the profile across imaging systems in a collaboration between industry, government (NIH, FDA, NIST), Pharma, clinicians, and academics. The RSNA/QIBA effort includes over 850 volunteers worldwide with Biomarker Committees for all major imaging modalities. The effort is expanding with collaborations in Japan (the Japan Radiological Society's QIBA-Japan) and in Europe (European Institute for Biomedical Imaging Research and the European Imaging Biomarker Alliance; EIBIR—EIBALL). The QIBA ultrasound group began with a committee to investigate and minimize sources of bias and variance in shear wave speed estimates for liver fibrosis assessment. The group has grown to about 200 members and includes committees to advance biomarkers based on measurement of volume blood flow and parameters derived from contrast-enhanced ultrasound. The status and implications of these efforts will be presented.

WEDNESDAY AFTERNOON, 30 NOVEMBER 2016 — CORAL 3, 1:00 P.M. TO 3:00 P.M.

Session 3pBAb

Biomedical Acoustics and Signal Processing in Acoustics: Bone Sonometry II: Ultrasonic Studies of the Physical Properties of Cortical and Cancellous Bone (Poster Session)

James G. Miller, Cochair
Physics, Washington U Saint Louis, Box 1105, 1 Brookings Drive, Saint Louis, MO 63130

Mami Matsukawa, Cochair
Doshisha University, 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan

All posters will be on display from 1:00 p.m. to 3:00 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 2:00 p.m. and authors of even-numbered papers will be at their posters from 2:00 p.m. to 3:00 p.m.

Contributed Papers

3pBAb1. Finite-difference time-domain simulations of piezoelectric effect in bone under ultrasound irradiation. Atsushi Hosokawa (Dept. of Elec. and Comput. Eng., National Inst. of Technol., Akashi College, 679-3 Nishioka, Uozumi, Akashi 674-8501, Japan, hosokawa@akashi.ac.jp)

To understand the piezoelectricity in bone at ultrasound frequencies, numerical simulations of piezoelectric effect under ultrasound irradiation was performed using an elastic finite-difference time-domain method with piezoelectric constitutive equations (PE-FDTD method). First, the effect of piezoelectricity on ultrasound wave propagation in a piezoelectric material was investigated with consideration of conductivity. Ultrasound pulsed waveforms propagating through piezoelectric ceramics, whose piezoelectric parameter values were known, were investigated by the PE-FDTD simulations. In the simulated results, the ultrasound speed in the ceramics increased with piezoelectricity, and the ultrasound attenuation caused by the piezoelectricity increased with conductivity. The simulated waveforms were in good agreement with the experimental waveforms, which demonstrated the validity of the PE-FDTD simulations. Next, the piezoelectric effect on ultrasound wave propagation in cortical bone was simulated. The piezoelectric effect on ultrasound wave propagation in the bone was scarcely observed because of the very low piezoelectricity. Finally, the piezoelectric potential induced in cortical bone by ultrasound irradiation was simulated. The simulated results showed that the piezoelectric potential decreased with conductivity. Moreover, using the cortical bone specimen, on the surfaces of which positive and negative electrodes were deposited, the piezoelectric potential was experimentally observed to compare with the simulated results.

3pBAb2. Effect of intracortical bone properties on the guided wave propagation using low-frequency axial transmission technique: Simulation study. Daniel Pereira (Mech. Eng., École de technologie supérieure, 1100 Rue Notre-Dame O, Montreal, QC H3C1K3, Canada, pereira.ufrgs@gmail.com), Guillaume Haïat (Laboratoire Modélisation et Simulation Multi Echelle, Paris, France), Julio Fernandes (Ctr. de recherche de l'Hôpital du Sacré-Cœur de Montréal, Montreal, QC, Canada), and Pierre Belanger (Mech. Eng., École de technologie supérieure, Montreal, QC, Canada)

In this work, a semi-analytical finite-element method was on the propagation of acoustic guided wave. The simulations were performed with a realistic bone cross-sectional geometry by considering an axial transmission configuration. A point source excitation was applied using a toneburst centered at 35 kHz as the input signal. The propagating waves were monitored using out-of-plane displacement in the axial direction at 32 positions separated 4 mm apart. The properties of the endosteal region of cortical bone were varied from healthy to early osteoporotic conditions. The use of the two-dimensional Fourier transform (2DFFT) allowed the measurement of individual mode velocity, which was compared to the velocity typically measured using first arrival signal (FAS). The results have shown that variations in the velocity measured using 2DFFT were mainly associated to a low order flexural-like mode among the eight modes supported by the waveguide. Furthermore, the measured velocity decreased approximately 10/% due to a degradation of 20/% of the biomechanical properties in the endosteal region. In contrast, the FAS velocity has shown to be affected by four modes, showing a non-linear variation on the velocity. The flexural mode measured using 2DFFT can therefore be considered promising to monitor changes in the endosteal region.

3pBAb3. A fast signal processing technique for characterizing Lamb wave propagation in viscoelastic cortical long bones using one transmitter and two receivers. Shigeaki Okumura (Graduate School of Informatics, Kyoto Univ., Yoshida-Honmachi, Sakyo-ku, Kyoto, Kyoto 6068501, Japan, sokumura@sato-lab.0t0.jp), Vu H. Nguyen (Laboratoire Modélisation et Simulation Multi Echelle, Université Paris-Est, MSME UMR 8208 CNRS, Créteil, France), Hirofumi Taki (Graduate School of Biomedical Eng., Tohoku Univ., Sendai, Miyagi, Japan), and Toru Sato (Graduate School of Informatics, Kyoto Univ., Kyoto, Kyoto, Japan)

Axial transmission technique is useful in assessing bone quality. Most conventional techniques require multiple transmitters and receivers. Decreasing the number of elements and computational complexity are desired for low-cost diagnosis. We propose a technique that characterizes the Lamb wave using one transmitter and two receivers. Because the transfer function of the Lamb wave is mainly determined by the longitudinal wave velocity (LWV), shear wave velocity (SWV), and bone thickness, these parameters may be determined in a fitting procedure using the transfer function. The number of combinations of these parameters is enormous, and we thus select the candidates for these parameters using spatial domain interferometry. The method estimates the phase velocity of zero-th order anti-symmetric mode. We applied the proposed method to numerical simulation data generated by the semi-analytical finite element method. We employed a viscoelastic plate with LWV, SWV, and thickness of 2120 m/s, 4430 m/s, and 1.0 mm, respectively. The estimation errors of proposed method for SWV, LWV, and thickness were within 5.0 m/s, 15 m/s, and 0.05 mm, respectively. The fitting residue was −18 dB. The calculation time was 3.6 s. These results demonstrate the high potential of the proposed method in the low-cost diagnosis of bone quality.

3pBAb4. Simulation study of ultrasound propagation in anisotropic and heterogeneous cortical bone model. Koki Takano (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, duq0366@mail4.doshisha.ac.jp), Yoshiki Nagatani (Kobe City College of Technol., Kobe, Japan), and Mami Matsukawa (Doshisha Univ., Kyotanabe, Japan)

Ultrasound propagation in the heterogeneous long cortical bone was studied. First, ring shaped specimens were obtained from the mid-shaft of a bovine radius. The longitudinal wave velocity distribution of each specimen was measured in the MHz range. Then, using the bilinear interpolation and the piecewise cubic Hermite interpolating method, a 3D axial velocity model was created with the resolution of 40 μm. Assuming the uniaxial anisotropy of the bone,[1,2] distributions of all elastic moduli of the initial 3D heterogeneous model were estimated. At the surface of the model, the elastic moduli were smaller than those of the inside parts. Elastic finite-difference time-domain method was used to simulate axial ultrasonic wave propagation in the model. The initial model and the thinner model, where the inner part of the cortical bone was removed, were compared. The wave front of the first arriving signal (FAS) depended on the heterogeneity in the model. However, the effects of removal on the arrival time of the FAS was not obvious. 1. Y. Yamato, *et al*. (2008). Calcif. Tissue Int. **82**, 162-169. 2. T. Nakatsuji, *et al*. (2011). Jpn. J. Appl. Phys. **50**, 07HF18.

3pBAb5. Modeling wave propagation through the skull for ultrasonic transcranial Doppler. Shreyank Gupta (École de technologie de supérieure, 8626 rue Waverly, Montreal, QC H2P 2R1, Canada, shreyank.gupta@gmail.com), Guillaume Haïat (CNRS, Laboratoire Modélisation et Simulation Multiéchelle UMR, Creteil, France), Catherine Laporte, and Pierre Bélanger (École de technologie de supérieure, Montreal, QC, Canada)

One problem associated with transcranial Doppler ultrasound (TCD) is the relatively low energy penetrating inside the brain through the skull, seriously limiting the image quality. This may be due to the impedance mismatch at the bone interface and to the bone frequency dependent attenuation. The objective of this paper is to model ultrasonic wave propagation through the skull. To do so, an analytical model was developed based on the estimation of the transmission coefficients inside the brain, leading to frequency dependent overall transmission coefficient for a given skin and bone thickness. Moreover, a finite element model was developed taking into account absorption phenomena. Both methods were validated experimentally by comparing the numerical and analytical results with results obtained from a phantom mimicking the skull having an attenuation coefficient equal to 30 dB/cm at 2.25 MHz and a thickness of 4.4 mm. A 2 mm layer of water mimicked the skin. The difference between the maximum amplitude of normalized received US signals obtained analytically and experimentally was 1%. The average relative difference between them was 0.3%. Thus, a working model is designed which can predict the energy inside the brain. Furthermore, the results show that impedance mismatch plays a major role in transmission loss rather than frequency dependent attenuation

3pBAb6. Comparison between an axial transmission device and high resolution peripheral quantitative computed tomography—A clinical evaluation of ultrasound for the assessment of cortical bone quality. Ryoichi Suetoshi, Dorian Cretin (Furuno Electric Co., Ltd., 9-52 Ashihara-cho, Nishinomiya, Hyogo 662-8580, Japan, ryohichi.suetoshi@furuno.co.jp), Ko Chiba (Nagasaki Univ., Nasgasaki, Japan), and Makoto Osaki (Nagasaki Univ., Nagasaki, Japan)

Quantitative ultrasound mainly evaluates cancellous bones, although cortical bone is now thought to be more important to determine bone fragility. Therefore, we developed an axial transmission device (f = 3MHz) to measure the cortical Speed of Sound (cSOS). The purpose of this study is to investigate the material properties in cortical bone that cSOS reflects by using High Resolution-peripheral Quantitative CT (HR-pQCT). In 19 patients and healthy volunteers (aged 33-73), the cSOS at the tibia and radius were measured by our ultrasonic device. Cortical bone parameters were measured by HR-pQCT (Scanco). Areal bone mineral density (aBMD) at the lumbar spine and femur were measured by DXA. cSOS had a strong correlation with the cortical volumetric mineral density (Ct.vBMD) measured by HR-pQCT at both the tibia and the radius ($r = 0.83$, 0.72, $p < 0.001$). A correlation was also found between cSOS and the cortical porosity at the radius ($r = -0.83$, $p < 0.001$). No correlation was found between cSOS and aBMD, which makes cSOS unlikely to depend on the bone quantity. The correlation between cSOS and Ct. vBMD may show that cSOS could reflect the bone quality related to mineralization and porosity. cSOS, not being related to DXA but linked to the bone quality, could be useful for the screening of the mineral properties of cortical bone simply and non-invasively.

3pBAb7. Induced electric potential in cortical bone and cartilage by ultrasound irradiation. Shunki Mori, Sayaka Matsukawa, Mami Kawase (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe, Kyoto 6100321, Japan, dmq1028@mail4.doshisha.ac.jp), Shinji Takayanagi (Nagoya Institute of Technol., Nagoya, Aichi, Japan), and Mami Matsukawa (Doshisha Univ., Kyotanabe, Kyoto, Japan)

LIPUS (low intensity pulse ultrasound) can reduce the time of bone fracture healing. The detailed mechanism of ultrasonic effects on bone, however, has not been clearly understood yet. One possible idea seems to be the piezoelectricity of bone. The purpose of this study is to evaluate the piezoelectricity of cortical bone and cartilage. In order to evaluate piezoelectricity of bone in the MHz range, bone transducers were fabricated. Circular plate cortical bone samples and cartilage samples (diameter; 10.0 mm, thickness; 1.00 mm) were used as piezoelectric materials of the transducer to receive ultrasound. The surface of bone plates was normal to the bone axis or the radial axis and that of cartilage plates was normal to bone axis. By irradiating ultrasound in the MHz range, induced electric potentials were successfully observed in all transducers. The maximum sensitivity of the cartilage transducer was about 93.6 nV/Pa and that of the bone transducer was about 27.5 nV/Pa. Induced electric potentials of cartilage transducers were always larger than those of bone transducers. The main part of the cartilage is type collagen. The results may indicate that the piezoelectricity of the cartilage and bone mainly comes from the collagen.

Session 3pBAc

Biomedical Acoustics and Signal Processing in Acoustics: Bone Sonometry III: Ultrasonic Studies of the Physical Properties of Cortical and Cancellous Bone

Mami Matsukawa, Cochair
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James G. Miller, Cochair
Physics, Washington U Saint Louis, Box 1105, 1 Brookings Drive, Saint Louis, MO 63130

Invited Papers

3:15

3pBAc1. A novel bone sonometer for estimation of calcaneal bone mineral density. Jonathan J. Kaufman and Gangming Luo (CyberLogic, Inc., 611 Broadway, Ste. 707, New York, NY 10012, jjkaufman@cyberlogic.org)

This study evaluated a new clinical ultrasound device, the *QRT 250*, in terms of its ability to estimate BMD at the calcaneus. The *QRT 250* rests on the floor and permits real-time evaluation of calcaneal BMD. The *QRT 250* utilizes a novel heel positioning system that relies on an individual's foot size to locate a specific region-of interest (ROI) in the calcaneus. This feature of the device was designed to account for the relatively high degree of BMD heterogeneity associated with the calcaneus. The device measures in thru-transmission a net time delay (NTD) parameter. NTD has been shown to be highly correlated with bone mass in simulations, *in vitro*, and in other clinical studies. A linear regression of the NTD is used to provide an estimate of calcaneal BMD that would be measured by DXA. A clinical IRB-approved study was carried out in which 145 adults were measured at the calcaneus using ultrasound and DXA (Lunar GE PIXI; WI, USA). In addition, thirty (30) subjects were each ultrasonically measured two (2) times and the short term precision was evaluated. A linear regression showed that $BMD_{US} = 0.12 \cdot NTD—0.11$ and that the linear correlation between BMD_{DXA} and BMD_{US} was 0.89 ($P<0.001$). We found that calcaneal ultrasound measurements yield results that are very closely associated with those from DXA. In the short-term precision study, we found that the mean percent precision was 1.37%. In summary, the *QRT 250* can be used to identify bone loss in osteoporosis and ultimately to reduce the incidence of fragility fractures.

3:35

3pBAc2. *In vivo* radius bone evaluation of women in their late teens by two wave apparatus. Mami Matsukawa, Shoko Nakanishi (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, mmatsuka@mail.doshisha.ac.jp), Isao Mano, Kaoru Horii, Yutaro Yoneda (OYO Electric Co., Ltd., Joyo, Japan), Shiori Umemura, and Etsuko Ozaki (Kyoto Prefectural Univ. of Medicine, Kyoto, Japan)

Making use of the fast and slow wave phenomenon in cancellous bone, an ultrasonic bone measurement system, LD-100 (OYO electric),[1] has been developed and is now commercially available in Japan. From the measurements of fast, slow, and echo waves, the system gives us cancellous bone density (mg/cm^3), cancellous bone elasticity (GPa), and the cortical thickness (mm). The measurement area is the distal 5.5 % site of the radius of the non-dominant hand. In this study, the system was modified for the small radius of teenagers using annular array elements. Under the permission of the ethics committee at Doshisha university, radius bones of 111 high school students were measured (15-17 years old, female). The mean values of cortical thickness of the students were 93.7—97.7% of the young adult mean (YAM). The cancellous bone densities were 82.6—94.0% of YAM. The standard deviations of these values were higher than the deviations of YAM. These data possibly indicate that the growth of radius bone depends on the site and the outer cortical shell grows fast. For further discussions, more data should be necessary. 1. H. Sai, *et al.*, Osteoporos Int. (2010) **21**, 1781-1790

3:55

3pBAc3. Ultrasonic backscatter theory, method, and diagnostic instrument for cancellous bone. Dean Ta and Chengcheng Liu (Dept. of Electron. Eng., Fudan Univ., 220 Handan Rd., Shanghai 200433, China, tda@fudan.edu.cn)

Recently, significant progress has been made in the backscatter measurement of cancellous bone. In this work, we will introduce the basic theory and some key techniques for backscatter cancellous bone evaluation. Backscatter measurements *in vivo* were also carried out with a diagnostic instrument, and the feasibility of using the backscatter for cancellous bone evaluation was investigated. Researchers had reported conflicting associations between backscatter and bone features. Our studies demonstrated that this phenomenon was due to different signal selections. The dense cortical shell obstructs the ultrasonic evaluation of the underlaid cancellous bone *in vivo*. We determined the influence of the overlying cortex on the backscatter signal and proposed a compensation method for the cortical influence. Using an ultrasonic backscatter bone diagnostic instrument, the measurements were performed on 1226 adults and on 467 neonates in the calcaneus. For the adults, the backscatter was significantly correlated with the bone mineral density (BMD) of the spine and hip. The backscatter parameters were moderately correlated with the gestational age and birthweight in the neonates. The *in vivo* findings demonstrate the ultrasonic backscatter as a useful adjunct measurement for the multi-feature assessment of cancellous bone.

4:15

3pBAc4. Detection and evaluation of airborne ultrasound passed through a heel for non-contact bone assessment. Shinnosuke Hirata and Hiroyuki Hachiya (Dept. of Systems and Control Eng., Tokyo Inst. of Technol., 2-12-1 Ookayama, S5-17, Meguro, Tokyo 152-8550, Japan, shin@ctrl.titech.ac.jp)

Ultrasonic bone assessment by quantitative ultrasound (QUS), measurement of ultrasonic propagation characteristics, is one of the diagnosis methods of osteoporosis. In typical QUS devices, opposed ultrasonic transducers are brought into contact with tissue surfaces through an ultrasonic gel to effectively propagate ultrasound. We have proposed non-contact measurement of propagation characteristics using airborne ultrasound passed through a heel. Generally, airborne ultrasound which passed through a tissue is extremely attenuated due to large reflections at boundaries between air and the tissue. In the proposed method, therefore, the signal-to-noise ratio of pass-through ultrasound is greatly improved by pulse compression using a higher-order M-sequence. Furthermore, each side of a heel is inclined to the direction of the ultrasonic beam. Pass-through ultrasound cannot fully be received by opposed transducers, because the ultrasonic beam inflects at each surface. Therefore, inclination of each heel side is measured by the pulse-echo method. Then, each transducer is inclined with considering the surface inflection. The propagation speed of ultrasound in the heel is estimated from the time of flight of pass-through ultrasound, the length of the propagation path, and the propagation speed of ultrasound in air.

4:35

3pBAc5. Ultrasound for bone tissue engineering and regenerative medicine. Frederic Padilla (LabTAU INSERM 1032, French National Inst. of Health and Medical Res. (Inserm) & Université de Lyon, 151 Cours Albert Thomas, Lyon 69390, France, frederic.padilla@inserm.fr), Mario L. Fabiilli (Dept. of Radiology, Univ. of Michigan Med. School, Ann Arbor, MI), Francisco M. Martín-Saavedra (Hospital Universitario La Paz-IdiPAZ, CIBER de Bioingeniería, Biomateriales y Nanomedicina (CIBER-BBN), Madrid, Spain), Kay Raum, Regina Puts (Berlin-Brandenburg School and Ctr. for Regenerative Therapies, Charité - Universitätsmedizin Berlin, Berlin, Germany), Laurence Vico (Inserm U1059 Lab Biologie intégrée du Tissu Osseux, French National Inst. of Health and Medical Res. (Inserm) & Université de Lyon, St Etienne, France), and Christopher G. Wilson (Boston University Business Innovation Ctr. · dBMP Program, Bioventus LLC, Boston, MA)

We will review the role of ultrasound in bone tissue engineering and regenerative medicine, for enhancement of fracture healing and reactivation of a failed healing process. These approaches rely typically either on the direct stimulation of the healing process or in the delivery at the injury site of scaffold-embedded biologics such as cells, signaling molecules, and genetic material. Different options are available for ultrasound enhancement of fracture healing such as low intensity pulsed ultrasound (LIPUS) and shock waves, for the delivery of regenerative factors and encoding genes, and for the assembly and patterning of scaffolds. A main emphasis will be put on LIPUS-related bioeffects, the most widespread and studied technique. Pronounced bioeffects on tissue regeneration have been reported employing intensities within a diagnostic range. We will describe the current knowledge of the biological response to LIPUS, involving numerous cell types and molecular pathways, hypothesis about LIPUS biophysics, and our recent findings on mechanosensitive transcriptional response of cells after LIPUS. We will also present the use of focused ultrasound to spatiotemporally control the delivery of growth factors, either by induction of transgene expression controlled by a heat-activated and ligand-dependent gene-switch, or by triggered release using acoustic droplets vaporization within biopolymer scaffolds.

4:55

3pBAc6. Acoustical estimation of endosseous implant stability: Finite element modeling and experimental validation. Vu-Hieu Nguyen, Romain Vayron, and Guillaume Haiat (Multiscale Modeling and Simulation Lab., CNRS, Laboratoire MSME UMR 8208, fac des Sci., UPEC, 61 av du gal de Gaulle, Creteil, France, vu-hieu.nguyen@u-pec.fr)

Various endosseous implants are widely used in orthopedic and dental surgeries. However, failures are observed, which may have dramatic consequences and are often due to a loss of implant stability. The aim of this presentation is to describe two acoustical methods developed to assess the stability of different implants. First, a quantitative ultrasound method was used to determine the stability of dental implants. A monoelement 10 MHz transducer is screwed into the implant and the signal is processed to obtain an indicator. The method was first validated in implants embedded in biomaterials used as bone substitutes, then *in vitro* and eventually *in vivo* by comparison with histology. A 3D FEM was developed to optimize the device conception and carry out sensitivity studies. Second, an impact hammer was used in order to investigate whether the AC implant primary stability can be evaluated using such approach. The method was first validated *in vitro* and in cadaveric experiments. The results showed that an indicator based on the impact momentum may be correlated to the implant pull-out force. A 3D FEM was also developed to analyze and better understand experimental results. This study shows the potential of acoustical methods to assess non-invasively implant stability.

Contributed Paper

5:15

3pBAc7. Acceleration of critical bone defect healing by guided low-intensity ultrasound in a rat tibial model. Yi-Xian Qin, Jingbo Liu, Dongye Zhang, and Xiaofei Li (Biomedical Eng., Stony Brook Univ., BioEng. Bldg., Rm. 215, Stony Brook, NY 11794, yi-xian.qin@stonybrook.edu)

Bone defects, caused by trauma, inflammation, and tumors, may lead to loss of limb functions and substantially affect the quality of patients' life. Low-intensity ultrasound (LIUS) can transmit through and into living tissues as acoustic pressure waves, and generate dynamic acoustic radiation force in the local region to induce stem cell recruitment, and mineralization. The aim of this study was to evaluate the effect of LIUS induced healing in a rat tibial critical defect. A total of 40 skeletally mature female Lewis rats were used in this study with a surgical bone defects of 2 mm in-diameter at proximal tibia. The defects of the left tibia (LIUS group) received daily treatment of ultrasound, and the defects in the right tibia (Control) received sham stimulation. A modified repetitive frequency at 100 Hz, characteristic frequency of 1 MHz at an intensity of 30 mW/cm^2 were used for the stimulation for 20-min/day. After 2 weeks, the μCT images showed that trabecular bone formed in the defect area of both LIUS and control groups. LIUS treated tibia showed more increased in BV/TV, Cont.D, Tb.N, and BMD, and significant decrease in Tb.Sp compared to control. Four-point bending testing showed LIUS treatment significantly increased the bone stiffness (kN/mm) relative to control group. The histological analysis and μCT data revealed that LIUS therapy can significantly increase new trabecular bone volume, so as the stiffness and ultimate stress of LIUS treated group with two-week as optimized treatment period for healing.

Session 3pEA

Engineering Acoustics and Physical Acoustics: Outdoor Sound Propagation and Outdoor Public Address

JohnPaul R. Abbott, Cochair
Department of Physics and Astronomy, National Center for Physical Acoustics, University of Mississippi, 1 Coliseum Dr., Room 1044, Oxford, MS 38677

Yoshifumi Chisaki, Cochair
Chiba Institute of Technology, 2-17-1, Tsudanuma, Narashino 2750026, Japan

Chair's Introduction—1:00

Invited Papers

1:05

3pEA1. Irregular qualities of sound sources appeared at the input of the outdoor emergency mass notification sound system. Kiyohiro Kurisu, Yasushi Matsumoto, and Ryota Matsuishi (TOA Corp., 2-1 Takamatsu-cho, Takarazuka, Hyogo 665-0043, Japan, kurisu_kiyohiro@toa.co.jp)

In order to obtain an intelligible speech at a listening point addressed by the mass notification sound system, not only high transmission quality of the system, but also a sufficiently clear speech sound source is needed. However, the qualities of speech sources may not be constant because they are supplied by different types of production and transmission methods, e.g., the variety of speech source production methods causes various specifications and irregular qualities of sources, and some types of transmission methods between the parent station (speech production system) and children stations (acoustic amplification system) cannot convey the source signal with sufficient quality. In this presentation, the authors showed examples of irregular source quality appeared at the input of the acoustic amplification system and discussed the necessity of verifying the signals at several points in the total system (i.e., the system between the speech source and the listening point) to regulate their qualities. Then, the guidelines for the performance verification of the outdoor emergency mass notification sound systems which was published by the Acoustical Society of Japan was introduced.

1:25

3pEA2. Improvement of signal to noise ratio at outdoor listing position by sound emission timing control over the Internet for mass notification system. Yoshifumi Chisaki (Chiba Inst. of Technol., 2-17-1, Tsudanuma, Narashino, Chiba 2750026, Japan, yoshifumi.chisaki@p.chibakoudai.jp) and Taira Onoguchi (Kumamoto Univ., Kumamoto, Japan)

Mass notification system has multiple nodes, where a node consists of some loudspeakers, in order to cover wide service area. Sound signal is emitted from grouped nodes simultaneously in general. Thus, the simultaneous emission from grouped nodes sometimes makes overlapping of sound signals at a listening point in the service area. Since it affects speech intelligibility, reduction of overlapping is a quite important issue not only for emergency but also reduction of wasteful signal power on mass notification system. Recently,global positioning device and network devices, such as 3G/LTE, become reasonable to be equipped to a node of mass notification system. Therefore, it makes easy to control emission timing at each node intelligently. Development of an intelligent algorithm for mass notification is introduced and a part of the algorithm is implemented on a single board computer, Raspberry PI. Performance on signal to noise ration at all the listing points in service area is presented and discuss it with other evaluation methods.

1:45

3pEA3. Speech intelligibility prediction method using machine learning for outdoor public address systems. Yosuke Kobayashi (Graduate School of Eng., Muroran Inst. of Technol., 27-1 Mizumoto-cho, Muroran 050-8585, Japan, ykobayashi@csse.muroran-it.ac.jp), Kengo Ohta (Anan College, National Inst. of Technol., Anan, Japan), Kazuhiro Kondo (Graduate School of Sci. and Eng., Yamagata Univ., Yonezawa, Japan), and Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan)

Subjective speech intelligibility assessment is important for the development of outdoor public address system. However, as this assessment is difficult in many cases, we propose an objective speech intelligibility evaluation system that includes a machine learning technique. In this talk, we have proved a subjective evaluation and objective prediction of speech intelligibility using the outdoor public address systems at 10 locations in Sendai City, where impulse responses were recorded after the Great East Japan Earthquake. First, the results of the subjective intelligibility evaluation by different test word lists with the same sound field conditions showed that the root mean squared error (RMSE) was very small, not exceeding 7.0%. Next, we generated the intelligibility prediction model trained with true/false results of 22 subjects using the support vector machine (SVM). This prediction model extracted the feature vector, using the ITU-T P.563 speech quality feature set of the test speech signal. We evaluated the predictive performance of the prediction model using data that was not used in training, and the RMSE obtained was 4.0%. This result was shown to be highly accurate with a value even less than the subject experiment result.

2:05

3pEA4. Optimization of speech presentation in mass-notification sound systems based on word intelligibility. Shuichi Sakamoto, Zhenglie Cui (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Masayuki Morimoto (Graduate School of Eng., Kobe Univ., Kobe, Japan), and Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan)

Outdoor mass-notification sound systems can effectively convey emergency announcements over a wide area without requiring the audience to use any special devices. However, the intelligibility of speech presented via these systems is often severely degraded by long-path echoes. Under such conditions, it is important to consider robust ways of presenting speech information. Various studies have highlighted that there are many factors which affect the speech intelligibility with long-path echoes. We have focused on the effect of word familiarity and inserting pauses between the words in a speech signal. High-familiarity words are easy to understand even when presented under challenging listening conditions. By inserting pauses, speech intelligibility could be improved because more of the speech signal can be received without significant overlapping long-path echoes. In this study, speech materials were modeled based on actual sentences used in emergency notifications given via the Japanese mass-notification system, J-Alert. By using this model, the effects of word familiarity and inserting pauses on word indelibility under long-path echo were examined. The results of the experiments suggest that speech signals can be robustly understood despite long-path echoes by using high-familiarity words with adequate inter-word pauses.

2:25

3pEA5. Relationship between speech intelligibility and objective measures in sound fields with a discrete long-path echo. Hayato Sato, Masayuki Morimoto, Yusuke Miyagawa (Environ. Acoust. Lab., Graduate School of Eng., Kobe Univ., 1-1 Rokkodai, Nada, Kobe 6578501, Japan, hayato@kobe-u.ac.jp), and Yôiti Suzuki (Human Information Systems Div., Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan)

The outdoor public address system is one of the important methods to notify evacuation information in disaster situations. However, when speech announcements are simultaneously radiated from a number of dispersedly located loudspeakers, speech intelligibility decreases because of temporal or spatial segregation of speech streams, just like a long-path echo in large closed spaces. The present study is focusing on relationship between speech intelligibility and spatial characteristics of long-path echo. Useful-detrimental ratio (U_{50}) was tested to predict speech intelligibility scores in sound fields with a discrete long-path echo. A noteworthy feature of U_{50} is that all energy of long-path echo is explicitly counted as detrimental energy. The results of intelligibility tests demonstrated that the feature led to lower intelligibility predictions, that is, predictions on the safe side. Meanwhile, the prediction errors increased when the difference between interaural differences for the direct sound and the long-path echo. The extended U_{50} to consider binaural effects referring to the concept of binaural speech transmission index [van Wijingaarden and Drullman, J. Acoust. Soc. Am. **123**, 4514-4523 (2008)] showed a higher correlation with the intelligibility scores relative to the original U_{50}.

2:45

3pEA6. Subwoofer array modeling and optimization. Wolfgang Ahnert and Stefan Feistel (Ahnert Feistel Media Group, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu)

In the last few years, configuring subwoofer arrays in order to achieve a desired sound radiation pattern has become one of the most discussed topics in the sound reinforcement industry. The directivity of such arrays of low-frequency loudspeakers can typically be controlled by means of delay, gain, and polarity settings applied to the individual elements. A new software solution is presented that allows modeling and optimizing such setups, including various cardioid configurations as well as end-fire and steered setups. Frequencies down to 20 Hz can be investigated. It is also explained how coherent sound superposition at low frequencies and incoherent interaction of sound waves at high frequencies can be simulated consistently by means of a cross-over frequency band. Finally, an optimization method is presented that allows computing elemental delays automatically based on a user-defined coverage angle for the array.

3:05–3:20 Break

3:20

3pEA7. A framework for providing real-time feedback of environmental noise levels over large areas. Edward T. Nykaza, Michael J. White, Jesse M. Barr, Matthew G. Blevins (ERDC, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@erdc.dren.mil), Steven L. Bunkley (ERDC, Vicksburg, MS), Nicole M. Wayant (ERDC, Alexandria, VA), and D. Keith Wilson (ERDC, Hanover, NH)

Environmental noise can cause sleep disturbance, annoyance, complaints, and quite possibly adverse health effects. This is true for continuous noise sources such as steady road traffic noise, impulsive noise sources such as blasts or sonic booms, or sources that fall in-between such as intermittent train and aircraft noise. One way to manage environmental noise is to use noise-monitoring technology to provide both the noise-producers and noise-experiencers feedback on the actual noise environment. Traditional noise-monitoring systems, however, only provide this information at a few locations resulting in an incomplete picture of the noise environment over the entire regions of interest. In this paper, we discuss a framework for providing real-time feedback of the noise environment over a large area (e.g., 100 km^2). We show all the steps that are needed to convert the raw noise-monitor data into noise maps and noise impact maps to help manage environmental noise. We discuss the complexity of the problem and present several different ways to visualize the data.

3:40

3pEA8. Formulation of sound propagation and scattering in forests as a radiative transfer theory. Vladimir E. Ostashev and David K. Wilson (U.S. Army Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@noaa.gov)

Several approaches have been used in the literature to characterize sound scattering in forests. In this presentation, an approach based on the radiative transfer equation (RTE) is considered. Although the RTE has been used for other types of scattering problems, it is new to forest acoustics. The RTE is an integro-differential equation for the specific intensity, which is the average power flux density within a unit solid angle in a certain direction of propagation. It describes the processes of sound scattering and absorption by discrete scatterers such as tree trunks, branches, and leaves. RTE is applied to sound propagation in a four-layer forest model (ground, trunks, canopy, and open air). The trunk layer is modelled with vertical cylinders, while the canopy layer is modeled by spheres. The scattering amplitudes and total scattering cross sections of these scatterers are identified. At the interface of the trunk layer and ground, the specific intensity is specularly reflected. Although the RTE is based on a phenomenological theory for propagation of intensities, the mutual coherence function of the sound field (which can be obtained from a Helmholtz equation and the multiple scattering theory) may be expressed in terms of the specific intensity.

4:00

3pEA9. Ubiquitous monitoring of infrasound for early detection of events relevant to disaster. Ryouichi Nishimura, Yasunori Owada, and Shinichi Taira (National Inst. of Information and Communications Technol., 2-1-3 Katahira, Aoba-ku, Sendai, Miyagi 980-0812, Japan, ryou@nict.go.jp)

A terrestrial event resulting in natural disaster often radiates infrasound when it occurs. Similar phenomenon can be observed in extreme weathers as well. Infrasound travels through the air approximately at the speed of sound and it is faster than traveling of physical kinetic energy which may sometimes bring about a disastrous impact on human lives. Therefore, early detection and source localization of such events are promising for hazard prevention and damage mitigation. To achieve this purpose, coverage of the land is important in terms of celerity and spatial precision, but facilities suitable for monitoring infrasound are not common. In Japan, one monitoring station, IS30, is in operation under Comprehensive Nuclear-Test-Ban Treaty (CTBT) and a few microbalrometer arrays are experimentally set up by Japan Weather Association (JWA) on some parts of Japanese coast. A group of University of Hawaii developed an app for capturing infrasound on iOS devices, and the captured data can be utilized for further analysis. We developed a similar app for Android and deployed smartphones running it on a wireless network to collect information for 24 hours a day stably. Combining the information gathered, we tried to estimate the source signal and its location.

4:20

3pEA10. The effects of wind noise on outdoor sound propagation and detection. JohnPaul R. Abbott (National Ctr. for Physical Acoust., Univ. of MS, 100 Bureau Dr. Stop 8361, Gaithersburg, MD 20899, johnpaul.abbott@gmail.com)

Wind noise is a form of pseudo-noise. It is generated and propagated by physical mechanisms that are different from acoustic noise; however, its effect on microphones and similar pressure sensitive devices is indistinguishable from acoustic noise. Wind noise is defined as the pressure fluctuations that are generated and propagated by the turbulent flow of an incompressible fluid. Acoustic noise is generated and propagated by the adiabatic compression and expansion of air. Since the effect of wind noise is the same as acoustic noise, it can have significant effect on the propagation and detection of desired acoustic signals at sonic and infrasonic frequencies. This can lead to adverse impacts on mass and early warning systems. This talk will present an overview of the effects of wind noise on outdoor sound propagation, detection, and efforts to mitigate and account for its influence on sonic and infrasonic acoustic signals. It will also discuss the potential impacts of sonic and infrasonic wind noise on mass and early warning systems.

4:40

3pEA11. Effects of local and global meteorological data on predicting blast noise levels. Matthew G. Blevins (U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, matthew.g.blevins@usace.army.mil), Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Hanover, NH), Steven L. Bunkley (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS), and Edward T. Nykaza (U.S. Army Engineer Res. and Development Ctr., Champaign, IL)

Predicting noise levels due to military blasts is important near training installations, since surrounding communities may be adversely affected. Prediction models, however, rely on atmospheric or weather data that is not readily available or easily attainable. In this study, blast noise level predictions 1-8 km from the source are made using two sources of meteorological data: publicly available data derived from weather stations that are tens of kilometers from a detonation site, and near-ground weather stations that are 2-6 km from the source. To predict the blast noise levels, statistical learning regression models are trained on each set of meteorological data and over 1000 measured blast events. Predictions from both meteorological data sources are compared to actual blast noise levels recorded during a field experiment by Valente *et al.* [2012], which were captured under a wide variety of atmospheric conditions. The root-mean-square errors for each model are compared, and the most important input parameters are identified.

Session 3pEDa

Education in Acoustics: Demonstrations and Tools in Acoustics Education

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

Akira Nishimura, Cochair
Media and Cultural Studies, Tokyo University of Information Sciences, 4-1, Onaridai, Wakaba-ku, Chiba 2658501, Japan

This walk-in demonstration session is open to all meeting attendees. Presenters will all be available during the entire session to show and explain their demonstrations, similar to a poster session. In addition to the listed abstracts, additional presenters, who have papers in other Education in Acoustics sessions, are participating. A list of their names with reference to their abstracts is given below.

Contributed Papers

3pEDa1. Applying the cellular automaton model in speech production process simulation. Koji Taka (Math. and Sci. Education Res. Ctr., Kanazawa Inst. of Technol., 7-1 Ohgigaoka, Nonoichi, Kanazawa, Ishikawa 921-8501, Japan, taka@neptune.kanazawa-it.ac.jp)

This study investigates the dynamic simulation of the speech production process. When an utterance occurs, the sound pressure and the particle velocity distribution of the vocal tract space changes from word to word. The author examined and visualized this change to illustrate how it happens. The author used the cellular automata as a model to calculate the acoustic characteristics of the vocal tract in each region. This study takes a different approach from the traditional z-transform to the cellular automata to calculate the dynamic sound pressure at each site of the vocal tract space. To simplify the calculation, an acoustic cylindrical tube model was used. Dynamic calculations were made even more manageable by changing the physical model from a three-dimensional to a one-dimensional cellular automaton model. Consequently, this study demonstrates that the relationship between the words and the sound pressure distribution can be examined dynamically using the cellular automata model.

3pEDa2. "A record without (or with) prior acoustic information" and "Given: 1.Manet, 2.Coil—Oscillation from a minimum unit of speaker". Kazuhiro Jo (Faculty of Design, Kyushu Univ. / YCAM, 4-9-1, Shiobaru, Fukuoka, Fukuoka 815-8540, Japan, jo@jp.org) and John Smith (none, Gunma, Japan)

In this demonstration, we'd like to introduce two of our work, "A record without (or with) prior acoustic information" and "Given: 1.Manet, 2.Coil—Oscillation from a minimum unit of speaker." The first one is a work in which we computationally draw a groove of a record as a vector graphic (with Adobe Illustrator or other tools) by calculating frequencies with a number of zigzags and engrave the groove on diverse materials on diverse material including paper, wood, or acrylic with laser cutter or cutting plotter to produce sounds. Through the demonstration, we show a process of making the record with a cutting plotter on the spot. The result and other examples would be played on a standard analog record player. The second one is a work in which we split a unit of a speaker into a magnet and a coil. We demonstrate an example of the work by Johnsmith as a cartilage conduction hearing with a set of neodymium magnets and a pair of coils with stereo mini jack. Through the demonstration, people could listen the sound from her/his portable audio player by directly vibrating her/his cartilages. Both of the work would present alternate embodiments of matured audio technologies (i.e., vinyl record and headphone) with a help of later technological developments. The demonstration shows basic principles of how audio technologies work. Through our practices at the intersection of media archeology and personal fabrication, we'd like to reconsider the ordinariness of acoustic media technologies.

3pEDa3. Realtime and interactive tools for speech and hearing science education. Hideki Kawahara (Wakayama Univ., 930 Sakaedani, 930, Wakayama, Wakayama 640-8510, Japan, kawahara@sys.wakayama-u.ac.jp)

Recent advances of computational power and software foundations make it possible to introduce interactive and realtime tools in speech and hearing science education with relatively low cost. The tool, SparkNG (Speech Production and Auditory perception Research Kernel, Next Generation) consists of four applications. They are (a) real-time FFT analyzer with interactive spectrogram, (b) real-time display of FFT spectrogram (narrow-band and wide-band) and auditory (ERB, Bark and 1/3 octave) spectrogram, (c) real-time vocal tract shape visualizer, and (d) interactive speech production simulator enabling vocal tract shape and transfer function manipulation with aliasing-free L-F model (Kawahara et al., 2015 APSIPA) manipulation. The tool is useful for teachers to demonstrate underlying concepts in an intuitive and attractive manner. It also is useful for researchers to check their ideas quickly. MATLAB source codes of the constituent functions are open to everyone. A compiled MATLAB license-free version is also available. [This work was supported by JSPS KAKENHI Grant Number JP16K12464.]

3pEDa4. Three-dimensional sound-field visualization system using head mounted display and stereo camera. Atsuto Inoue, Yusuke Ikeda, Kohei Yatabe, and Yasuhiro Oikawa (Intermedia Art and Sci., Waseda Univ., 59-407-2, 3-4-1, Okubo, Shinjyuku, Tokyo 169-8555, Japan, han-overeem@toki.waseda.jp)

Visualization of a sound field helps us to intuitively understand various acoustic phenomena in sound design and education. The most straightforward method is to overlap the measured data onto a photographic image. However, in order to understand an entire three-dimensional (3D) sound field by using a conventional two-dimensional screen, it is necessary to move a camera and measure repeatedly. On the other hand, the augmented reality (AR) techniques such as an optical head mounted display (OHMD) have been rapidly developed. In this study, we propose a sound field visualization system using an OHMD and a handheld four-point microphone. This system calculates sound intensity from the four sound signals in real time. Then, the sound intensity distribution is depicted as arrows in the 3D display. The position and angle of the microphones and user's head are acquired via AR markers and a gyro sensor of the OHMD. The system realizes simple and effective visualization of 3D sound field information from the various directions and positions of view. For the experiments, the sound field generated by a two-way loudspeaker was visualized. The results

suggested that the proposed system can present information of the field in easily recognizable manner.

3pEDa5. Making headphones using plastic bottles and a milk carton. Shinjiro Seguchi (D&I Creation Dept., Sony/Taiyo Corp., 1402-14, kansui,ohga, Hiji-machi,Hayami-gun, Oita-ken 879-1504, Japan, Shinjiro.Seguchi@jp.sony.com)

The SONY Science Program provides tomorrow's principles and the opportunity for hands-on experience for children. It will hopefully motivate them to acquire skills needed to make a better society by applying the power of science. An inclusion workshop is one of the Science Programs, where children can learn about scientific principles and technology through demonstrations. Making headphones from PET bottles and a milk cartons, organized by technicians from Sony/Taiyo Corporation, is one of the topics of the inclusion workshop. Children learn how the pieces they constructed vibrate and generate sounds. The program is also designed to make the participants aware of the varied individuality of each and every one of them, and to deepen mutual understanding through this experience of diversity and inclusion.

3pEDa6. A modular guitar for teaching musical acoustics. Jeremy Marozeau (Hearing Systems Group, Tech. Univ. of Denmark, HEA Bldg. 352, DTU-ELEKTRO, Lyngby 2800, Denmark, jemaroz@elektro.dtu.dk)

In order to keep students activated in a course on musical acoustics, they were asked to build a modular guitar, designed to be updated throughout the course. In the first stage, dedicated to the physics of strings, a guitar was made out of three strings attached to a long piece of wood. The students measured the effect of the place of plucking on the mode of the vibrations of the strings. The second stage was dedicated to the acoustic resonances. Using a laser cutter, the students built a wooden box that was coupled to their guitar using straps. New acoustical measurements were made to study the effect of the shape of the resonator on the spectrum of the sound. In the third stage, as the different tuning systems were learned, the students built a fingerboard with the appropriated positions of the frets. In the last stage, the students have implemented some digital effects and tested them on their guitar using a piezo-electrical pickup. As nothing was glued, the students were able to easily change each part of the guitar (resonator, sound hole, fret positions, microphone, …) in order to experience their direct effect and their interactions.

1aED3. Tools for education in acoustics and phonetics — Takayuki Arai

1pED1. Advanced technical listening training program at Kyushu University — Kazuhiko Kawahara

1pED2. The use of all-in-one sensor devices in the general education acoustics laboratory — Andrew Morrison

1pED4. Visualizing sound directivity via smartphone sensors — Scott Hawley

1pED5. Efficacy of a new spatial ear training program for "Ensemble width" and "Individual source width" — Hidetaka Imamura

4pEDa7. Introducing programming through acoustics and audio at Belmont University — Eric Tarr

WEDNESDAY AFTERNOON, 30 NOVEMBER 2016 KAHILI, 4:00 P.M. TO 5:20 P.M.

Session 3pEDb

Education in Acoustics: Education in Acoustics for Kids (K-12 students)

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

Fumiaki Satoh, Cochair
Chiba Institute of Technology, Tsudanuma 2-17-1, Narashino 275-0016, Japan

Invited Papers

4:00

3pEDb1. Activities of the research committee on education in acoustics in ASJ for children. Fumiaki Satoh (Architecture, Chiba Inst. of Technol., 2-17-1 Tsudanuma, Narashino-shi, Chiba 275-8588, Japan, fumiaki.satoh@it-chiba.ac.jp), Takayuki Arai (Sophia Univ., Chiyoda-ku, Tokyo, Japan), and Akira Nishimura (Tokyo Univ. of Information Sci., Chiba-shi, Chiba, Japan)

In the Acoustical Society of Japan (ASJ), the Research Committee on Education in Acoustics was established in 1997. Although its history is relatively short in comparison with the Administrative Committee on Education in Acoustics in the Acoustical Society of America (ASA), many activities have been steadily carried out. For example, surveys of syllabi (educational plans of subjects in universities) were made. Demonstration tools used in university classes were also surveyed. The extent of our research was not limited to only universities. Textbooks and teaching materials used in elementary schools, junior high schools, and senior high schools were also researched. Furthermore, activities in cooperation with the National Museum of Nature and Science in Japan have been taking place, including an exhibition and a class with acoustic demonstrations and handicrafts for children. We believe that such activities are very important as social contributions of an academic society. In this presentation, some of our activities focusing on the education for children will be introduced.

4:20

3pEDb2. Assessment of the Acoustical Society of America educational activity kit: Survey results from K-12 teachers. Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, vigeant@engr.psu.edu) and Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, Joliet, IL)

The Committee on Education in Acoustics within the Acoustical Society of America developed an activity kit for K-12 teachers to use in the instruction of the fundamentals of sound in 2011. The kit includes activities that cover a range of topics, including musical instruments, animal bioacoustics, and hearing loss. Supplies for many of the activities are provided, including tuning forks, a sound level meter, and materials to make instruments. Many electronic resources are also available online, including lesson plans. Since 2011, over 475 kits have been distributed to K-12 teachers across the United States. To evaluate the effectiveness of the kits, a survey was sent out to all kit recipients. The response rate was 23% (n = 102), with 58% of the respondents being high school teachers. Out of the eight primary activities, the two activities with the highest ratings were (1) striking and listening to the tuning forks, which demonstrates the concepts of frequency and wavelength, and (2) touching a ping pong ball suspended on a string, to a tuning fork, which demonstrates the concepts that sound is a form of vibrational energy and amplitude. The presentation will highlight the key results from the survey and include a sample of the comments.

4:40

3pEDb3. Descriptions of sound in the textbooks of living environment studies in elementary schools and the development of a life and sound activity programs. Miki Toyomasu (Oita Univ., 5-31-10-403 Chihaya Higashi-ward, Fukuoka, Fukuoka 813-0044, Japan, toyomasu@zd5.so-net.ne.jp), Sayo Suzuki, and Yoshiko Akitake (Univ. of Teacher Education Fukuoka, Munakata, Japan)

In Japan, children learn about the subject Living Environment Studies in the first and second grades at elementary school. Eight companies published the 2015-approved elementary school Living Environment Studies textbooks. In this study, we investigated and analyzed the references to sound in all the Living Environment Studies textbooks and the development of life and sound activity programs. Textbooks were found to include contents relating to nature sounds (e.g., sounds made by insects and the sound of rain); musical sounds; sounds of life (e.g., sound of toys); and sound of voices (e.g., loud screams or shrieks in response to danger seeking help), etc. We developed two activity programs, a quiz on sounds and a competition for loud voices. The purpose of these programs was to learn about enjoying "sound" and understand its role in life. These activity programs conducted in Clubs for After School Activities for Children at F city. A questionnaire survey was conducted after two activity programs. Eight children answered that the quiz on sounds was fun and nine children answered that the loud voice competition was fun ($N = 14$). Further investigation and improvements are required.

5:00

3pEDb4. Factors influencing students' choice of musical instruments: A case study of Tertiary Institutions in South-Eastern Nigeria, a survey of University of Nigeria, Nsukka, College of Education Eha-Amufu Nsukka and Nnamdi Azikiwe University, Awka. Stephen G. Onwubiko (Music, Univ. of Nigeria, Nsukka Enugu State, Nsukka-Enugu, Nsukka-Enugu 234042, Nigeria, stephen.onwubiko@gmail.com)

The choice of musical instruments is among the most important factors in determining the course of students' music education. Instruments and its/their selection can be a challenging process with attendant factors. Musical acoustics which falls into intrinsic-extrinsic motivating factor and gender stereotyping of instruments are among those factors. The association of musical acoustics with particular instrument can significantly influence students' choice of musical instrument, thereby resulting in numerous negative-hatred consequences, including fewer instrument choices, limited ensemble participation, and peer disapproval. The findings show that musical acoustics reliably predicted the choice of musical instrument. In order for students to make the right instrument choice, musical acoustics should be encouraged, not the family (parents) forcing their children into musical instrument choice careers. Teachers, like parents, work very closely with learners in schools and they know the abilities of their students in various instruments. Therefore, teachers should guide their students in their musical instrument acoustics before their choice and aspirations in line with their abilities in various instruments. Musical acoustic guidance and counselling at tertiary entry level will enable the students learn about and explore careers that ultimately lead to career choice of the musical instrument; questionnaires as a means of data collection.

Session 3pNSa

Noise, Psychological and Physiological Acoustics and ASA Committee on Standards: Current Issues in Hearing Protection and Hearing Conservation II

Elliott H. Berger, Cochair
Personal Safety Division, 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650

Jinro Inoue, Cochair
Department of Health Policy and Management, University of Occupational and Environmental Health, Japan, 1-1 Iseigaoka, Yahatanishi-ku, Kitakyushu 807-8555, Japan

Chair's Introduction—1:00

Invited Papers

1:05

3pNSa1. Effectiveness of personal noise exposure measurement in the noisy workplace. Koko Takezawa (East Japan Railway Co., Sendai 980-8508, Japan, kokotakahashi@med.uoeh-u.ac.jp), Jinro Inoue (Dept. of Health Policy and Management, Inst. of Industrial Ecological Sci., Univ. of Occupational and Environ. Health, Kitakyushu, Japan), Koki Mori (East Japan Railway Co., Sendai, Japan), Seichi Horie, and Syoko Kawanami (Dept. of Health Policy and Management, Inst. of Industrial Ecological Sci., Univ. of Occupational and Environ. Health, Kitakyushu, Japan)

In Japan, the Working Environment Measurement Standards require employers to measure noise of working environment by two kinds of methods using a sound level meter (SLM) once every 6 months. A-Sampling method is to be measured equivalent continuous sound level for at least 10 min. B-Sampling method is to be performed close to workers at times when the sound level appeared highest. We perform the risk assessment based on the values of the field measurement. In the East Japan Railway Company, workers move around several workshops and use hand tools. Therefore, it is difficult to estimate the personal noise exposure using only an SLM. To compare dose measurement with the Japanese field measurement, we carried out dose measurements for inspection or repair workers using a Personal Noise Dose Meter (Type 4448; Brüel & Kjær). The results revealed that all sound levels using a dosimeter were less than 85 dBA, while the sound levels using an SLM were more than 90 dBA. In this case, the risk can be overestimated. Thus, we conclude that the Working Environment Measurement Standards should adopt the dose measurement.

1:25

3pNSa2. Bacterial attachment and insertion loss of earplugs used for fixed periods in noisy workplaces. Jinro Inoue, Yumi Nakagawa, Aya Nakamura, and Shoko Kawanami (Dept. of Health Policy and Management, Univ. of Occupational and Environ. Health, Japan, 1-1 Iseigaoka, Yahatanishi-ku, Kitakyushu, Fukuoka 807-8555, Japan, j-inoue@med.uoeh-u.ac.jp)

In noisy workplaces, employees often tend to use earplugs for a long time in Japan. We have previously reported conditions related to bacterial attachment and the insertion loss of earplugs using data collected from six companies. In the present study, we distributed different kinds of foam earplugs at five companies and collected them after 1-, 2-, 4-, and 8-week usage ($n = 123$). We examined the total viable counts and presence of *Staphylococcus aureus* using 3M Petrifilm. We evaluated the insertion losses by means of the GRAS 45CB Acoustic Test Fixture. We detected a large number of viable counts in 5% of the earplugs. We found *Staphylococcus aureus* in 7% of the earplugs. We observed a small deterioration in insertion loss after 8-week usage. Neither the condition of bacterial attachment nor the insertion loss correlated with duration of use. We discuss the proper usage of earplugs in terms of industrial health.

1:45

3pNSa3. Acoustic characteristics of respirators. Takao Sakuma (JFE Steel Corp., 1 Kawasakicho, Chiba, Chiba 260-0835, Japan, hyakuyen56@gmail.com), Jinro Inoue, Toshitaka Yokoya, and Yukimi Endo (Dept. of Health Policy and Management, Univ. of Occupational and Environ. Health, Japan, Kitakyushu, Fukuoka, Japan)

In some workplaces, employees are obliged to wear a respirator to protect them from harmful substances. However, an acoustical problem with respirator design is that respirators diminish the voice. Frequency analysis of sound through respirators could help tackle the problems associated with this design. We used a mannequin with a loudspeaker attached to its mouth and emitting pink noise. We measured the noise with 59 combinations of respirators and filters. For half-face masks with one or two filters, attenuation of the noise under 1 kHz was relatively small. The attenuations over 1 kHz varied widely among the tested respirators. With full-face masks, the attenuated frequency was lower than with the half-face masks. Respirators with a speaking diaphragm conducted louder sound over 3 kHz than did respirators without a speaking diaphragm. Among disposable masks, the attenuations were smaller than with half-face masks. We discuss the requirements for smooth communication among workers using respirators in light of the attenuation values of respirators at different frequencies.

Session 3pNSb

Noise: Ambient Listening, Communication and Noise Reduction of Special Hearing Protectors

Hilary Gallagher, Cochair
Battlespace Acoustics Branch, Air Force Research Laboratory, 2610 Seventh Street, Bldg. 441, Wright-Patterson AFB, OH 45433

Richard L. McKinley, Cochair
Battlespace Acoustics, Air Force Research Laboratory, 2610 Seventh Street, AFRL/711HPW/RHCB, Wright-Patterson AFB, OH 45433-7901

Invited Papers

2:15

3pNSb1. Communication and situation awareness needs of hearing protection systems in high noise environments. Kurt Yankaskas (Office of Naval Res., 875 N. Randolph St., Arlington, VA 22203-1995, kurt.d.yankaskas@navy.mil)

The primary mission of aircraft carriers is to launch and recover military aircraft. Critical to the highly choreographed operations on the flight deck of the aircraft carrier is communications between aircraft directors, plane crews, and pilots as well as maintaining situation awareness in a highly dynamic environment. This must occur in an acoustic environment of 110 to 150+ dBA. These noise levels readily exceed the requirements for double hearing protection. A dilemma is generated in trying to provide adequate hearing protection while maintaining critical communications and situation awareness during flight operations at these noise levels. This paper will review previous studies of hearing protection effectiveness, compliance, factors affecting compliance, and impact on job performance in environments that readily exceed legacy hearing protection. To address the situation awareness shortfalls, a tri-service group (US Army, Air Force and Navy) is developing a standard for detection and localization criteria (commonly known as situation awareness) to assess operational suitability. The standard is intended to drive manufacturers to measure beyond the attenuation a hearing protector provides to address operational needs. New products and techniques are presented which demonstrate a potential solution sets to this critical issue.

2:35

3pNSb2. Assessing the performance capabilities of tactical hearing protection and communication devices. Hilary Gallagher (Battlespace Acoust. Branch, Air Force Res. Lab., 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, hilary.gallagher.1@us.af.mil), Eric R. Thompson (Battlespace Acoust. Branch, Air Force Res. Lab., WPAFB, OH), and Richard L. McKinley (Oak Ridge Inst. of Sci. and Education, WPAFB, Ohio)

Military personnel work in unpredictable noise environments, which require flexible types of hearing protection (i.e., tactical hearing protection systems) in order to maintain mission effectiveness and situation awareness while reducing the risk of hearing loss. Acquisition decisions need to be made relative to accurate and complete measures of the total performance capabilities of tactical hearing protection systems and their effect on the user. Understanding the noise attenuation performance of tactical hearing protection systems has been a priority in order to protect the user from excessive noise exposure. However, active electronic tactical hearing protection systems have been designed to allow for enhanced communication and situation awareness, while at the same time protecting the auditory system from both impulsive and continuous noise. The Air Force Research Laboratory conducted a multifactorial assessment on currently available tactical hearing protection systems to determine the overall impact of these devices on performance and to enable users to make data-driven, informed acquisition decisions. The assessments included the following measurements: continuous noise attenuation, impulsive peak insertion loss, auditory localization, and speech intelligibility. The methods and results will be presented as well as a discussion on how the results promote an informed device selection.

2:55

3pNSb3. Noise reduction ratings of active noise cancellation headsets. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov), Hilary L. Gallagher, and Richard L. McKinley (Battlespace Acoust. Branch, US Air Force Res. Lab., Wirght Patterson Air Force Base, OH)

The recent promulgation of the MIL-STD 1474 E, “Design Criteria Standard Noise Limits,” combines data from the Real Ear Attenuation at Threshold (REAT) and the Microphone in Real Ear (MIRE) measurements to estimate the noise reduction rating of active noise cancellation (ANC) hearing protection devices. This paper will examine the American National Standards Institute standards, S12.6, S12.42, and S12.68 to assess the performance of four aviation headsets measured with REAT and MIRE methods. We will compare the protection estimates for different noises with the Noise Reduction Statistic for A-weighting (NRS_A), Noise Reduction Statistic

3p WED. PM

for Graphical (NRS_G), and Octave Band rating methods. The NRS_G method provides an efficient and accurate alternative to the Octave Band rating method. In the International Standards Organization, the ISO 4869 standard part 6 is under development to measure the attenuation of ANC protectors. Comparison with the proposed ISO 4869-6 standard will be conducted.

3:15

3pNSb4. Sound localization performance with level-dependent hearing protection devices. Eric R. Thompson and Griffin D. Romigh (711th Human Performance Wing, Air Force Res. Lab, 2610 7th St., Bldg. 441, Wright Patterson, OH 45433, eric.thompson.28@us.af.mil)

Level-dependent hearing protection devices (HPDs) offer the promise of the ability to hear low-level sounds while providing protection against dangerous high-level sounds. These devices are useful for people who may be exposed to intermittent, unexpected loud sounds while operating in environments that are otherwise generally quiet. Often, it is important for these users to be able to localize sounds and to maintain auditory situation awareness in these environments while wearing the HPDs. Measurements were made of listeners' abilities to localize quiet and loud sound sources while wearing different level-dependent devices, as well as with unprotected ears. Electroacoustic measurements were also made with the devices on an acoustic test fixture to characterize their input-output functions, effective bandwidths, and directional transfer functions. Sound localization performance was typically worse when wearing the HPDs than with unprotected ears, with more front/back confusions, especially for the devices that have the least variance in their directional transfer functions. Auditory models were used to predict the human localization performance with the electroacoustic measurements as inputs.

3:35–3:50 Break

3:50

3pNSb5. An objective, efficient auditory situation awareness test battery for advanced hearing protectors and tactical communications and protective systems (TCAPS): DRILCOM (detection-recognition/identification-localization-communication). John G. Casali and Kichol Lee (Auditory Systems Lab, Virginia Tech, 250 Durham Hall, Blacksburg, VA 24061, jcasali@vt.edu)

An objective, computer-controlled test battery was developed for measuring the effects of hearing protection devices (HPDs) and TCAPS, on auditory situational awareness (ASA). Four independent ASA tasks with military relevance were tested: Detection-Recognition/Identification-Localization-Communication (DRILCOM). Detection employed threshold-shift with seven 1/3-octave bands, an AK-47 burst, and rifle cocking. Detection signals emanated from in front of the subject (0 degrees), or at 90, 180, or 270 degrees, to evaluate directionality. The Recognition/Identification task employed 36 signals combined in triads, wherein one "target" sound from 0 or 90 degrees had to be identified, at S/N ratios of -10, 0, and +10. Localization employed a dissonant pure tone chord, spanning 104-7880 Hz, which provided interaural time and interaural level cuing. Localization entailed 360-degrees of azimuth in 30-degree increments, and 30-degrees in frontal elevation, at a S/N of +10. The Communications task, addressing pass-through communications, utilized QuickSIN messages from 4 directions. In a proof-of-concept experiment on 10 subjects, 2 in-ear TCAPS, 1 earmuff TCAPS, a passive Combat Arms Earplug in "open" setting, and an EB-15LE™ electronic earplug were tested, as well as the open ear. DRILCOM demonstrated sensitivity to performance differences among devices and the open ear, as well as diagnosticity to within-device performance differences across different ASA tasks. The results have implication for selection and deployment of TCAPS and HPDs.

4:10

3pNSb6. Standard methods for measuring the effects of special hearing protectors on auditory performance. Richard L. McKinley (Battlespace Acoust., Air Force Res. Lab., 2610 Seventh St., AFRL/711HPW/RHCB, Wright-Patterson AFB, OH 45433-7901, rich3audio@aol.com), Hilary L. Gallagher, and Eric R. Thompson (Battlespace Acoust., Air Force Res. Lab., Dayton, OH)

Traditional hearing protectors attenuate noise via passive and or active technology. The advent of special hearing protectors, i.e., those that provide communications capability, ambient listening/situation awareness function, and/or level dependent attenuation of continuous and/or impulsive noise, has generated additional needs for performance measurement and characterization. ANSI Standards, representing the consensus of the U.S. scientific community, provide accurate, reliable, and repeatable methods for measuring performance. Current standards address the measurement of passive noise attenuation, active and level dependent insertion loss, impulsive peak insertion loss, and speech intelligibility. However, a need exists for a standard measurement method/s to characterize sound localization ability and auditory situation awareness. The presentation will address the application of current standards as well as a framework for a new standard describing three possible methods to measure sound localization performance and auditory situation awareness.

4:30

3pNSb7. Evaluation of extended-wear earplug technology as a transparent hearing-protection device. Douglas S. Brungart (Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil), LaGuinn Sherlock (Army Public Health Ctr. (Provisional), Bethesda, MD), Nandini Iyer, Elizabeth A. McKenna, Eric R. Thompson (Air Force Res. Lab., WPAFB, OH), and Ashley C. Zaleski (Walter Reed NMMC, Bethesda, MD)

One of the greatest challenges in hearing conservation is providing adequate protection for those listeners who spend most of their time in quiet environments, but are occasionally exposed to high levels of noise that occur at unpredictable times. A classic example of this problem occurs for dismounted soldiers, who rely heavily on their natural hearing acuity to detect threats and opportunities on the battlefield but always face the risk of becoming engaged in a dangerously noisy firefight with little or no warning. One emerging technology that might have application in this domain is the extended-wear hearing aid, which is capable of remaining deep in the ear canal for weeks or months before being removed. Preliminary results suggest that the extended wear earplug can provide protection from blast or continuous noise that is comparable to a conventional earplug or muff, and that it can provide detection and localization performance comparable to that achieved with the open ear. Its form factor also makes it more likely to be compatible with headsets, radios, and personal protective equipment than traditional earplug or earmuff devices. The potential benefits and limitations of the extended-wear

earplug as a hearing protector will be discussed. [Work supported by MRMC Grant W81XWH-14-1-0254; The opinions and assertions presented are the private views of the authors and are not to be construed as official or as necessarily reflecting the views of the Department of Defense.]

Contributed Papers

4:50

3pNSb8. Effect of a level-dependent hearing protector on detection thresholds, perceived urgency, and localization performance of reverse alarms. Chantal Laroche, Christian Giguère, Véronique Vaillancourt, Evgenia Shmigol, Mani Rahnama, Anne Gravel, Tanya Vaillancourt, Jérémie Chiasson, and Véronique Rozon-Gauthier (Audiol. and SLP Program, Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H 8M5, Canada, claroche@uottawa.ca)

Previous studies have shown that perception of warning signals in noisy workplaces can be compromised when wearing conventional hearing protectors. Level-dependent hearing protectors have been developed to increase environmental awareness and, by extension, to improve worker safety. A set of studies was conducted to assess the potential benefits of a level-dependent hearing protector operating in its passive and active modes, in quiet and in noise, for a range of listeners with normal hearing listening to reverse alarms. As expected, the detection of reverse alarms in quiet was improved in the active mode compared to the passive mode, but no significant benefit was found in noise. The active mode of the device only marginally helped to restore the sense of urgency conveyed by back-up alarms in noise, which was largely affected by passive attenuation.The localization of reverse alarms was slightly worse when wearing the device compared to unprotected listening, and was not improved in the active mode compared to the passive mode. In summary, when choosing a hearing protector to ensure adequate perception of reverse alarms, for example, different psychoacoustic tasks must be considered, also keeping in mind that level-dependent hearing protection does not necessarily restore performance to unprotected targets.

5:05

3pNSb9. Speech perception and production with one level-dependent hearing protector. Christian Giguère, Chantal Laroche, Véronique Vaillancourt (Audiology/SLP Program, Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H8M5, Canada, cgiguere@uottawa.ca), Ghazaleh Vaziri, Nicolas N. Ellaham (School of Rehabilitation Sci., Univ. of Ottawa, Ottawa, ON, Canada), and Hilmi R. Dajani (School of Elec. Eng. and Comput. Sci., Univ. of Ottawa, Ottawa, ON, Canada)

Electronic level-dependent hearing protectors aim to enhance communications and situational awareness in the noisy workplace compared to passive hearing protectors. From a listener's perspective, the fixed attenuation provided by passive hearing protection interferes with speech recognition in noise in users with hearing loss. From a talker's perspective, they also disrupt the Lombard reflex and induce an occlusion effect, which leads to a reduced vocal output in noise and compounds the problem. This paper presents laboratory data on speech recognition and production in noise with one level-dependent earmuff device. The device was tested in the level-dependent mode at two different gain settings as well as in a powered OFF condition to simulate passive protection within the same device. Level-dependent hearing protection provided significant speech recognition benefits in noise compared to passive protection, especially for users with hearing loss, and performance often exceeded unprotected listening. Talker speech output levels were also found to be higher in noise when the hearing protector was operated in a level-dependent mode compared to passive protection, but not as high as when unprotected, indicating only a partial recovery of speech production mechanisms.

5:20–5:35 Panel Discussion

Session 3pPAa

Physical Acoustics and Biomedical Acoustics: Solid State Acoustics

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Hirotsugu Ogi, Cochair
Osaka University, Machikaneyama 1-3, Toyonaka 560-8531, Japan

Chair's Introduction—1:10

Invited Papers

1:15

3pPAa1. Parity-time synthetic phononic media. Johan Christensen (ICMM, Madrid, ICMM, Madrid 28935, Spain, johan.christensen@gmail.com)

Classical systems containing cleverly devised combinations of loss and gain elements constitute extremely rich building units that can mimic non-Hermitian properties, which conventionally are attainable in quantum mechanics only. Parity-time (PT) symmetric media, also referred to as synthetic media, have been devised in many optical systems with the ground breaking potential to create non-reciprocal structures and one-way cloaks of invisibility. Here we demonstrate a feasible approach for the case of sound where the most important ingredients within synthetic materials, loss and gain, are achieved through electrically biased piezoelectric semiconductors. We study first how wave attenuation and amplification can be tuned, and when combined, can give rise to a phononic PT synthetic media with unidirectional suppressed reflectance, a feature directly applicable to evading sonar detection. Nature Rev. Mater. **1**, 16001 (2016); Phys. Rev. Lett. **116**, 207601 (2016); EPL, **114**, 4, 47007 (2016).

1:35

3pPAa2. Continuous monitoring of fatigue process with surface wave resonance: In the case of rotating bending fatigue. Hirao Masahiko (Osaka Univ., Grad. School of Eng. Sci., Toyonaka, Osaka 560-8531, Japan, hirao@me.es.osaka-u.ac.jp)

Prediction of fatigue failure of metals has not become reality like that of earthquake despite the intensive studies over many decades. But, there is an exception. The accumulating damage and the remaining life of rotating bending fatigue can be known from the ultrasonic observations through the lifetime. Electromagnetic acoustic resonance (EMAR) makes it possible to monitor the evolution of phase velocity, attenuation, and nonlinearity of surface shear wave exited and detected exclusively by electromagnetic acoustic transducer (EMAT), which operates without any contact with the rotating shafts. The probing shear wave travels in the circumferential direction with the axial polarization and penetrates into the metal to the submillimeter depth. Clear indications occur in the attenuation and nonlinearity measurements, and tell the pertinent metallurgical events within the penetration depth. They are caused by the dislocation mobility and restructuring due to cyclic loading, and occur at the specific fractions to the fatigue life, being independent of the bending stress amplitude. Continuous observation with EMAR can only detect such precursors and tell the remaining lifetime of metal components being fatigued. Discussions are given for the mechanism being based on the replication of small cracks and the TEM observation of dislocations at the crack-tip zone.

1:55

3pPAa3. Structure and Magnetism of $EuTiO_3$. Veerle M. Keppens (Dept. Mater. Sci. and Eng., Univ. of Tennessee, Knoxville, TN 37996, vkeppens@utk.edu)

The tilting of the oxygen octahedra in cubic perovskites is known to induce structural phase transitions, which are often associated with the emergence of intriguing physical phenomena. While $SrTiO_3$ is one of the most extensively studied perovskite oxides for its structural phase transition at 105 K, the discovery of magnetoelectric coupling in isostructural $EuTiO_3$ has triggered many theoretical and experimental studies focused on this compound. I will present resonant ultrasound studies of the extremely subtle cubic to tetragonal structural phase transition at 288 K in $EuTiO_3$ single crystals and the evolution of the physical properties upon chemical doping. The collective set of experimental data contributes to a better understanding of the link between lattice, magnetic, and electrical degrees of freedom in the $EuTiO_3$ system, helps evaluate the similarities and differences between $SrTiO_3$ and $EuTiO_3$, and provides insight in the possible origin of the much higher structural transition temperature for $EuTiO_3$ compared to $SrTiO_3$.

2:15

3pPAa4. Watching gigahertz acoustic waves confined in cavities. Oliver B. Wright (Faculty of Eng., Hokkaido Univ., Sapporo, Hokkaido 060-8628, Japan, olly@eng.hokudai.ac.jp)

Surface-acoustic and bulk-acoustic-wave wave devices based on waves in resonators have found extensive application in high-frequency signal processing. In particular, phononic crystals, metamaterials, and micron to sub-micron structures exhibit interesting physical properties, such as omnidirectional stop bands or tight wave confinement, that allow potential improvements to these devices. Here, we present results of real-time imaging and tracking of laser-excited acoustic waves at frequencies from 100 MHz up to ~100 GHz in various novel micron-scale cavity geometries: phononic-crystal slab cavity structures, metamaterial extraordinary-transmission structures and sub-micron fibres. Applications include gigahertz sound control and nanoscale acoustic imaging. [1] T. Dehoux *et al.*, Light Sci. Appl. **5**, 16082 (2016). [2] O. B. Wright and O. Matsuda, Philos. Trans. Roy. Soc. A **373**, 20140364 (2015). [3] J. J. Park *et al.*, Phys. Rev. Lett. **110**, 244302-1-5 (2013).

2:35

3pPAa5. From MHz to THz: How ultrasound and inelastic scattering see lattice dynamics. Raphael Hermann (Mater. Sci. and Technol., Oak Ridge National Lab., 1 Bethel Valley Rd. MS 6064, Oak Ridge, TN 37831, hermannrp@ornl.gov)

Resonant ultrasound spectroscopy and direct phonon spectroscopy methods based on inelastic scattering of x-rays or neutrons provide complementary but also sometimes apparently contradicting pictures of the lattice dynamics. In order to resolve these apparent contradictions, the probe energy and length scales must be considered. I will review results for a few classes of energy material, such as bulk [Sergueev *et al.*, Phys. Rev. B **91**, 224304 (2015)] and nanostructured [Claudio *et al. PCCP* **16,** 25701 (2014); *pss B* **251,** 919 (2014); **213,** 515 (2016); Klobes *et al.*, *Nanoscale* **8,** 856 (2016)] thermoelectric materials, magnetic and magnetocaloric materials [Herlitschke *Phys. Rev. B* **93,** 094304 (2016); Klobes *Phys. Rev. B* **92,** 014304 (2015)], or rutile oxides [Budai *et al.*, in preparation]. [Work at Oak Ridge National Laboratory was supported by the U.S. Department of Energy, Office of Science, Basic Energy Sciences, Materials Sciences and Engineering Division; work at Forschngszentrum Jülich was supported by the Helmholtz Association VH-NG-407, HRJRG-402, DFG SPP-1386, and BMBF 03X3540. The European Synchrotron Radiation Facility, Advanced Photon Source, Petra III at DESY, and Institut Laue Langevin are acknowledgment for beam time allocation. The invaluable contributions of all collaborators are grateful acknowledged.]

2:55–3:15 Break

Contributed Papers

3:15

3pPAa6. Tuning to a particular acoustic whispering-gallery mode in the GHz range. Sylvain Mezil, Kentaro Fujita, Montonobu Tomoda (Div. of Appl. Phys., Faculty of Eng., Hokkaido Univ., N13W8, Kita-Ku, Sapporo 060-8628, Japan, sylvain.mezil@eng.hokudai.ac.jp), Matt Clark (Faculty of Eng., Div. of Elec. Systems and Optics, Nottingham, United Kingdom), Oliver B. Wright, and Osamu Matsuda (Div. of Appl. Phys., Hokkaido Univ., Sapporo, Japan)

Surface Acoustic Waves (SAWs) generated with sub-picosecond light pulses are commonly used in non-destructive testing. The absorption of pump light by the medium generates SAWs that are detected by delayed probe light pulses. The spatiotemporal evolution of the SAWs can be imaged by scanning the position and time delay of the focused probe light pulses. Commonly used setups show two main constraints: the pump light is focused to a circular spot that generates SAWs in all directions, which does not allow control of directionality, and the laser repetition rate f_{rep} limits accessible frequencies to $nf_{rep}(n=1,2,\ldots)$. In the case of laser excitation and detection of whispering-gallery modes (WGM) on a disc, only certain frequencies are detectable, and counter-propagating modes cannot be separately excited. Here, we overcome these limitations in experimental GHz SAW imaging of WGMs. To access arbitrary frequencies, we periodically modulate in intensity of the pump and probe beams and make use of both in-phase and in-quadrature lock-in detection [1]-[3]. To generate WGMs propagating in a single direction, we make use of a spatial light modulator and computer-generated holograms. These new results extend the possibilities of SAW imaging by allowing fine control of excited surface acoustic modes. [1] S. Kaneko, M. Tomoda, and O. Matsuda, AIP Adv. **4**, 017124 (2014); [2] O. Matsuda *et al.*, IEEE Trans. Ultrason. Ferroelectr. Freq. Control **62**, 584-595 (2015); [3] S. Mezil *et al.*, Opt. Lett., **40**, 2157-2160 (2015).

3:30

3pPAa7. Discussion on very small temperature dependence of diamond's elastic constants. Akira Nagakubo, Hirotsugu Ogi, and Masahiko Hirao (Graduate School of Eng. Sci., Osaka Univ., 1-3, Machikaneyamacho, Toyonaka, Osaka 560-8531, Japan, akira.nagakubo@abc.me.es.osaka-u.ac.jp)

Diamond shows high Debye temperature (~2200 K) and small Gruneisen parameter (~1), which represent its small anharmonicity. Therefore, the temperature dependence of elastic constants is very small, preventing accurate measurement. Recently, we measured the temperature dependence of C_{11} of diamond by using picosecond ultrasonics. [A. Nagakubo *et al.*, Appl. Phys. Lett. **108**, 221902 (2016).]. Our results indicate that diamond has a further higher Debye temperature than previous measurements, and we found that elastic constants of high-Debye-temperature materials should be measured over a wide temperature range to extract the Debye temperature and the Gruneisen parameter, accurately. We also calculate the temperature dependence of C_{11} by an *ab-initio* method, which agree with our measurement. In this study, we discuss the temperature dependence of other elastic constants. We found that careful consideration is required for bond-bending and bond-stretching resistance to evaluate the temperature dependence of its elastic constants.

3:45

3pPAa8. Acoustic study of Wigner crystal melting in n-GaAs/AlGaAs at high magnetic fields. Alexey Suslov (NHMFL, Florida State Univ., 1800 E. Paul Dirac Dr., Tallahassee, FL 32310, suslov@magnet.fsu.edu), Irina Drichko, Ivan Smirnov (A. F. Ioffe PTI of RAS, St.-Petersburg, Russian Federation), Loren Pfeiffer (Elec. Eng., Princeton Univ., Princeton, NJ), Ken West (PRISM, Princeton Univ., Princeton, NJ), and Yuri Galperin (Dept. of Phys., Univ. of Oslo, Oslo, Norway)

We have measured absorption and velocity of surface acoustic waves (SAWs) in high-mobility samples *n*-GaAs/AlGaAs in magnetic fields (12–18) T (filling factors ν = 0.18–0.125) at temperatures T = (40–340) K and

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SAW frequencies f = (30–300) MHz. From the measurement data, the complex AC conductance, $\sigma^{AC}(\omega) \equiv \sigma_1(\omega)—i\sigma_2(\omega)$ and its dependences on frequency, temperature, and the amplitude of the SAW-induced electric field were found. We conclude that in the studied interval of the magnetic field and at $T < 200$ mK, the electronic systems forms pinned Wigner crystal, the so-called Wigner glass. The estimate of the correlation (Larkin) length of the Wigner glass is about 3 μm, which is much larger than both the distance between the electrons and the magnetic length in the studied field range. At some temperature T_m, the temperature dependences of both components of the complex conductance get substantially changed: from the dielectric behavior at $T < T_m$ to the metallic one at $T > T_m$. We ascribed this change of the conduction mechanism to melting of the Wigner crystal and studied the dependence of the so-defined melting temperature on the electron filling factor. [This work was supported by RFBR 14-02-00232, NSF Cooperative Agreement DMR-1157490, the State of Florida, the Gordon and Betty Moore Foundation through the EPiQS initiative Grant GBMF4420, and the NSF MRSEC Grant DMR-1420541.]

4:00

3pPAa9. Effect of stress wave irradiation on structural change in a colloidal system. Nobutomo Nakamura, Tasuku Okuno, Hirotsugu Ogi, and Masahiko Hirao (Graduate School of Eng. Sci., Osaka Univ., 1-3 Machikaneyama, Toyonaka, Osaka 560-8531, Japan, nobutomo@me.es.osaka-u.ac.jp)

Annealing is a typical method for controlling crystallographic structure of solids, and by increasing the number of phonon modes by heating, structural changes are caused. Considering that stress waves also enhance lattice vibrations, stress-wave irradiation should have similar effect on structural changes. In this study, we investigate the effect of stress-wave irradiation on crystallization of an amorphous solid using a colloidal system. A colloidal system used in this study is a mixture of a solution dyed with a fluorescein sodium salt and silica particles. It shows phases similar to those of atomic systems, and ordered (crystalline) and disordered (amorphous) structures are obtained. Because particle-scale analysis can be performed by using the confocal laser scanning microscopy, it has been used as a model system of atomic materials. We evaluate structural change of a colloidal glass after stress-wave irradiation and find that crystallization is accelerated at a specific condition. In the presentation, details of the results are described, and its origin is discussed.

4:15

3pPAa10. Resonance-enhanced compact nonlinear acoustic source of low frequency collimated beam for imaging applications in highly attenuating media. Cristian Pantea and Dipen N. Sinha (MPA-11, Los Alamos National Lab., MS D429, Los Alamos, NM 87545, pantea@lanl.gov)

Acoustic imaging in highly attenuating materials requires special acoustic sources that can generate a collimated beam of low frequency. Lower frequencies, in the range of 10-120 kHz, have the advantage of deeper penetration in the medium due to lower acoustic attenuation. However, typical acoustic sources at these low frequencies have a large beam spread, resulting in poor lateral resolution. We report on the latest advancements in the development of a very compact source, with approximate dimensions of a cylinder with a diameter of 25-50 millimeters and approximately 10 mm tall. Low frequency, collimated and steerable acoustic beam source are some of the main characteristics of this source. The newly developed source takes advantage of (1) frequency mixing in an acoustical nonlinear fluid in a cavity to generate the difference frequency of two high frequencies, around 1 MHz, and (2) resonance enhancement of the difference frequency in the cavity. An order of magnitude enhancement in amplitude was observed between on- and off-resonance conditions, with a collimation of the beam of approximately 6 degrees. Laboratory experimental data will be presented, and advantages over traditional and other nonlinear acoustic sources will be discussed.

4:30

3pPAa11. Resonant ultrasound spectroscopy study of novel ceramics designed for high temperature hydrogen fuel cells. Josh R. Gladden, Sumudu Tennakoon, and Ashoka Karunarathne (Phys. & NCPA, Univ. of MS, 108 Lewis Hall, University, MS 38677, jgladden@olemiss.edu)

Elastic constant measurements, along with temperature and pressure derivatives, are of interest to both the physics and engineering communities. One of the most precise and efficient methods for such measurements is resonant ultrasound spectroscopy (RUS) in which the vibrational resonance spectrum of a sample is used to determine the full elastic tensor of the material. In this talk, we will present the temperature dependent elastic constants of a series of ceramics, including glass ceramics and DMMA, between room temperature and 800C. These ceramics have been designed for use in high temperature fuel cell by LG Fuel Cell Systems. We will discuss the effects of thermal cycling. Some of the ceramics are porous in nature which increases acoustic damping. Damping mechanisms in porous materials and experimental techniques to reduce damping will also be discussed.

4:45

3pPAa12. Rapid wave velocity measurement by Brillouin scattering using artificially induced phonon. Yoshiaki Shibagaki, Masahiko Kawabe (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, duq0358@mail4.doshisha.ac.jp), Shinji Takayanagi (Nagoya Institude of Technol., Nagoya, Japan), Kazuma Mori (Doshisha Univ., Kyotanabe, Kyoto, Japan), Takahiko Yanagitani (Waseda Univ., Tokyo, Japan), and Mami Matsukawa (Doshisha Univ., Kyotanabe, Kyoto, Japan)

Brillouin light scattering enables measurements of hypersonic longitudinal and shear velocities in the small area (diameter about 50 μm). One problem of the conventional Brillouin technique is the weak Brillouin scattering peak from thermal phonons and this results in the longer measurement time. To overcome this problem, we have proposed measurements of strong coherent phonons by fabricating a high frequency transducer on the sample surface. However, the fabrication of the transducer is not suitable for simple and nondestructive measurements. In this study, a small glass device with a ZnO piezoelectric thin film was used as a hypersonic transducer around 1 GHz. By attaching this transducer to the sample with coupling liquid, artificial coherent phonons were successfully induced in a sample. In the case of a quarts sample, strong Brillouin scattering peaks were observed using a tandem Fabry-Perot interferometer. The measured frequency shift of the peaks was equal to the excitation frequency of the ZnO piezoelectric film (resonance frequency: 822 MHz). The intensity of Brillouin peak from the induced phonons was about 375 times as strong as the peak from the thermal phonons. The technique of induced phonons can realize nondestructive and rapid velocity measurements in the small area.

Session 3pPAb

Physical Acoustics and Biomedical Acoustics: Acoustic Micro- and Nanofluidics II

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Daisuke Koyama, Cochair
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Tony Jun Huang, Cochair
Duke University

Invited Paper

1:20

3pPAb1. Actuation and manipulation of nano-confined fluids and particles via room-temperature lithium niobate bonding. Morteza Miansari and James Friend (Mech. and Aerosp. Eng., Univ. of California, San Diego, 345F Structural and Mech. Eng., M.S. 411 Gilman Dr., La Jolla, CA 92093, jfriend@eng.ucsd.edu)

Controlled nanoscale manipulation of fluids and colloids is made exceptionally difficult by the dominance of surface and viscous forces. Acoustic waves have recently been found to overcome similar problems in microfluidics, but their ability to do so at the nanoscale remains curiously unexplored. Here, we show that 20 MHz surface acoustic waves (SAW) can manipulate fluids, fluid droplets, and particles, and drive irregular and chaotic fluid flow within fully transparent, high-aspect ratio 50-250-nm tall nanoslits fabricated via a new direct, room temperature bonding method for lithium niobate (LN). Applied in the same direction, SAW increases the capillary filling rate of the hydrophilic LN nanoslit by 2–5 times. Applied in opposition, the SAW switches the flow direction and drains the channel against 1 MPa capillary pressure, and can be used to controllably manipulate ~10 fL droplets. Finally, entire 10 μL droplets can be sieved via SAW through the nanoslit to pass only particles smaller than its height, providing pumpless size exclusion separation.

Contributed Paper

1:40

3pPAb2. c-axis tilted ScAlN shear wave acoustic Bragg reflect resonator for gigahertz viscosity measurement. Takeshi Mori (Nagoya Inst. of Technol., Gokiso-cho, Showa-ku, Nagoya, Aichi, 466-8555 Japan, Nagoya 466-8555, Japan, t.mori.397@nitech.jp), Yui Yamakawa, Rei Karasawa, Ko-hei Sano (Waseda Univ., Tokyo, Japan), Shinji Takayanagi (Nagoya Inst. of Technol., Nagoya, Japan), and Takahiko Yanagitani (Waseda Univ., Tokyo, Japan)

c-axis tilted ScAlN film is attractive for resonators exciting shear wave ultrasonic. Shear mode resonators are suitable for biomarker sensing and viscosity sensing because shear mode resonators can operate without energy leakage to liquid. These sensors are well known as a QCM (Quartz Crystal Microbalance) consisting of AT-cut quartz crystal which excites shear wave ultrasonic. When the resonator is immersed in liquid, resonant frequency decreases due to the viscosity perturbation layer between the resonator and liquid interface. The rate of resonant frequency decrease depend on the mass ratio of viscosity perturbation layer and the resonator layer. Therefore, thinner high frequency resonator allows high sensitivity viscosity measurement. The maximum resonant frequency of the standard QCM plate is about 30 MHz. On the other hand, c-axis tilted ScAlN film can operates over 1 GHz. The sensitivity of the piezoelectric film sensors are much higher than that of the QCM sensors. In this study, we deposited shear wave c-axis tilted ScAlN film on the acoustic Bragg reflector. We clearly observed the shear wave excitation at the 990 MHz by using a network analyzer in air. Apparent resonant frequency shift of -16 MHz and -28.5 MHz was observed, when the sensor was immersed in the ethanol and pure water, respectively.

Invited Paper

1:55

3pPAb3. Non-contact acoustic manipulation in microchannel. Teruyuki Kozuka (Elec. and Electronics Eng., Aichi Inst. of Technol., Yakusa-cho, Yachigusa 1247, Toyota, Aichi 470-0392, Japan, kozuka-t@aitech.ac.jp)

Noncontact micromanipulation technique is needed in micromachine technology, biotechnology, and other fields. In the present paper, a standing wave field is generated in a microchannel and a geometric space, and it is possible to trap small objects at nodes of the sound pressure distribution in the medium. A microchannel and a geometric space were made at the center of a glass plate of 50 mm x 50 mm x 5 mm. In the experiment, when the liquid water containing alumina particles was injected into the microchannel on glass plate irradiated by ultrasound, the particles flowed along several layers in the microchannel. In the geometric space, the particles were trapped in the sound pressure nodes of the standing wave field. By changing the frequency, the geometric pattern of particles aggregation interestingly changed. When the geometric space is a triangular space, the particle moved toward the top from the base of the triangle. When the microchannel is branched at the half circular geometric space, it was able to control the direction of the particle flow by changing the ultrasound frequency in the branched microchannels. Moreover, a sound field was numerically simulated by FEM under the experimental conditions, and the experimental results were discussed.

Contributed Papers

2:15

3pPAb4. A study of the limits of microparticle collection over a broad frequency range. Prashant Agrawal, Prasanna S. Gandhi (Indian Inst. of Technol. Bombay, Mumbai, Maharashtra, India), and Adrian Neild (Dept. of Mech & Aero Eng., Monash Univ., Clayton, VIC 3800, Australia, adrian.neild@monash.edu)

Acoustic fields can be used to collect microparticles over a wide range of frequencies. Within the ultrasonic range acoustic radiation forces cause patterns of particles to form, typically along pressure nodes. At much lower frequencies (in the order of hundreds of Hertz), the collection mechanism is inertial in nature, a particle's inability to follow the fluid motion causes drag forces to act on the particle which are non-zero when integrated across a time cycle due to gradients in the flow field. In both cases, however, the ability to collect particles is limited by acoustic streaming—the steady state fluid flow which results from gradients in the harmonic flow field. In this study, we examine the smallest particle size that can be collected as a function of frequency. For the low frequency vibration, an open fluid chamber is oscillated in the horizontal plane, these conditions are also applied to the ultrasonic case, and the acoustic energy is kept constant across frequencies. It is found that the minimum particle size can be collected at each end of the frequency scale.

2:30

3pPAb5. Unconstrained manipulation of micro-particles using phase-control of standing ultrasound wave fields. John Greenhall, Fernando Guevara Vasquez, and Bart Raeymaekers (Univ. of Utah, 1495 East 100 South, MEK 1550, Salt Lake City, UT 84112, john.greenhall@utah.edu)

We demonstrate a method of unconstrained manipulation of a spherical microparticle submerged in a fluid medium using a standing ultrasound wave field. The method works by displacing the microparticle in small increments through independent adjustment of the phases and amplitudes of two opposing ultrasound transducers. We model the dynamic behavior of the microparticle during each incremental displacement, taking into account the acoustic radiation force and the time-dependent and time-independent drag force acting on the microparticle. Using this dynamic model, we characterize the transient and steady-state behavior of the fluid-microparticle system during each incremental displacement as a function of the microparticle and fluid medium properties and the phases and amplitudes of the ultrasound transducers. The results show that the displacement time and percent overshoot of the microparticle trajectory are dependent on the ratio of the acoustic radiation force and time-independent damping force. We experimentally demonstrate the method by manipulating the microparticle in an unconstrained manner over multiple wavelengths. In contrast to existing methods that require all acoustic reflections be absorbed, this method takes all reflections into account, removing the need for a complex experimental setup.

2:45

3pPAb6. Micro and nanofluidics of the cochlea: Trade-offs of sensitivity and noise in an active biological system. Aritra Sasmal and Karl Grosh (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, asasmal@umich.edu)

The cochlea performs an analog time-frequency analysis on the incoming acoustic signal via a coupled electro-mechanical-fluidic response. The ultimate mechanical event that triggers the opening of the sensory mechanically gated ion channels is the fluidic shearing of the stereocilia of the inner hair cells (IHC) of the cochlea. Each IHC is equipped with a tuft of free standing stereocilia which are bathed in a 2-6 micron viscous fluid gap between the reticular lamina and the viscoelastic tectorial membrane. We use thin-film lubrication theory and asymptotics to model and solve a coupled viscous fluid-structure interaction problem to predict HB sensitivity to disturbance. The noise at the HB is due to the dissipation by the viscous forces (as quantified by the fluctuation-dissipation theorem) and stochastic fluctuations in the channel. We used this analysis study the challenges faced by evolution to sense the range of acoustic frequencies important to mammals. Structural acoustic calculations show that the parameters that increase noise may also increase the sensitivity of the system. In this study, the parametric trade-offs between noise and sensitivity of the system has been elucidated.

Session 3pPP

Psychological and Physiological Acoustics: Recent Progress in Auditory Perceptual Organization Studies

Makio Kashino, Cochair
Human Information Laboratory, NTT Communication Science Laboratories, 3-1, Morinosato Wakamiya, Atsugi 2430198, Japan

Barbara Shinn-Cunningham, Cochair
Boston University, 677 Beacon Street, Boston, MA 02215-3201

Chair's Introduction—1:00

Invited Papers

1:05

3pPP1. Perceptual interactions between adjacent time intervals marked by sound bursts. Yoshitaka Nakajima (Dept. of Human Science/Res. Ctr. for Appl. Perceptual Sci., Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka, Fukuoka 815-8540, Japan, nakajima@design.kyushu-u.ac.jp)

Perceptual interactions take place between adjacent time intervals up to ~600 ms even in simple contexts. Let us suppose that two adjacent time intervals, T1 and T2 in this order, are marked by sound bursts. Their durations are perceptually assimilated in a bilateral manner if the difference between them is up to ~50 ms. When T1 $\lesssim$ 200 ms and T1 $\lesssim$ T2 < T1 + 100 ms, T2 is underestimated systematically, and the underestimation is roughly a function of T2—T1. Except when T1 $\simeq$ T2, this is assimilation of T2 to T1, partially in a unilateral manner. This systematic underestimation, time-shrinking, disappears when T1 > 300 ms. When T2 = 100 or 200 ms and T1 = T2 + 100 or T2 + 200 ms, T1 is perceptually contrasted against T2: T1 is overestimated. When 80 $\leq$ T1 $\leq$ 280 ms and T2 $\geq$ T1 + 300 ms, T2 is contrasted against T1: In this case, T2 is overestimated. Assimilation and contrast are more conspicuous in T2 than in T1. For three adjacent time intervals, T1, T2, and T3, the perception of T3 can be affected by both T1 and T2, and the perception of T2 by T1.

1:25

3pPP2. Informational masking of vocal signals in a nonhuman animal. Mark Bee (Ecology, Evolution, and Behavior, Univ. of Minnesota, 140 Gortner Lab., 1479 Gortner Ave., St. Paul, MN 55108, mbee@umn.edu)

Informational masking (IM) interferes with speech perception in the presence of multiple talkers. However, its impact on vocal communication in other animals has received little attention. This talk will present behavioral and neurophysiological evidence—much of it preliminary, circumstantial, or both—suggesting that frogs are also susceptible to IM in the context of vocal communication in noisy social environments. When choosing a mate in a chorus, female treefrogs must extract information related to a male's suitability and quality as a mate from the temporal envelope of his vocalizations. In behavioral experiments, temporally structured maskers constrain a female's ability to process the temporal envelopes of target vocalizations, even when targets and maskers are sufficiently different in frequency to be transduced by different inner ear papillae. (The frog's peripheral auditory system—consisting of multiple sensory papillae that encode different frequency ranges of airborne sound—provides a natural "spectral protection zone" for minimizing the impacts of energetic masking in experimental studies of IM.) In neurophysiological experiments, potential correlates of IM can be observed in the responses of single neurons in the inferior colliculus. Future work will employ a multi-tone masking paradigm to determine how spatial separation, masker uncertainty, and target-masker similarity impact "subcortical" signal processing mechanisms in the context of IM and vocal communication.

1:45

3pPP3. Uninformative dynamic visual stimuli aid in segregating two similar acoustic stimuli, but not in detecting a single stimulus in noise. Ross K. Maddox (Dept. of Biomedical Eng., Dept. of Neurosci., Univ. of Rochester, 1715 NE Columbia Rd., Box 357988, Seattle, Washington 98195, rkmaddox@uw.edu), Dean A. Pospisil, and Adrian K. Lee (Dept. of Speech and Hearing Sci., Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA)

Both the comprehension and detection of speech in noise are improved when the listener sees the talker's mouth. There are multiple reasons for this, from basic physical temporal correlations to higher order linguistic cues; we have recently performed several experiments investigating the former. They were based on artificial stimuli with speech-like dynamics but no linguistic information. Auditory stimuli were a tone or tone complex with randomly modulated amplitude. Visual stimuli were a disc with a randomly modulated radius. We manipulated the correlation between the visual stimulus and each auditory stimulus. In all experiments, the visual stimulus provided no information about the task. In the first study, we presented two competing auditory stimuli and had listeners respond to events in the

3p WED. PM

target stimulus (brief pitch or timbre fluctuations). Performance was better when the visual stimulus matched the auditory target than when it matched the masker. The second study employed a two-interval two-alternative forced choice detection task. Despite a range of stimulus variations, no effect of audio-visual coherence on auditory detection was ever observed. Taken together, these results suggest that listening improvements provided by visual stimuli derive from improvements in segregation and scene analysis, more than overcoming simple energetic masking.

2:05

3pPP4. Varieties of attention affect auditory perception of scenes. Joel S. Snyder, Breanne Yerkes, Vanessa Irsik, and Christina Vanden Bosch der Nederlanden (Psych., Univ. of Nevada Las Vegas, 4505 Maryland Pkwy, MS 5030, Las Vegas, NV 89154, joel.snyder@unlv.edu)

Past empirical work on perception of complex auditory scenes has demonstrated striking effects of attention on perception. Here, we present data from recent studies of auditory stream segregation and change deafness, demonstrating that different types of attention influence what is perceived. During auditory stream segregation, we found that the increasing tendency to perceive segregation as more alternations of low and high tones are presented (i.e., buildup) is facilitated by attending to the tone patterns, regardless of whether participants were actively making judgments of segregation. In contrast, performing the segregation task was necessary to enhance the influence of the frequency separation of the immediately prior tone pattern, suggesting a form of task-based (as opposed to stimulus-based) attention. During change detection, we found that cueing participants to which object is likely to be replaced in a scene of multiple objects facilitates successful detection of changes, while providing an invalid cue impairs detection. Furthermore, an object-encoding task performed after each change detection response suggests that attending to objects specifically involved in the change facilitates awareness of changes. These two lines of research together suggest the need to systematically investigate under what conditions various types of attention influence auditory perception.

2:25

3pPP5. Stimulus phase locking of cortical oscillation for perceptual organization. Hirokazu Takahashi (Res. Ctr. for Adv. Sci. and Technol., The Univ. of Tokyo, 4-6-1 Komaba, Meguro-ku, Tokyo 1538904, Japan, takahashi@i.u-tokyo.ac.jp)

The phase of cortical oscillations contains rich information and is valuable for encoding sound stimuli. We tested whether and how spatial patterns of oscillatory phase locking within the auditory cortex better predicted perceptual organization than those of response amplitude. A high-density microelectrode array with 10×10 sites within 4×4 mm^2 mapped LFPs at the 4th layer of auditory cortex in rats. First, in response to ABA-sequences with different inter-tone intervals and frequency differences, neurometric functions based on stimulus phase locking in the gamma band could better describe van Noorden's perceptual boundary than the LFP amplitude. Second, when a random tone sequence was switched to a regular tone sequence, the gamma-band phase locking increased immediately after the transition from random to regular sequences. The amplitude of the tone-evoked response, on the other hand, increased with frequency separation with respect to the prior tone, irrespective of putative organization of auditory objects. These results suggest that the evoked-response amplitude encodes the local order of tones, while the phase locking better predicts the perceptual organization.

2:45

3pPP6. Listening in crowded environments: How attention shapes brain responses to unattended sounds. Daniel Bates, Katharine Molloy, Nilli Lavie, and Maria Chait (Univ. College London, 332 Gray, London WC1X 8EE, United Kingdom, m.chait@ucl.ac.uk)

I will review recent work in my lab concerning the effect of attentional focus on brain responses to unattended sounds. The first series of experiments aimed to understand how the neural representations of tracked (attended) and ignored sources change with growing scene size. We used acoustic "scenes," comprised of multiple concurrent tone-pip streams, and a task which required listeners to track one of the streams over a long duration (30 seconds). EEG results show that attention boosts/reduces responses to the tracked/ignored sources, manifested as enhanced/reduced spectral power at its AM rate. The strength of this effect is dependent on the number of competing sources. We also demonstrate increased power coupling between activity in the alpha band and the brain responses to ignored sounds, implicating alpha in the process of active suppression of task-irrelevant information. In the second series of experiments (Molloy et al., 2015), we investigate the effect of focused visual attention on responses to simple sounds (pure tones). High, compared to low, perceptual load led to reduced early (~100ms) auditory-evoked activity and later suppression of the P3 "awareness" response. These findings support an account of shared audio-visual resources, which, when depleted under load, result in failure of perception. Reference: K. Molloy, T. D. Griffiths, M. Chait, N. Lavie (2015) "Inattentional deafness—an MEG study," *J Neurosci*.**35**, 16046-54.

3:05–3:25 Break

3:25

3pPP7. Auditory scene analysis: Patterns of things to come? Susan Denham (School of Psych., Univ. of Plymouth, Plymouth pl4 8aa, United Kingdom, s.denham@plymouth.ac.uk)

Perception depends on discovering and forming representations of sensory patterns to make sense of the world. Patterns playing out both in time and feature space allow sensory systems to form expectations, and provide a basis for decomposing the world into discrete objects and for detecting meaningfully novel events. Without expectations, the system has no way of autonomously evaluating its understanding of the world. However, patterns come and go. So, the perceptual system needs some way of allowing its operational representations to form and dissolve, dominate and yield, in a way that facilitates veridical perception. In this talk, I will discuss some of the issues of pattern discovery, maintenance, and deletion at different timescales and levels of complexity using exemplar models of SSA and auditory streaming, and new perceptual data.

3:45

3pPP8. Detecting auditory changes by pupillary response. Hsin-I Liao, Makoto Yoneya (Human Information Sci. Lab., NTT Commun. Sci. Labs., 3-1, Morinosato Wakamiya, Atsugi, Kanagawa 243-0198, Japan, liao.hsini@lab.ntt.co.jp), Shunsuke Kidani (Hokuriku Univ., Kanazawa, Japan), Nicolas Barascud (Laboratoire de Sci. Cognitives et Psycholinguistique (LSCP), Ecole Normale Supérieure, Paris, France), Sijia Zhao, Maria Chait (Ear Inst., Univ. College London, London, United Kingdom), Makio Kashino, and Shigeto Furukawa (Human Information Sci. Lab., NTT Commun. Sci. Labs., Atsugi, Japan)

Detecting changes is an essential function that helps the organism to constantly monitor and update the information from the environment. In this talk, I will review our recent work, demonstrating that the human pupillary response can be used as a physiological marker for certain aspects of auditory change detection, in both simple and complex acoustic streams. In study 1, using simple repetitive tones with occasionally presented deviants (oddballs), we found that the oddball induces a pupillary dilation response (PDR), that is modulated by the stimulus properties irrespective of whether attention is directed to the sounds (Liao *et al*. 2016). In study 2, we examined whether PDR also reflects changes in complex acoustic patterns. We adopted the stimuli in Barascud *et al*. (2016) which contain transitions between random and regular tone-pip patterns. Results revealed an asymmetry: the PDR was observed only to changes from a regular to a random pattern, but not vice-versa. Overall, the results suggest that the PDR is not simply evoked by any kind of perceptual change per se. Rather, the pattern of results is consistent with the PDR reflecting changes that violate expectation. Possible underlying mechanisms and future directions are discussed.

4:05

3pPP9. Investigating bottom-up auditory attention. Mounya Elhilali (Elec. and Comput. Eng., Johns Hopkins Univ., 3400 N Charles St. Barton Hall, Baltimore, MD 21218, mounya@jhu.edu)

In everyday life, we are surrounded by a cacophony of sounds that our brain has to sort through in order to focus on important information. The conspicuity of certain sounds allows them to grab our attention and “stand out” relative to the soundscape. In this work, we hypothesize that as sound statistics change over time, variance from these statistics drive bottom-up attention processes that direct our focus to certain objects in the scene. We investigate the perceptual space that renders certain sounds salient relative to their context, and examine neural underpinnings of emergence of these salient sounds using complex acoustic scenes as stimuli. Results reveal adaptive neural representations of the context and salient objects reflecting deviations from the statistical structure of the scene. We speculate that principles of predictive coding could explain a number of observed neural and perceptual findings and discuss relevance of such framework in understanding processes of auditory scene analysis.

4:25

3pPP10. Difficulty with selective listening in Autism spectrum disorder. Makio Kashino and I-Fan Lin (Human Information Lab., NTT Commun. Sci. Labs., 3-1, Morinosato Wakamiya, Atsugi, Kanagawa 2430198, Japan, kashino.makio@lab.ntt.co.jp)

Individuals with autism spectrum disorder (ASD) often experience difficulty with selective listening in the presence of multiple sounds despite their normal puretone thresholds. We have been conducting a series of studies to uncover neural bases for such difficulty in ASD adults without intellectual disability. First, we compared basic auditory functions between ASD and neurotypical (NT) groups, and found that (1) sensitivities to interaural time and level differences were lower in ASD than NT, and (2) a sub-group of ASD showed low sensitivity to temporal fine structure of acoustic waveforms. These findings suggest deficits in auditory processing in the brainstem of ASD. Second, we compared target detection thresholds between the two groups in the presence of distractors having little energetic masking on the target. Detection performance was dependent on the spectrotemporal coherence of target and distractor components in NT, whereas no such difference was observed in ASD, suggesting that ASD may lack automatic grouping across frequency channels. Additionally, we have observed significant differences between the two groups in behavioral, perceptual, autonomic, and brain responses to speech and/or non-speech stimuli. Taken together, difficulty with selective listening in ASD may involve several different mechanisms in both cortical and sub-cortical neural sites.

4:45

3pPP11. How auditory scene understanding can fail in special populations. Barbara Shinn-Cunningham, Scott Bressler, and Le Wang (Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu)

One way to gain insight into how the brain normally processes auditory scenes is to explore the perceptual problems in special populations. In our lab, we are studying two types of listeners who, although very different, both have trouble making sense of complex scenes: listeners with autism who are minimally verbal (MVA listeners), and blast-exposed military Veterans. Neither population shows evidence of specific deficits in how well information is encoded subcortically. However, both show deficits when it comes to analyzing sound mixtures with multiple sound sources. Neural responses from MVA listeners suggest that their brains do not automatically organize sound mixtures that typically developing listeners hear as distinct objects, which likely impairs their ability to analyze the content of one sound source in a mixture. In contrast, Veterans exposed to blast appear to have difficulties focusing selective attention on a sound source in order to select it from a mixture, consistent with behavioral deficits when trying to track one sound stream. The problems these two populations have in processing complex scenes supports the idea that successful everyday listening depends both on perceptual organization of the scene and top-down control of attention.

Session 3pSA

Structural Acoustics and Vibration, Physical Acoustics, Engineering Acoustics, and Signal Processing in Acoustics: Non-Contact Vibration Excitation and Sensing Techniques

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Chair's Introduction—1:15

Invited Papers

1:20

3pSA1. Measurement of the structural admittance matrix using external random noise sources. Earl G. Williams (Acoust., Code 7106, Naval Res. Lab., 4555 Overlook Ave., Washington, DC, DC 20375, earl.williams@nrl.navy.mil), Jeffery D. Tippmann (Scripps Inst. of Oceanogr., Univ. of California, LaJolla, CA), Sandrine T. Rakotonarivo (Lab. of Mech. and Acoust., Aix-Marseille Univ., Marseille, France), and W. A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California, LaJolla, CA)

The structural admittance **Ys** also called the structural Green's function characterizes the vibration, radiation, and scattering physics of a vibrator. When it is known, the vibration of the radiating surface, and hence the acoustic radiation from the surface, is also known for any specified surface load. We demonstrate in this talk that **Ys** can be constructed when the object is placed in an isotropic random noise field. This construction consists of ensemble averages of cross-correlations of the resulting total normal velocity and total pressure (incident plus scattered) measured over the complete surface of the object. However, the measurement of the surface fields could be prohibitive as the temporal frequency of interest increases. Instead of a surface measurement, one can use a dual conformal surface of microphones placed near to the object together with near-field acoustical holography (NAH) to determine the fields on the surface via the back projection capability of NAH. Although a large number of sensors may be required, the advent of inexpensive MEMS microphones combined with a 3D printing capability to construct the support structure makes this measurement very plausible. [Work supported by the Office of Naval Research.]

1:40

3pSA2. In-plane wavefield detection in lattice structures via digital image correlation. Massimo Ruzzene, Giuseppe Trainiti, and Marshall Schaeffer (Georgia Inst. of Technol., 270 Ferst Dr., Atlanta, GA 30332, ruzzene@gatech.edu)

We describe an experimental procedure to measure in-plane waves in an hexagonal lattice structure through digital image correlation. Measurement are performed by exciting the structure with a narrowband signal, then recording the evolution of the wavefield by using a single high-speed camera pointed at the lattice. By tracking the position of the lattice intersections, we are able to obtain the time evolution of the interpolated in-plane displacement field, which we use to separate the longitudinal and shear components of motion through Helmholtz decomposition. The proposed approach allows us to investigate the dispersion characteristics of P- and S-modes. For each of these two modes, we focus on the directionality of wave propagation by obtaining experimental dispersion surfaces, which are in excellent agreement with the theoretical ones. We also compare the RMS plots of the experimental displacement field to the components of the group velocity vector for the P and the S -modes computed theoretically, showing the direction where energy preferentially propagates in the structure. Digital image correlation is a promising tool for investigating in-plane motion in lattice structures, displaying interesting features with respect to analogous techniques, such as the 3D Laser vibrometry.

2:00

3pSA3. Electronic speckle-pattern interferometry: Techniques for non-contact vibration measurements of musical percussion instruments. Randy Worland (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Electronic speckle-pattern interferometry (ESPI) provides a non-contact optical system that can be used to record images showing operating deflection shapes of structures being driven sinusoidally at resonant frequencies. In high-Q systems, these shapes can approximate the normal modes of the test object. Due to the sensitivity of the optical measurement system, non-contact excitation can be achieved acoustically using a speaker. The use of ESPI is described in relation to measurements of musical bars, plates, and membranes under various boundary conditions, though the methods described can be applied to many other types of structures. In addition, a dual-ESPI system that provides simultaneous measurements of two surfaces of an object is illustrated (e.g., two heads of a musical drum). These measurements provide information regarding the degree of coupling and the relative phase of the two vibrating surfaces. Finally, the ESPI techniques are applied to the fluid-structure interaction of an orchestral crotale (a thick metal disk) at various degrees of submersion in water.

2:20

3pSA4. Non-contact source for nonlinear acoustics for crack detection. Pierre-yves Le Bas, T. J. Ulrich, Marcel Remillieux (Geophys. Group, EES-17, Los Alamos National Lab., MS D446, Los Alamos, NM 87545, pylb@lanl.gov), Brian E. Anderson (Brigham Young Univ., Provo, UT), and Lucasz Pieczonka (AGH Univ. of Sci. and Technol., Krakow, Poland)

Over the last decades, nonlinear acoustic techniques have been developed to detect mechanical damage in solids. They have been proven to be far more sensitive to early damage than standard linear acoustic techniques. Unfortunately, they often require high amplitude waves to propagate within the sample. To be practical in industrial applications, signals have to be generated without contact. Here, we will present a novel non contact acoustic source that can produce higher amplitude than standard transducers and how it can be used to image defects into a carbon fiber reinforce plastic plate damaged by an impact. We will show that we can not only image cracks but also separate two overlapping defects according to their orientation. [This work was supported by the US Department of Energy via the used fuel disposition campaign of the nuclear energy program.]

2:40

3pSA5. Automated acoustic evaluation of concrete bridge decks. Jinying Zhu (Civil Eng., Univ. of Nebraska-Lincoln, 1110 S 67th St., PKI 205B, Omaha, NE 68182, jyzhu@unl.edu), Suyun Ham (Civil Eng., Univ. of Texas at Arlington, Arlington, TX), and Hongbin Sun (Civil Eng., Univ. of Nebraska-Lincoln, Omaha, NE)

Chain drag testing is commonly used in current practice for bridge deck evaluation due to its low cost and ease of use. However, this method is subjective, and highly depends on the experience of the operators. Ambient noise caused by traffic affects the test speed and accuracy of results. This paper describes a recent research to develop an automated chain drag acoustic scanning system to detect delaminations in concrete structures, including bridge decks. The system consists of an array of chains, a noncontact MEMS microphone sensor array, multi-channel data acquisition device, positioning system, and signal processing schemes. The multi-channel design improves the spatial coverage and resolution of testing. An algorithm for interpreting acoustic signals from the automated chain drag test is developed. This automated system will enable real time visualization of tested areas. Compared to the conventional manual chain drag test, the automated system provides improved accuracy, spatial resolution, repeatability, and practicality.

3:00

3pSA6. Electromagnetic acoustic transducers for nondestructive inspection of dry storage canisters for used nuclear fuel. Hwanjeong Cho, Sungho Choi (Eng. Sci. and Mech., Penn State, 212 EES Bldg, University Park, PA 16802), Matthew Lindsey (Structural Integrity Assoc., State College, PA), and Cliff J. Lissenden (Eng. Sci. and Mech., Penn State, University Park, PA, Lissenden@psu.edu)

Inspection for stress corrosion cracking in stainless steel canisters that store used nuclear fuel must be done within a harsh environment that requires robotic delivery of the sensing system. The system must tolerate high temperatures and radiation and it must operate in constrained spaces. In order to be delivered robotically noncontact transducers are preferred. Electromagnetic acoustic transducers (EMATs) are noncontact transducers that can be made from materials capable of resisting high temperature and radiation and therefore are an excellent choice for this application. This presentation will describe the requirements of the dry storage environment, the ability of compact EMATs to detect cracks oriented both transverse and parallel to the wave vector using shear horizontal guided waves, and the nondestructive inspection methodology developed for full penetration weld regions. Laboratory experiments indicate that the pulse-echo mode is effective at detecting crack-like notches. B-scans constructed from the heat affected zone of welds show both the defects and the welds, which are distinguishable by their time-of-flight.

3:20–3:40 Break

Contributed Papers

3:40

3pSA7. Estimation of material properties using vibration magnification. Alyssa T. Liem and J. G. McDaniel (Dept. of Mech. Eng., Boston Univ., Boston, MA 02215, jgm@bu.edu)

This presentation describes the use of video magnification to estimate material properties of a structure within a region of interest. Video magnification is a recently introduced technique that magnifies motions which are mathematically present in a video but often not visible to the human eye. Several recent papers have shown that this technique may be used to derive quantitative information about vibration without contacting the structure. Furthermore, the technique amplifies all components of the three-dimensional vibration field that are present in the plane of the video. In the present work, a mechanical excitation is applied to the center of a region of interest on a structure. The excitation is concentrated in space and time. The response of the structure is captured at high temporal and spatial resolution by a video camera focused on the area of interest. Material properties are then estimated by adjusting values in a model of the region until agreement is obtained throughout the temporal and spatial responses in the video. The present work allows one to understand physical mechanisms and improve numerical models. Use of an impulsive excitation limits the extent of testing to a region of interest. Experimental results will be presented.

3:55

3pSA8. Extracting vibration characteristics of a guitar using finite element, modal analysis, and digital image correlation techniques. Jiaqi Huang, Kiran Patil, Javad Baqersad (Mech. Eng., Kettering Univ., 1700 University Ave., Flint, MI 48504, jbaqersad@kettering.edu), and Daniel Ludwigsen (Phys., Kettering Univ., Flint, MI)

The sound quality generated by the guitar depends on the vibration characteristics (i.e. natural frequencies and mode shapes) of this instrument. Thus, it is of particular interest to the guitar manufactures to be able to extract global information about the characteristics of the guitar. Traditional sensors can only measure at discrete locations. However, digital image correlation (DIC) can measure full-filed data on the surface of the structure. In the current paper, a finite element (FE) model of a guitar was developed using quadratic shell, and solid elements. An eigensolution was performed on the FE model to extract its natural frequencies and mode shapes. In order to validate the numerical results, a modal impact hammer test was performed on the instrument. Furthermore, a measurement was performed on the guitar using the DIC technique. In this measurement, the guitar was placed in a free-free configuration and was excited using a broadband excitation generated by a sound source. The response of the guitar to the excitation was recorded using a pair of high-speed cameras. The recorded images

were processed to extract the natural frequencies and mode shape of the guitar. The results show strong correlation between the numerical model and experimental results.

4:10

3pSA9. Modelling and experimental mapping of the ultrasound pressure field generated from focused ultrasonic transducers using fiber optic acoustic sensors. Songmao Chen, Alessandro Sabato, Christopher Niezrecki, and Peter Avitabile (Structural Dynam. and Acoust. Systems Lab, Univ. of Massachusetts Lowell, One University Ave., Lowell, MA 01854, Songmao_Chen@student.uml.edu)

The dynamic focused ultrasound radiation force has been recently used for excitation of structures with sizes ranging from micro to macro scale having a frequency bandwidth between tens of Hertz and up to 100 kHz. Therefore it can potentially be used as an alternative non-contact excitation method for conventional contact type excitations such as modal impact hammer or shaker excitations for experimental modal analysis. However, the dynamic focused ultrasound radiation force remains to be quantified and calibrated. In this paper, we present the results of numerically modeling and experimentally mapping the pressure field generated by a focused ultrasonic transducer (UT) with a radiating diameter of 50 mm and center frequency of 359 kHz. In the simulation, an acoustic model is created using the Rayleigh Integral and the boundary element method. For the experimental testing, a precision microphone and a fiber optic acoustic sensor are used to map the pressure field generated from the UT driven by double sideband suppressed carrier amplitude modulation signals. This novel fiber optic sensor is custom developed with a sensing element having a diameter of $\Phi 125\ \mu$m, enabling high spatial resolution measurement. The simulation and experimental results are compared and are shown to have good agreement.

4:25

3pSA10. The non-contact acoustic inspection method for concrete structures using the defect detection algorithm based on the statistic evaluation for a healthy part of concrete. Kazuko Sugimoto, Tsuneyoshi Sugimoto, Nobuaki Kosuge (Graduate School of Eng., Toin Univ. of Yokohama, 1614 Kurogane-cho, Aoba-ku, Yokohama, Kanagawa 225-8503, Japan, kazukosu@toin.ac.jp), Chitose Kuroda, and Noriyuki Utagawa (Satokogyo Co., Ltd., Atsugi, Kanagawa, Japan)

The deterioration of concrete structure becomes a social problem in Japan. Social needs increase as for maintenance, check and renewal of concrete structures such as a tunnel or a bridge. The check of concrete structures has been performed conventionally by a hammering test. Not the inspection depended on a human experience and sense, but the inspection by a quantitative measuring system are demanded. We have developed the technique "non-contact nondestructive acoustic inspection method" to measure the internal defects near the concrete surface to a depth of about 10cm using airborne sound and the laser Doppler vibrometer at a distance 5-10 m far from measurement surface. Depending on the surface state (reflectance, dirt, etc.) of concretes, there is a problem to decrease the quantity of the light of the returning laser and there arise optical noise resulting from the leakage of light reception. To remove an abnormal measurement point, we proposed a defect detection algorithm, in which vibrational energy ratio and spectrum entropy are combined. As a result, it enables to distinguish a defect part, healthy part and an abnormal measurement point. However, for a real concrete structure, the gray area exists. We evaluate quantity of acoustic characteristics against a healthy part of concrete statistically. We found the distribution of each acoustic characteristic against a healthy part follow a normal distribution by excluding some outlier. A defective part is separated from a healthy part clearly and visualized vividly.

4:40

3pSA11. Study on the measurement speed and signal to noise ratio of multi tone burst wave for high speed non-contact acoustic inspection method. Nobuaki Kosuge, Tsuneyosi Sugimoto, Kazuko Sugimoto (Graduate School of Eng., Toin Univ. of Yokohama, 1614,Kurogane-cho, Aoba-ku, Yokohama, Kanagawa 2258503, Japan, tm25b10u@ust.toin.ac.jp), Chitose Kuroda, and Noriyuki Utagawa (Tech. Res. Inst., SatoKogyo Co.,LTD., Atsugi, Kanagawa, Japan)

The non-contact acoustic inspection method using an air-borne sound can detect the cavity defect and crack near the measurement surface by using flexural resonance. By this method which uses the conventional single tone burst wave, since only one frequency was used for one sound wave emission, the length of measurement time had become a problem. Therefore, the multi-tone burst wave was devised for high speed improvement. However, measurement time and signal to noise (S/N) ratio change with the parameters in which a multi-tone burst wave is contained at the time of one sound wave emission, such as the number of frequency, the arranging method, pulse length, and the average number of times. Therefore, the experiment using the concrete test object having the styrofoam which imitated the cavity defect was conducted and a multi-tone burst wave was used, it was investigated how the measurement time and the S/N ratio of this method would change. From the experimental result, it became clear that the improvement in the measurement speed of several or more times was realizable maintaining a comparable S/N ratio as compared with the conventional single tone burst wave.

4:55

3pSA12. Remote acoustic sensing of mechanical changes in a plate in an unknown reverberant environment. Tyler J. Flynn and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, 1231 Beal Ave., Ann Arbor, MI 48109, tjayflyn@umich.edu)

An experimental method for remote acoustic sensing of changes in radiating structures in an unknown reverberant environment is presented. Acoustic radiation from a mechanical structure due to oscillatory forcing may be described by a frequency response function (FRF) dependent on the structure's material, geometry, and boundary conditions. Mechanical changes may alter the structure's FRF in detectible and potentially predictable ways. However, collecting suitable FRF measurements may be difficult in reverberant environments over the necessary extended measurement times due to the influence of reflected and scattered sound. Here, experimental results are presented for the remote acoustic detection of mechanical changes to a vibrating 0.3-m-square by 3-mm-thick aluminum plate in a reverberant environment. The plate has nominally clamped edges and is subject to swept-frequency base excitation. Sound from the plate with and without a mechanical change is recorded remotely with a 15 microphone linear array. These recordings are then processed using Synthetic Time Reversal to reconstruct a single radiated sound signal that is corrected for the environment's unknown reverberation, and this corrected signal is examined to detect mechanical changes to the test plate. Results from the proposed technique are compared with equivalent results from conventional approaches. [Sponsored by NAVSEA through the NEEC.]

5:10

3pSA13. High speed non-contact acoustic inspection method using multi tone burst wave. Tsuneyoshi Sugimoto, Kazuko Sugimoto, Nobuaki Kosuge (Graduate School of Eng., Toin Univ. of Yokohama, 1614, Kurogane-cho, Aoba-Ku, Yokohama, Kanagawa 2258503, Japan, tsugimot@toin.ac.jp), Chitose Kuroda, and Noriyuki Utagawa (Tech. Res. Ctr., SatoKogyo, Co. Ltd., Atsugi, Kanagawa, Japan)

The non-contact acoustic inspection method using an air-borne sound can detect the cavity defect and crack near the measurement surface by using flexural resonance. By this method using a single tone burst wave, the length of the measurement time of a two-dimensional scan was a practical problem. This is because only one frequency was used for one sound wave emission so that a high signal to noise (S/N) ratio could be obtained with the laser Doppler vibrometer (LDV) of a weak laser power (e.g., He-Ne 1mW). However, two or more frequency can be used at the time of one sound wave emission (multi tone burst wave) by taking into consideration the difference

of the propagation velocity of a sound wave and laser light. That is, it becomes possible to perform high-speed measurement as compared with the conventional method, using effectively the time & frequency gate for raising a S/N ratio by terminating measurement before the reflective sound wave reaches a laser head from the measurement surface. The experiment using a concrete test object was carried out and the validity of this multi tone burst wave as high-speed measurement was confirmed from the experimental result.

WEDNESDAY AFTERNOON, 30 NOVEMBER 2016 CORAL 3, 1:00 P.M. TO 5:30 P.M.

Session 3pSC

Speech Communication: Speech Production and Perception (Poster Session)

Elizabeth A. McCullough, Chair
Linguistics, Ohio State University, Box 352425, Seattle, WA 98195

All posters will be on display from 1:00 p.m. 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

3pSC1. A data-driven solution to the invariance problem in speech perception. Tasha Nagamine and Nima Mesgarani (Elec. Eng., Columbia Univ., 560 Riverside Dr. 17B, New York, NY 10027, nima@ee.columbia.edu)

A fundamental and unanswered question of speech perception is the ability of a listener to map the highly variable acoustic features of phones to perceptual categories such as phonemes. Several solutions to this problem have been proposed, which include the motor theory, direct realist, distinctive features, and exemplar models. In parallel, the same problem has been studied extensively in algorithmic emulation of speech perception, where the first step is to estimate phonemic categories from the acoustic signal. The recent breakthroughs in machine learning methods, particularly models inspired by neural circuits, have resulted in nearly human-level performance. However, the strategy exploited by the models in these data-driven approaches remains elusive. Here, we analyzed a deep neural network model trained to perform the acoustic-to-phoneme transformation and determined the representational and computational properties of the network as it forms categorical distinctions. We found that the network internally created a representation of phonetic features, such as manner and place of articulation, and explicitly modeled the phone variability. Furthermore, the network selectively and non-linearly warped the acoustic space to separate the more similar acoustic distinctions. This study provides an empirical solution to the invariance problem that can be evaluated in perceptual and neural studies of speech perception.

3pSC2. Mapping the functional pathways to speech intelligibility using neuroimaging techniques: A study of incremental speech perception. Emily Byers (Speech and Hearing Sci., Indiana Univ., 200 S Jordan Ave., Bloomington, IN 47405, elbyers@indiana.edu) and Jaimie Murdock (Cognit. Sci., Indiana Univ., Bloomington, IN)

Speech perception involves a complex feedforward/feedback process of decoding bottom-up auditory input while utilizing top-down cognitive/linguistic knowledge to map stored word forms onto incoming acoustic waves (Kinchla & Wolfe, 1979; Sloos & McKeown, 2015). Speech perception in the brain is believed to occur as a result of cortical spreading from the core of the auditory cortex (and peripheral areas), diffusing into non-speech-specific areas of the brain responsible for binding information and memory retrieval. The current study traces the functional pathways between regions associated with speech perception using a combination of fMRI and EEG imaging. Our principle aim is to determine how normal functional connectivity between language-processing areas differs from speech perception where speech ranges from unintelligible (e.g., English phonetic non-word sentences) to intelligible with spontaneous disruptions (code-switched utterances) up to fully intelligible speech. Testing disruption of speech intelligibility in healthy adults using naturally-occurring stimuli recreates the type of breakdown in intelligibility experienced by patients with language disorders as well as typically aging adults. Preliminary results indicate that spontaneous disruption of speech intelligibility due to mixed-language utterances differs from traditional information-masking stimuli both in reaction time to the stimulus change and in the amount of time required to resume online speech processing.

3pSC3. Aspects of generating syntax-free word lists for quantifying speech intelligibility. Stefanie Keller, Werner Hemmert (Bio-Inspired Information Processing, Tech. Univ. of Munich, Garching, Germany), and Florian Völk (WindAcoust. UG, Boltzmannstraße 11, Garching 85748, Germany, florian.voelk@mytum.de)

Several possibilities exist for quantifying speech intelligibility in noise, for example tests with single words or sentences. In the clinical routine, usually a single sentence test is used repeatedly, so that its vocabulary is often known to the patients. Thus, not only speech intelligibility but also long term memory is tested. New speech material is then required to explicitly address intelligibility. With the aim of providing new and optimized material, in this study, semantically, syntactically, and phonetically balanced words were taken from German sentence tests, based on databases for written and spoken German words. The resulting vocabulary consists of 54 words (nouns, adjectives, verbs, and numbers), for which word-reception thresholds were determined for ten normal-hearing subjects. With these thresholds, five-word lists of equal semantic context, equal word intelligibility, and with balanced phonetic cues were built. Non-acoustic advantages of an intelligibility test constructed this way are phonetic balance and semantic coherence without syntactic context, which aim at preventing syntactic and phonetic cues from affecting speech-intelligibility thresholds. As an initial verification, list intelligibilities were measured with the above subject sample. The results show no systematic difference for consonants and vowels. This indicates that the lists were successfully constructed without phonetic biases.

3pSC4. Eliciting natural conversational Lombard speech in realistic acoustic environments. Timothy Beechey (Linguist, Macquarie Univ., Australian Hearing Hub, 16 University Ave., Macquarie Univ., NSW 2109, Australia, tim.beechey@nal.gov.au), Jorg Buchholz (Linguist, Macquarie Univ., Sydney, NSW, Australia), and Gitte Keidser (National Acoust. Labs., Sydney, NSW, Australia)

Speech produced in noisy environments (Lombard speech) is characterized by a range of acoustic and phonetic changes. These changes stem from increased speaking effort which reflects communicative intent as well as decreased auditory feedback of the speaker's own voice. An accurate understanding of real-world Lombard effects is important in hearing science for the development and assessment of signal processing strategies targeting realistic speech signals. While Lombard effects are well known from the literature, studies of Lombard speech have typically been based on relatively unnatural speaking tasks such as reading from a script and have been measured in simplified acoustic backgrounds such as stationary noise or constructed babble noise. Lombard speech produced under such unnatural conditions may differ significantly from speech produced in real-world settings. This study describes a novel method of eliciting natural conversational speech across five highly realistic everyday acoustic environments. Through the increased realism of both the speaking task and acoustic backgrounds this study aims to provide a more ecologically valid approximation of real-world Lombard speech than has been previously reported. Based on recordings of conversations between 10 pairs of young, normal-hearing people, a continuum of ordered acoustic and phonetic changes in speech is described in relation to changes in acoustic environments and is related to self-reported listening effort ratings across acoustic environments.

3pSC5. Online testing for assessing speech intelligibility. Jonathan E. Peelle, Tracy Zhang, Nisha Patel, Chad S. Rogers (Dept. of Otolaryngol., Washington Univ. in Saint Louis, 660 South Euclid Ave. Box 8115, Saint Louis, MO 63110, peelle@wustl.edu), and Kristin J. Van Engen (Dept. of Psychol. and Brain Sci., Washington Univ. in Saint Louis, Saint Louis, MO)

The use of online testing in the behavioral sciences is increasing due to the potential for lower cost and faster completion than traditional in-person laboratory testing. Online testing presents a special challenge for speech research due to the variety of listeners' acoustic environments. For example, listeners may use headphones or speakers of various types, complete the task in different levels of background noise, or vary in their hearing abilities. Here we presented spoken sentences in speech-shaped noise to participants from the United States recruited online using Amazon Mechanical Turk at SNRs of -2, -5, and -8 dB ($n \geq 50$ per SNR). We compared these online results to normative data collected using these same sentences in the lab (SNR of -5 dB; $n = 30$). Preliminary results suggest a reasonable correspondence between average intelligibility scores for individual sentences presented using the two methods (Spearman rho = 0.79). Standard deviations suggest greater variability in the online responses (average SD across sentences = 1.24) relative to the lab responses (average SD = 0.86). We conclude that despite increased variability, online testing can provide reasonable ratings of intelligibility and in many circumstances is a feasible method to supplement or replace laboratory testing.

3pSC6. Variation in normal hearing thresholds predicts word recognition in noise. Kristin Van Engen (Washington Univ. in St. Louis, One Brookings Dr., Campus Box 1125, Saint Louis, MO 63130-4899, kvanengen@wustl.edu)

Analyses of speech identification data collected from young, normal-hearing adults often do not include measures of listeners' hearing. Sub-clinical variation in hearing, it is assumed, will not significantly affect the identification of speech signals presented well above threshold levels. This study tested that assumption in the context of an experiment that investigated young adults' recognition of words with few versus many phonological neighbors produced in clear and conversational speaking styles. Words were presented in speech-shaped noise at a +5 dB SNR. A mixed effects regression analysis was performed to account for the contributions of lexical difficulty, speaking style, order of word presentation, and listeners' hearing (as measured by pure-tone average (PTA)). This analysis showed, as expected, that correct identification was predicted by word type (easy > hard) and speaking style (clear > conversational). It also showed that identification improved over the course of the experiment. Finally, it showed that PTA was a significant predictor of identification for words produced in conversational speech, even for listeners with normal hearing (PTA range: -3.33-11.66 dB; mean: 2.23 dB). These findings show not only that sub-clinical hearing differences can significantly affect speech identification, but also that speaking clearly can mitigate the effect.

3pSC7. Interaction of cognitive load, sentence predictability, and speech degradation in spoken word recognition and memory. Cynthia R. Hunter, Caylee Adams, Laura Rund, and David B. Pisoni (Psych., Indiana Univ., 1101 E. 10th St., Bloomington, IN 47405, cynthunt@indiana.edu)

Whether cognitive load and/or listening effort needed to understand degraded speech differentially affect bottom-up and top-down processes is not well understood. The current project examined effects of sentence predictability, speech degradation, and cognitive load on the recognition of sentence-final words and on the recall of short (low-load) or long (high-load) sequences of visual digits presented prior to each spoken sentence. In addition to main effects of sentence predictability and spectral degradation, word recognition in both high- and low-predictability sentences was modulated by cognitive load when accuracy was between 35 and 80 percent. Words were identified more accurately under low load than high load. Digit recall was affected by load, speech degradation, and sentence predictability. Results indicate that cognitive load affects processes used to identify words in both low- and high-predictability sentences, and that listening effort affects memory for visual digits.

3pSC8. Isolating the informational component of speech-on-speech masking. Tim Schoof (Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, tim.schoof@northwestern.edu), Axelle Calcus (Boston Univ., Boston, MA), Stuart Rosen (Univ. College London, London, United Kingdom), Barbara Shinn-Cunningham (Boston Univ., Boston, MA), and Pamela Souza (Northwestern Univ., Evanston, IL)

Speech perception in the presence of a competing talker involves both energetic (EM) and informational masking (IM). This study aimed to isolate the informational component of speech-on-speech masking. EM can be eliminated by presenting the target and masker to opposite ears (i.e., dichotically). However, this also dramatically reduces the effects of IM by providing listeners with lateralization cues. Previous research using tonal sequences has shown that IM can be isolated by presenting the target and masker dichotically while rapidly switching the two streams across the ears. The question remains whether this technique can also be used for speech materials. Speech reception thresholds (SRTs) were measured for sentences produced by a female talker in the presence of a competing male talker under three conditions: diotic, dichotic, and switching. In the switching condition, target and masker were presented dichotically, but their lateralization was switched after every word in the target sentence. Preliminary data suggests that SRTs for the switching condition are higher than for the dichotic condition but lower than for the diotic condition. This suggests that, contrary to findings for tonal sequences, rapidly switching the target and masker speech across the ears does not fully reintroduce IM. [Work supported by NIH.]

3pSC9. Separation and integration of sound sources in auditory processing. Mio Horigome (Rehabilitation Dept., Okazaki City Hospital, 3-1 Goshoai, Koryuji cho, Okazaki, Aichi 4448553, Japan, horigome.mio@okazakihospital.jp) and Kazuhiko Kakehi (Study of Artificial Interigence, Chukyo Univ., Toyota, Aichi, Japan)

Some perceptual experiments were conducted to elucidate sound source separation processing in speech perception. The phenomenon of phonemic restoration was tested if it occurs in dichotic listening. An original speech is Japanese /VCV/, the stimulus is made by manipulating the /VCV/ as was replaced the consonant part with noise. The duration of replacement was varied at six levels: 10, 20, 40, 60, 80, and 100 ms. There were two hearing modes: in one case, the monaural stimulus was presented to both the ear, and in the other case, the speech part of the stimulus was presented to an ear

and the noise part to the opposite ear simultaneously. Thirty subjects in the age of twenties participate the listening test. The results show that phonemic restoration effect is clearly observed in both hearing modes. The rate of restoration is 92%, and 95% even in dichotic listening. There was no feature of speech in the noise has no relation with speech sound. Moreover, the direction of the noise sound was opposite to that of speech sound source, different in 180 degree. Even though the two sound signal are somehow integrated to hear restored speech.

3pSC10. Estimation of binaural speech intelligibility based on the better ear model. Kazuya Taira (Graduate School of Sci. and Eng., Yamagata Univ., 4-3-16 Jonan, Yonezawa, Yamagata 992-8510, Japan, thh04707@st.yamagata-u.ac.jp), Yosuke Kobayashi (Graduate School of Eng., Muroran Inst. of Technol., Muroran, Hokkaido, Japan), and Kazuhiro Kondo (Graduate School of Sci. and Eng., Yamagata Univ., Yonezawa, Yamagata, Japan)

When intelligibility is measured, objective estimation is more convenient than subjective. However, most objective estimation methods estimate monaural intelligibility using monaural signals. Therefore, it is necessary to estimate binaural intelligibility using binaural signals in order to take into account that a person listens using two ears. We evaluated an intelligibility estimation method using the Better Ear Model which selects better value out of the left and right feature values, and found that high estimation accuracy is possible. Accordingly, we proposed and evaluated an extension to this model which divides the signal into critical frequency bands, and takes the better value between left and right channel in each band (Band-Selection Better Ear Model). Furthermore, we tried estimation using machine learning in addition to regression analysis which we applied previously. Neural network and support vector machine were used here. Comparison between the Better Ear Model and the Band-Selection Better Ear Model showed no significant differences in the estimation accuracy. Also, the introduction of machine learning yielded a higher accuracy than that of regression analysis in a closed test. Correlation between subjective and estimated intelligibility increased by 0.1 or more. So far, mixed results are seen in the open test.

3pSC11. Semantic predictability mediates effects of phonetic reduction and social variation on word recognition. Zack Jones, Megan M. Dailey, and Cynthia G. Clopper (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, jones.5028@osu.edu)

Phonetic reduction due to lexical frequency, phonological neighborhood density, and discourse mention, as well as speaking style and social-indexical variation can constrain listeners' ability to identify spoken single word tokens. In phrasal contexts, however, semantic predictability facilitates word recognition. The aim of the current study was to investigate how semantic predictability influences the intelligibility of words that vary in their lexical, stylistic, and socio-indexical properties. Listeners were presented with auditory English phrases extracted from read passages and were asked to identify each phrase. Phrases were mixed with speech-shaped noise and each contained a target word of interest. Linguistic and social properties of the target words were used to predict listeners' target word recognition accuracy. These factors included semantic predictability, lexical frequency, neighborhood density, speaking style, discourse mention within the passage, talker gender, and talker dialect. The results revealed that greater semantic predictability increased word recognition accuracy, but only for the most phonetically reduced words (e.g., high-frequency words, second mentions, and words in a plain style). These results suggest that the semantic predictability benefit is enhanced primarily for words that might otherwise be difficult to recognize when removed from their semantic context.

3pSC12. Are suffixed words different? Evidence from a modified lexical decision task. Anne Pycha (Linguist, Univ. of Wisconsin, Milwuakee, P.O. Box 413, Milwaukee, WI 53211, pycha@uwm.edu)

We investigated whether listeners process suffixed words differently than prefixed words. We presented primes that combined a clear-speech root plus degraded-speech affix (such as *kin-xx,* where *xx* refers to degraded speech), and measured lexical decision RTs to subsequent clear targets (*kin-ship*). Degradation used low-pass filtering (< 500 Hz), such that affixes were speech-like but incomprehensible. Thus, the prime *kin-xx* sounds like a complete suffixed word, yet it is compatible with the target *kin-ship* and should not compete with it for activation. The crucial comparison was between prefixed (*re-group*) versus suffixed (*kin-ship*) targets, which were matched for frequency, familiarity, probability of phonotactic transition across morpheme boundary, and affix type-parsing ratio (Hay & Baayen, 2002). On each trial, participants heard a prime, a 1000 ms ISI, then a target to which they made a speeded decision. Pilot results ($n = 5$) suggest that comparable speech input activated suffixed roots less strongly than prefixed roots. While *xx-group* reduced RTs to *re-group* by a large amount (-236.65 ms), *kin-xx* reduced RTs to *kin-ship* by a much smaller amount (-133.20 ms). Pseudo-affixed words (*resort, worship*) did not show a comparable difference, suggesting that the effect arises from morphological constituency.

3pSC13. The effect of semantic predictability on word intelligibility in natural and spectrally degraded sentences. Bahar S. Shahsavarani, Thomas Carrell, and Ashok Samal (Univ. of Nebraska-Lincoln, Commun. Disord., University of Nebraska-Lincoln, Lincoln, NE 68583, bahar@huskers.unl.edu)

One of the most consistent findings across a wide range of methodologies and stimuli is the positive effect of semantic context and predictability on speech intelligibility. However, there is evidence that this may not be true for a limited range of stimuli. The present study investigated how sentences with spectrally-limited acoustic information constrained the use of semantic predictability in speech perception. This was studied by presenting listeners with spectrally-degraded and natural-speech sentences with high- and low-predictability words. The SPIN sentence lists and the associated multi-talker babble was used (Bilger, 1984). The results demonstrated that listeners benefited from high predictability in both cases but the degree of this effect was not equal. High predictability aided the perception of natural and 8-band speech equally. In contrast high predictability had a significantly smaller effect on speech intelligibility for 4-band speech. These findings are of special interest to investigations of speech perception in individuals with poor frequency representation such as those with cochlear implants.

3pSC14. Event-related potential responses reveal simultaneous processing across multiple levels of representation in spoken word recognition. Emma C. Folk and Joseph C. Toscano (Psych., Villanova Univ., 800 E Lancaster Ave., Villanova, PA 19085, efolk@villanova.edu)

A controversial issue in spoken language comprehension concerns whether different sources of information are encapsulated from each other. Do listeners finish processing lower-level information (e.g., encoding acoustic differences) before beginning higher-level processing (e.g., determining the meaning of a word or its grammatical status)? We addressed these questions by examining the time-course of processing using an event-related potential experiment with a component-independent design. Listeners heard voiced/voiceless minimal pairs differing in (1) lexical status, (2) syntactic class (noun/verb distinctions), and (3) semantic content (animate/inanimate distinctions). For each voiced stimulus in a given condition (e.g., lexical status pair TUB/tup), there was a corresponding pair with a voiceless ending (tob/TOP). Stimuli were cross-spliced, allowing us to control for phonological and acoustic differences and examine higher-level effects independently of them. Widespread lexical status effects are observed shortly after the cross-splicing point (i.e., the time when the lexical properties of the word can first be determined) and persist for an extended time. Moreover, there is considerable overlap in the times during which both lexical status and semantic content effects are observed. These results suggest that listeners multiple types of linguistic representations are active simultaneously during spoken word recognition, consistent with parallel processing models.

3pSC15. The perception of speech rate for time manipulated and vocoded sentences. Paul E. Reed, Allen Montgomery, Dan Fogerty, and Charley Adams (Commun. Sci. and Disord., Univ. of South Carolina, Keenan Bldg., Ste. 300, Columbia, SC 29201, reedpe@email.sc.edu)

Naturally produced fast speech reduces certain acoustic-phonetic features that may limit intelligibility relative to linear time compression. However, how reduction affects judgments of speaking rate has not been

systematically investigated. The purpose of this study was to examine the effects of linear time compression and spectral reduction on judgments of speaking rate. Listeners provided speech rate judgments of sentences. Conditions compared rate perceptions for naturally spoken sentences at slow and fast rates with linearly time-compressed/expanded versions of the same sentence, matched for duration. Conditions also examined rate judgments for noise vocoded sentences that varied in intelligibility and signal-correlated noise that examined rate judgments based only on temporal acoustic features. Our preliminary results demonstrate that linear time-compressed/expanded sentences were judged as faster than naturally produced sentences. This difference was also found for noise-vocoded versions of the sentences, as well as for signal-correlated noise. Additionally, vocoded stimuli were perceived as faster than naturally produced stimuli. These preliminary results suggest that acoustic-phonetic reductions in naturally produced speech do not appear to increase the perceived speaking rate relative to linear time manipulations. Significantly, temporal properties of speech rhythm appear responsible for coding perceptual aspects of speaking rate independently from factors related to linguistic processing.

3pSC16. Integration and maintenance of gradient acoustic information in spoken language processing. James B. Falandays (Psych., Villanova Univ., 38 North Cliffe Dr., Wilmington, Delaware 19809, jfaland1@villanova.edu), Joseph Toscano (Psych., Villanova Univ., Villanova, PA), and Sarah Brown-Schmidt (Vanderbilt Univ., Nashville, TN)

Models of speech processing seek to explain how continuous acoustic input is mapped onto discrete symbols at various levels of representation, such as phonemes, words, and referents. While recent work has supported models that posit maintenance of fine-grained information, it is not clear how continuous, low-level information in the speech signal is integrated with discrete, higher-level linguistic information. To investigate this, we created acoustic continua between the pronouns "he" and "she" by manipulating the amplitude of frication in the initial phoneme. Using the visual world eye-tracking paradigm, listeners viewed scenes containing male and female referents and heard sentences containing a pronoun, which later disambiguated to a single referent. Measures of eye-gaze revealed immediate sensitivity to both graded acoustic information and discourse-level information. Moreover, when listeners made an initially incorrect interpretation of the referent, recovery time varied as a function of acoustic step along the pronoun continuum, showing that graded acoustic information was maintained over at least a five-word delay. The results suggest that not only are listeners highly sensitive to fine-grained acoustic information in the speech signal but also that non-categorical representations are used to guide linguistic interpretation for extended periods of time.

3pSC17. Prediction of listener perception of reduced, voice stop consonant simulations based on patterns of formant deflections. Megan Willi and Brad Story (Univ. of Arizona, P.O. Box 210071, Tucson, AZ 85721, mkittles@email.arizona.edu)

Previous research on stop consonants found that less than 60 percent of the stops sampled from a speech corpus contained a clearly defined period of silence or prevoicing prior to the plosive release [Crystal & House, JASA, 1988]. How listeners perceive a reduced form of stop consonants without these cues is not well understood. The purpose of this experiment was to investigate whether recasting typical formant transitions into a measure called a "relative formant deflection pattern" provides a means of predicting listeners' perceptions of approximant-like, voiced stop consonant variants. A computational model of speech production, in which consonant constriction location was varied along the length of the vocal tract, was used to generate place continua of approximate-like, voiced stop consonants imposed on a vowel-to-vowel transition. Stimuli were presented to listeners in three conditions: 1) normal simulated speech, 2) sinewave speech in which three tones replicated the time course of the F1, F2, and F3 contours in the simulated samples, and 3) sinewave speech in which three tones were present, but selected combinations of F1, F2, and F3 were set to a flat contour. Perceptual responses will be compared to the predictions based on relative formant deflection patterns across conditions.

3pSC18. Cue shifting bias between acoustically correlated cues. Meng Yang (Linguist, Univ. of California, Los Angeles, 3125 Campbell Hall, Los Angeles, CA 90095, mengyang@ucla.edu)

Acoustic cues signaling the same phonetic contrast are weighted differently by listeners in sound categorization. From the body of research on perceptual cue weighting, two major views have emerged. On one hand, research has shown that secondary cue weights depend on the distributional informativeness of cues. On the other hand, cues that enhance the percept of other cues have been demonstrated to get a "boost" in cue weight. The current study extends these theoretical claims to cue shifting using breathiness and pitch as cues in a novel sound categorization paradigm that induces changes in cue weights. Two factors affecting cue shifting in adult listeners were tested: distributional learning and perceptual enhancement. While a change in cue distinctiveness did cause listeners to redistribute cue weights to favour the more distinctive cue, cue shifting was not equal in all conditions. The shift was facilitated when the task allowed listeners to make use of the enhancing quality of the two cues. This finding cannot be explained by the distributional learning account. Additionally, listeners found it easier to shift cue weight from pitch onto breathiness than from breathiness onto pitch. This is problematic for both the distributional learning theory and the auditory enhancement theory.

3pSC19. Online perception of coda glottalization in American English. Adam J. Chong (UCLA, 3125 Campbell Hall, Los Angeles, CA 90095, ajchong@ucla.edu) and Marc Garellek (UC San Diego, La Jolla, CA)

In American English, voiceless codas /t, p/ are often realized with glottalization on the preceding vowel. Previous claims suggest that such glottalization can serve to enhance /t/ or, more generally, voicelessness of coda stops. This study examines the timecourse of word recognition to test these claims. 40 American English listeners participated in an eye-tracking study, where they heard synthesized glottalized and non-glottalized versions of CVC English words ending in /p, t, b, d/ while looking at a display with two words presented orthographically. Target words were presented with a minimal pair differing in place of articulation (e.g., cop-cot), or voicing (e.g., bat-bad, cap-cab). Our results indicate that listeners fixated to target words ending in /t/ marginally faster when they heard the glottalized version and when the competitor presented was a word ending in /p/. Glottalization did not result in faster fixation to targets for words ending in /p/. We also found a strong inhibitory effect—lower proportion fixation to targets—for words ending in voiced stops when the preceding vowel was glottalized. Altogether, these results support the claim that glottalization enhances voiceless codas (both /t/ and /p/), but that the effect is strongest for /t/.

3pSC20. The effect of co-occurrence restriction on perception of laryngeal contrast in Seoul Korean and East Kyungsang Korean. Hyun Kyung Hwang (National Inst. for Japanese Lang. and Linguist, Midori-Cho, 10-2, Tachikawa 1908561, Japan, hwang@ninjal.ac.jp)

This study investigates dialect differences between Standard Korean and East Kyungsang Korean with respect to the role of co-occurrence restriction on perception of laryngeal contrast in stops and fricatives. The results obtained from the identification test first confirm that the laryngeal contrast is present only in the stops, but not in the fricatives in East Kyungsang Korean. In order to further explore the role of laryngeal co-occurrence restriction in the two varieties of Korean, identification tests using nonce compounds were conducted. The results provide experimental evidence of the laryngeal co-occurrence restriction; Both Seoul and East Kyungsang speakers's perception of laryngeal contrast in stop consonants is conditioned by laryngeal specification of the preceding consonant. With respect to the perception of laryngeal contrast in fricatives, however, only Seoul Korean speakers exhibit the effect of the co-occurrence restriction. These results lead to the conclusion that the co-occurrence restriction plays a crucial role in the perception of laryngeal contrast both in stops and fricatives in Seoul Korean, while the laryngeal features are phonologically irrelevant for fricatives in East Kyungsang Korean.

3pSC21. How listeners recognise vowel sounds under highpass or low-pass filtering of vowel-specific frequency ranges. Dieter Maurer (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Toni-Areal, Pfingstweidstrasse 96, Zurich 8031, Switzerland, dieter.maurer@zhdk.ch), Thayabaran Kathiresan (Phonet. Lab., Dept. of Comparative Linguist, Univ. of Zurich, Zurich, Switzerland), Heidy Suter (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Zurich, Switzerland), and Volker Dellwo (Phonet. Lab., Dept. of Comparative Linguist, Univ. of Zurich, Zurich, Switzerland)

The present paper reports findings of two experiments on filtered sounds of the Standard German vowels /i-y-e-ø-ɛ-a-o-u/ produced by a female speaker at two fundamental frequencies $f_o = 220$ Hz and 659 Hz and a male speaker at $f_o = 131$ Hz and 523 Hz. High-pitched sounds were included in order to account for a possible impact of the f_o level on the perception of filtered vowel sounds. In the first experiment, the frequency region of the first formant of the sounds was highpass filtered, and in the second experiment, the frequency region of the second formant of the sounds was lowpass filtered. Vowel recognition of all sounds was investigated in a listening test. Results revealed shifts in the perceived vowel quality which varied across (i) vowel categories, (ii) f_o, and (iii) filter cutoff frequencies. Details of the filter parameters and of the perceived vowel quality shifts are given in the paper and implications for the relationship between acoustic cues and vowel recognition are discussed.

3pSC22. Perception of place of articulation of assimilated nasal and oral stops: What do response times and eye fixations tell us? Mercedeh Mohaghegh (Linguist, Univ. of Toronto, 309-15 Gamble Ave., Toronto, ON M4K 2H3, Canada, mercedeh.mohaghegh@mail.utoronto.ca) and Craig Chambers (Psych., Univ. of Toronto at Mississauga, Toronto, ON, Canada)

Two forced-choice identification and two visual world eyetracking experiments examined perception of the place of articulation (PoA) of word-final nasal and oral stops either in canonical (coronal) or assimilated form (coronal-to-labial place assimilation, e.g., '*phone^m/bat^p button*'). Listeners' response times (RT) and eye fixations were measured as they heard and, using a screen-based paradigm, identified assimilated or unassimilated words presented auditorily either in isolation (excised from recorded sentences) or with the assimilation-triggering context present (next word began with a labial consonant). Listeners were slower to identify isolated words ending in nasals, especially when words were assimilated. When the triggering context was present, RTs were overall faster and no longer different between nasal and oral stops. The eye fixation data further showed an early sensitivity to PoA cues carried by vowel transitions for assimilated oral stops. For words ending in assimilated nasal stops, however, fixation patterns only showed sensitivity to the PoA cues at a later point where the assimilation-triggering context was heard. These findings indicate a distinction in terms of perceptual uptake of acoustic cues between nasal and oral stops. The results also suggest the precise mechanisms involved in compensation for assimilation may vary across sound classes (nasal vs. oral stops).

3pSC23. Affricate-fricative perception in Korean listeners: Evidence for universal and language specific biases. Youngja Nam and Linda Polka (McGill Univ., 8th Fl., 2001 McGill College, Montreal, QC H3A 1G1, Canada, linda.polka@mcgill.ca)

Although language experience has a profound impact on phonetic perception, there is increasing evidence that phonetic perception is also shaped by universal biases which can be revealed as asymmetries in discrimination performance. In the present study, we explore potential perceptual asymmetries in adult Korean perception of four English affricate-fricative contrasts. Korean adults completed a native-language assimilation task and a category-based AX discrimination task with the phonemic contrast /tʃa-sa/ and non-phonemic contrasts /tʃa-ʃa/, /dʒa-za/ and /dʒa-ʒa/. Both voiceless contrasts—/tʃa-sa/ and /tʃa-ʃa/—were assimilated to distinct Korean affricate and fricative categories and were discriminated very well (>90%); performance revealed no perceptual asymmetries. Both voiced contrasts—/dʒa-za/ and /dʒa-ʒa/—were assimilated to the same Korean affricate category (/tʃa/) and were poorly discriminated (63-65%); performance was asymmetric on different pairs for both contrasts (fricative-affricate pairs>affricate-fricative pairs) and on same pairs for /dʒa-ʒa/ (fricative-fricative pairs>affricate-affricate pairs). These findings, and prior research, show that asymmetrical performance on different pairs is highly uniform and predicted by phone type, pointing to a possible universal bias favoring sharp amplitude onsets. However, asymmetries in same pair performance are predicted by language categorization and thus appear to be shaped by language-specific experience.

3pSC24. Cross-linguistic differences in articulatory timing lag in consonant cluster perception. Harim Kwon, Ioana Chitoran (Université Paris Diderot, 5 rue Thomas Mann, CEDEX 13, Paris 75205, France, harim.kwon@univ-paris-diderot.fr), Marianne Pouplier, Tom Lentz, and Philip Hoole (Ludwig-Maximilians-Universitaet, Muenchen, Germany)

Cross-language research on consonant cluster production has shown that consonant clusters in different languages are produced with different degrees of articulatory timing lags. The present study examines perceptual sensitivity to these cross-linguistic timing differences in consonant clusters. Native German listeners were tested on an AXB similarity judgment test using stimuli including consonant clusters produced by German and Georgian speakers. (German consonant clusters are produced with relatively shorter lag between two consonants than Georgian ones.) Stimuli were /bla, gla, gna/ syllables recorded along with articulatory (EMA) data. Short lag German tokens and long lag Georgian tokens were selected as A and B, with Xs of varying degrees of lag chosen from either Georgian or German recordings. Results showed that German listeners are sensitive to the cross-linguistic differences in articulatory timing lag: when the timing lag of X was closer to A, participants were more likely to choose A. Moreover, listeners' sensitivity was influenced by the types of clusters: listeners were more sensitive in /bla/ than they were in /gla/ and in /gna/. The effects on the similarity judgment of different measures of articulatory lag, of vocalic releases produced within clusters, and of other sub-phonemic details were investigated. Overall, the results show that the lag differences are salient to German listeners while German and Georgian clusters can differ in a number of respects.

3pSC25. Washback and language change: An investigation of teachers' perception of the Korean vowel length contrast. Goun Lee (Dept. of the English Lang. and Lit., Yonsei Univ., Seoul, South Korea, cconni80@gmail.com) and Dong-Jin Shin (Hankuk Univ. of Foreign Studies, Seoul, South Korea)

The goal of the current study is to investigate whether the vowel length contrast in Korean is retained in teachers' perception due to a washback effect. Previous studies have demonstrated that the vowel length contrast in Korean has been completely lost in the production of contemporary Seoul Korean (Kang *et al.*, 2015). However, considering that this vowel contrast is still being prescriptively taught in the elementary school curriculum, we hypothesized that teaching linguistically extinct contrasts might influence speech perception. In order to investigate whether Korean teachers have regained the vowel length contrast, we conducted two perception experiments. Twenty-three Korean elementary school teachers and 30 university students (control group) from the Seoul area completed a discrimination task as well as a forced-choice identification test in which vowel length contrasts were manipulated at a ratio of 1:3. The results showed that teachers' perceptual identification accuracy was significantly higher than that of the control group. Interestingly, the control group's identification accuracy also exceeded chance level, indicating that Korean listeners still retained some perceptual sensitivity. These results suggest that the Korean vowel length contrast may not be completely lost in perception, and also that a washback effect from language teaching may inhibit language change.

3pSC26. The N1 event-related potential component as an index of speech sound encoding for multiple phonetic contrasts. Olivia Pereira and Joe Toscano (Psych., Villanova Univ., 800 Lancaster Ave., Villanova, PA 19085, opereira@villanova.edu)

Listeners rely on many phonetic cues to perceive speech, but it is not clear how acoustic and phonological differences are encoded at early stages of perception. Previous work has begun to address this using the event-related potential (ERP) technique, demonstrating that the amplitude of the auditory N1 ERP component varies linearly with differences along VOT continua and suggesting that it can serve as an index of cue encoding. However, it is not clear how the N1 varies more generally for other phonetic distinctions. We present ERP data for a large set of naturally-produced word-initial minimal pairs spanning 18 consonants (/b,d,tʃ,f,g,dʒ,k,l,m,n,p,ɹ,s,ʃ,t,v,w,z/), as well as stimuli varying along voice onset time (voicing) and burst frequency (place of articulation) continua for all six stop consonants. The results reveal widespread differences in N1 amplitude for place of articulation and voicing. The N1 is larger for voiced consonants than voiceless consonants. Moreover, N1 amplitude as a function of place of articulation patterns differently for fricatives (larger N1 for alveolar and post-alveolar phonemes) than for stops (smaller N1 for these place of articulation categories). Overall, these results suggest that the N1 serves as a useful index of speech sound encoding across a range of phonetic contrasts.

3pSC27. Identification of stop consonants produced by an acoustically-driven model of a child-like vocal tract. Brad H. Story and Kate Bunton (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

A model of a child-like vocal tract has been developed such that the deformation patterns superimposed on a vowel substrate to generate coarticulated consonants are specified by a time-varying set of directional shifts in the first three resonance frequencies. These deflection patterns are denoted as a combination of three numbers each of which can vary between -1 and 1; a negative value implies a downward shift in a resonance frequency whereas an upward shift results for positive value. For example, a “bilabial” consonant specified as [-1,-1,-1] would be transformed via calculations of acoustic sensitivity functions to a time-varying vocal tract shape that presents the expected constriction at the lips, but also modifies other parts of the vocal tract that may be necessary for producing the appropriate formant transitions into and out of the consonant. Using this model, three sets of 30 VCV utterances were generated in which the values of deflection patterns were set to produce vocal tract shapes that hypothetically produce the stop consonants /b/, /d/, and /g/ embedded in three different vowel-vowel contexts. A perceptual experiment was performed to test their identification by listeners. [Work supported by NIH R01-DC011275 and NSF BCS-1145011.]

3pSC28. Asymmetries in the perception of sub-categorical variation. Kuniko Nielsen (Linguist Dept., Oakland Univ., 1025 HHB, Rochester, MI 48309-4401, nielsen@oakland.edu) and Rebecca Scarborough (Linguist Dept., Univ. of Colorado Boulder, Boulder, CO)

Both VOT and vowel nasality show low-level variation in English that may be perceptually motivated (and which, in any case, has perceptual consequences). This study examines the perceptual salience of such variation, investigating in particular a possible perceptual asymmetry between increased vs. decreased degrees of these phonologically relevant features. On each trial of an AXB perceptual task, listeners heard three repetitions of the same word: a token with unchanged phonetic feature (X) and tokens with phonetic features that were artificially increased and decreased to the same degree (A and B). Listeners then had to determine which of the two flanking items (A or B) sounded more similar to the middle item (X). Test words included 18 monosyllabic words with initial /p/ (in simple onsets) and 19 monosyllabic words with nasal codas and were heard by 40 listeners. Based on preliminary work, participants are expected to judge decreased-feature stimuli of both types (VOT and nasality) as more similar to the natural stimuli, suggesting that increasing these phonetic features is perceptually more salient than decreasing them. These findings demonstrate asymmetrical sub-categorical sensitivity for features known to be perceived categorically (VOT), as well as for features not primary to phonological contrast (vowel nasality).

3pSC29. Perceptual adaptation to nonnative speech by school-aged children. Tessa Bent and Julianne Frye (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

With training, listeners can improve their understanding of speech in adverse listening conditions, including those stemming from the environment (e.g., competing noise) and the talker (e.g., unfamiliar accents). For example, listeners can increase their ability to understand a specific nonnative talker and, when trained with multiple talkers with the same nonnative accent, can generalize their learning to novel talkers with that accent. The current study extends these findings by investigating children’s perceptual adaptation to nonnative speech. School-aged children can retune specific speech sound category boundaries, but little is known about their abilities to adapt to naturally produced nonnative speech. To assess children’s adaptation to nonnative speech, 6- and 9-year-old children were presented with sentences produced by multiple Mandarin-accented talkers and repeated what they heard (training condition) or were not given training (control condition). A posttest assessed their understanding of sentences produced by a novel Mandarin-accented talker. Performance on the posttest was more accurate for children in the training condition than those in the control condition, which may have resulted from task learning or perceptual adaptation. In addition to phonetic category retuning, the results suggest that children may be able to improve their decoding of nonnative speech.

3pSC30. Acoustic cues underlying the adjustment to talker sex in perception of fricative consonants. Ashley N. Moore and Matthew Winn (Speech & Hearing Sci., Univ. of Washington, 4217 NE 42nd St., Seattle, WA 98105, anmoore@uw.edu)

Sibilant fricatives are perceived mainly by spectral peak frequency, but are also influenced by surrounding context, which includes the vowel following the fricative. The vowel contains formant structure, formant transitions out of the consonant, and information about talker sex. These factors affect fricative categorization, even if fricative energy remains unchanged. For example, frequency components in voices are lower for men than for women, and the perceived boundary between relatively high (“s”) and low (“sh”) fricatives is accordingly shifted down for men. In this study, we examined various acoustic cues that underlie accommodation to talker sex, including fundamental frequency (F0), vocal tract length (formant spacing), and breathiness/spectral tilt as they affect fricative perception. Stimuli included a fricative continuum between “s” and “sh” and a variety of vocalic contexts spoken by women and men that were controlled to vary by the aforementioned cues. Listeners identified fricatives in “s”- or “sh”-onset words, and responses were analyzed using binomial logistic regression to measure the effect of contextual cues. Results demonstrate that despite the strength of F0 as a cue for talker sex, formant spacing and spectral tilt contribute most to the context effect, consistent with the importance of spectral contrast in speech perception.

3pSC31. Sociophonetic analysis of vowel variation in African American English in the Southern United States. Yolanda F. Holt (Commun. Sci. and Disord., East Carolina Univ., 300 Moye Bv 3310-X HSB, MS 668, Greenville, NC 27834, holty@ecu.edu)

Sociophonetic analysis is a quantitative method used to measure the dynamic acoustic properties of speech. Sociophonetic analysis pairs the demographic, geographic, and sociopolitical components of sociolinguistic inquiry with instrumental phonetic measurement techniques. This paper applies sociophonetic analysis to the study of regional vowel variation in African American English. Previous research has established robust variation in White American English. Only recently have explorations of dialect variation in African American English been completed. Analyses of vowel space area, vowel fronting, raising, and vowel dispersion reveal both regional alignment and racial/ethnic alignment of vowels produced by African American English speakers in the Southern United States. Regional variation in vowel production in African American English is observed. Findings are discussed with respect to the Southern Vowel Shift, the hypothesized African American vowel system and a proposed omniscient observer effect that directly impacts vowel production in African American English speakers in the United States. The impact of these findings is discussed for their importance in the analysis of theories of speech production and communication.

3pSC32. Bermudian English: An acoustic analysis of vowel properties with implications for sociophonetic variation. Nicole Holliday (Linguist and Cognit. Sci., Pomona College, Pomona College, Edmunds 185, 185 E Sixth St., Claremont, CA 91711, nicole.holliday@pomona.edu)

Bermudian English is a variety of interest due to its distinctiveness from most varieties of British and American English, but to date, few studies have documented the properties of the dialect's vocalic system (Ayers 1933). This study provides an initial description of the vowel systems of Bermudian English speakers, especially as compared to the systems of speakers of Mainstream U.S. English (MUSE). The study was conducted with seven native Bermudian speakers between the ages of 18 and 21. Subjects participated in a word list task as well as a picture task to elicit naturalistic speech. Results of regression analyses comparing F1, F2, and F3 values of Bermudian speakers with formant values of Mainstream U.S. English speakers from Hillenbrand (1995). indicate that Bermudian speakers differ from MUSE speakers in several striking ways. Bermudian speakers produce significantly fronter /u/ and /o/, and lower /i/ than MUSE speakers. Bermudian speakers also show a near complete merger of /ɛ/ and /ae/, though speakers also show some variation between productions in the word list vs. picture tasks, indicating a potential casual/careful distinction. Male Bermduian speakers also show a merger of /iɹ/ and /ɛɹ/ which is a stigmatized production according to speakers, though this does not occur for female speakers. The differences between casual/careful speech in front vowels and the gender differences in the production of prerhotic vowels provide valuable directions for future sociolinguistic and phonetic work on the variety.

3pSC33. Pre-nasal short-a in Northern California: Merged in formant space with /ɛ/, but distinct in duration and degree of nasal coarticulation. Georgia Zellou and Renee Kemp (Linguist, UC Davis, 469 Kerr Hall, Davis, CA 95616, gzellou@ucdavis.edu)

Short-a, or /æ/, production varies substantially across American English dialects and is noted to be useful in describing regional pronunciation differences. The current study examines the production and perception of /æ/ in California English, which raises in words with a final nasal coda (both velar and non-velar nasals). We explore both the acoustic characteristics of this potential merger with /ɛ/, along with lexical confusability between words with /æ/ and /ɛ/ in prenasal position. Productions of 32 native California speakers reveal that the formant space of prenasal /æ/ indeed overlaps with /ɛ/. However, these vowels have reliably different patterns of two secondary acoustic features: duration and nasal coarticulation. /æ/ is produced with both a longer duration and a greater degree of nasal coarticulation than /ɛ/. Results from a lexical identification task presenting unaltered, as well as duration-neutralized, items to 50 listeners reveal that these vowels are not confusable despite considerable formant overlap. Furthermore, modeling results indicate that greater differences in nasalization between prenasal /æ/-/ɛ/ pairs lead to more accurate identification. Overall, the results from this study indicate that these secondary features help maintain the perceptual distinctiveness of /æ/-/ɛ/ words where formant merger has taken place.

3pSC34. Understanding the relationship between acoustics and articulation of nasal and oral vowels. Marissa Barlaz, Sarah Johnson, Ryan Shosted (Linguist, Univ. of Illinois at Urbana-Champaign, 707 S. Mathews Ave., MC 168, Urbana, IL 61820, goldrch2@illinois.edu), Christopher Carignan (Western Sydney Univ., Sydney, NSW, Australia), Maojing Fu, Zhi-Pei Liang, and Brad Sutton (Beckman Inst., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

While real-time magnetic resonance imaging (rt-MRI) provides high spatiotemporal resolution for speech research, the associated audio is noisy, presenting a challenge for research on the relationship between articulation and the acoustic signal and solving the articulatory inversion problem. Using state-of-the-art denoising methods, the current study denoised rt-MRI audio associated with nasal and oral French vowels produced by one speaker, and extracted F1-3 from the midpoints of each vowel for /a, o, e/ and their nasal counterparts. Oblique images were taken of the velopharyngeal port at 25 frames/second, and average pixel intensity (API) in the velopharyngeal region was taken from images corresponding to the vowel midpoint. General additive models showed a significant relationship between API and F1 for oral and nasal vowels. (Lower API indicates a wider velopharyngeal opening.) F1 of /ã/ was lower than /a/, while F1 was higher in the nasal /õ/ and /ẽ/ than their oral counterparts, all of which are expected effects of naso-pharyngeal coupling. These results show that the acoustic recordings produced during rt-MR imaging can be used to explore the relationship between articulation and acoustics, and can give insight into the articulatory effects of nasality, something difficult to understand through acoustics alone.

3pSC35. The role of visual speech cues in sound change: A study of the cot-caught contrast among Michigan speakers. Jonathan Havenhill (Linguist, Georgetown Univ., 3700 O St. NW, Washington, DC 20057, jeh241@georgetown.edu) and Youngah Do (The Univ. of Hong Kong, Hong Kong, Hong Kong)

Interspeaker articulatory variation has been argued to be a driving force in sound change (Baker *et al.* 2011), yet the factors governing such variation are not entirely understood. Fronted /ɔ/ in the Northern Cities Shift is characterized by an increase in F2, and can be produced through either tongue-fronting or lip-unrounding (Havenhill 2015), while fronted /u/ in British English is achieved through tongue-fronting alone (Harrington *et al.* 2011). We investigate the hypothesis that visual speech cues restrict the ways in which articulatory patterns may vary between speakers. Participants were exposed to congruous and incongruous audiovisual nonce word stimuli containing /ɑ/ and /ɔ/, produced by a Michigan speaker. The perceived vowel was identified by selecting a rhyming English word (e.g., the choices for [zɔt] were "cot" and "caught"). When paired with visual lip-rounding cues, participants perceived auditory /ɑ/ as /ɔ/. Yet the same auditory stimulus presented with unround lips was perceived as /ɑ/, suggesting that the visual cue may be sufficient to maintain a contrast, despite acoustic similarity. However, conditions are favorable for merger when speakers produce both /ɑ/ and /ɔ/ with unround lips. As a result, visibly distinct articulatory variants may be preferred when pressure to maintain a contrast is high.

3pSC36. Mapping vowel categories at high fundamental frequencies using multidimensional scaling of cochlea-scaled spectra. Daniel Friedrichs, Stuart Rosen, Paul Iverson (Dept. of Speech, Hearing and Phonetic Sci., Univ. College London, 2 Wakefield St., London WC1N 1PF, United Kingdom, daniel.friedrichs@gmail.com), Dieter Maurer (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Zurich, Switzerland), and Volker Dellwo (Phonet. Lab., Univ. of Zurich, Zurich, Switzerland)

Recent studies have shown that accurate vowel category perception can be maintained at fundamental frequencies (f_o) up to at least 880 Hz. In such cases, the typical first formant frequency (F_1) of most vowels is exceeded by f_o and the vocal tract transfer function is to a high degree undersampled. It seems therefore unlikely that common formant patterns are the main acoustic features used by listeners to recognize vowels at high f_os. Here, we present results from multidimensional scaling (MDS) analyses calculated by averaging cochlea-scaled spectra across multiple steady-state vowels (N = 324; all 250 ms) produced by a female native German speaker at nine f_os within a range of 220-880 Hz. All vowels (/i, y, e, ø, ɛ, a, o, u/) were recognized accurately by listeners (N = 20) in a previous study [Friedrichs *et al.*, The phonological function of vowels is maintained at fundamental frequencies up to 880 Hz, J. Acoust. Soc. Am. **138**, EL36-EL42 (2015)]. MDS reveals that with increasing f_o, the vowel height dimension partially collapses, but the front/back distinction expands, thus allowing the vowels to be distinguished. This indicates that the perceptual space is reorganized and vowel height and frontness are being combined in a correlated way at higher f_os.

3pSC37. Stability of the main resonance frequency of fricatives despite changes in the first spectral moment. Christine H. Shadle, Wei-rong Chen, and D. H. Whalen (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu)

Spectral moments have been taken as the primary measurements of fricatives, but resonances are evident as well. To contrast the two, the X-ray Microbeam Database (XRMB) was used to investigate acoustic and articulatory behavior in [s] for 10 /sVd/ words and 9 /sCV*/ words for 24 subjects as in a previous study [Iskarous *et al.*, JASA 129:2, 2011]. Spectral

3p WED. PM

parameters were adapted from analysis of an acoustic corpus of adolescents' [s] production [Koenig *et al*., JSLHR 56, Aug. 2013]. The time series of the frequency of the main resonance (F_M) and the first spectral moment (M1) were compared to the constriction location estimated from the most anterior tongue pellet T1. F_M quickly settles into a constant value largely determined by the position of T1; vowel context affects F_M throughout [s] for some subjects. M1 however rises and then falls during [s], matching jaw-raising behavior (as estimated by the y-component of the jaw pellet) and correlated with an estimate of high-frequency energy. The results show that some aspects of the fricative acoustics behave differently over time and by context. It remains to be seen what the perceptual consequences of these differences are. [Work supported by NIH NIDCD-DC-002717.]

3pSC38. Acoustics of speech, articulatory compensation, and dental overjet in Cantonese. Lauretta Cheng, Murray Schellenberg, and Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, lauretta.cheng@alumni.ubc.ca)

Studies relating dental anomalies to misarticulations have noted that potential correlations appear to be obscured by articulatory compensation. Accommodation of tongue or mandible positions can help even individuals with severe malocclusion approximate perceptually typical speech [Johnson and Sandy, Angle Orthod. **69**, 306-310 (1999)]. However, associations between malocclusion and articulation could surface if examined with acoustic analysis. The present study investigates the acoustic correlates of Cantonese speech as it relates to degree of overjet (horizontal overlap of upper and lower incisors). Production data was collected from native Cantonese-speaking adults, targeting the vowels /i, u, a/, and fricatives /f, s, ts, ts^h/, previously found to be vulnerable phonemes in Cantonese speakers with dentofacial abnormalities [Whitehill *et al*., J Med Speech Lang Pathol. **9**, 177-190 (2001)]. Measures of dental overjet and language background were included as well. Preliminary results from trained listeners show that productions were perceptually typical. Acoustic analysis consisted of spectral moments for fricatives and formant values for vowels. The results improve our understanding of the relationship between malocclusion, compensation and speech production in non-clinical populations.

3pSC39. Reduction of intervocalic coronal stops in colloquial words, formal words and pseudowords in American English: A large-scale production study. Vsevolod Kapatsinski (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, vkapatsi@uoregon.edu)

Words occurring in reduction-favoring contexts have been hypothesized to become associated with reduced sound variants, reducing more than other words even when the contexts disfavors reduction. In a controlled production study, this paper investigates both this hypothesis and its converse: that words occurring often in reduction-disfavoring contexts become associated with non-reduced sound variants, compared to pseudowords. 130 speakers of American English were recorded reading sentences. Interspersed among fillers, designed to prime either formal or colloquial style, were sentences containing intervocalic /t/ and /d/ in the flapping environment. These sentences formed triplets placing different words in the same context, e.g., "*She is looking for the {butter; jitter; witter}*." Words expected to disfavor reduction were overattested in formal registers of the British National Corpus. Those expected to favor it were overattested in American movie subtitles, compared to British ones, and overattested in colloquial registers. Colloquial words exhibited significantly more flap and approximant realizations of the /t/ and /d/ than either formal words or pseudowords. Furthermore, within sound variant (flap vs. stop vs. approximant), colloquial words exhibited shorter closure durations, even with local speech rate controlled. However, pseudowords behaved like formal words, suggesting that full pronunciations are used by default, despite their vanishingly low frequency.

3pSC40. Coronal stop deletion in Hawai'i English. Kavon Hooshiar, Katie Drager, and Cassidy Copeland (Linguist, Univ. of Hawai'i at Mānoa, 1890 East-West Rd., UH 569 Moore, Honolulu, HI 96822, kavon@hawaii.edu)

Coronal Stop Deletion (CSD) is believed to be constrained by the phonological environment and the word's morphological make-up (i.e., monomorphemic, semi-weak past tense, regular past tense) (Guy 1991; Hazen 2011). In this talk, we present results from an analysis of CSD in the speech of four speakers of Hawai'i English. In addition to using a traditional approach (i.e., auditory analysis of CSD in word-final position), we also analyzed CSD in non-final position (e.g., followed by plural -s) and conducted acoustic phonetic analysis of /t/ in tokens where it was realized. The results demonstrate highly significant effects of environment. Additionally, there is no effect of morpheme type; instead, the results provide evidence that deletion is favored in words with a greater number of morphemes, regardless of whether the coronal stop is found in the stem or the affix. Additionally, the acoustic analysis suggests that the effect of following environment is acoustically gradient: tokens followed by a voiceless, non-alveolar consonant (i.e., the environment that most strongly disfavors deletion in our data) are most likely to be realized with complete occlusion in the vocal tract. Implications for models of speech production are discussed.

3pSC41. Temporal structure of repetition disfluencies in American English. Zara Harmon and Vsevolod Kapatsinski (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, zforough@uoregon.edu)

A repetition disfluency involves an interruption in the flow of speech followed by a restart, leading to repetition of one or more words. We analyzed the temporal structure of one-word, two-word, and three-word repetition disfluencies in the Switchboard Corpus ($n_{one\text{-}word} = 30546$, $n_{two\text{-}word} = 8102$, $n_{three\text{-}word} = 1606$). Comparing durations of words preceding an interruption point to their repeated counterparts, we find that repetition is typically accompanied by prolongation, which mainly influences the last word preceding the interruption point. When prolongation does not provide enough time for planning upcoming speech—as there seems to be a temporal limit to prolongation—the speaker repeats parts of the utterance just produced. Our results demonstrate that the number of words repeated is determined both by word duration and by co-occurrence relations between words. Mixed effects logistic regression analysis revealed that longer words are less likely to be repeated (z=-24.45, p<.0001). However, the number of words repeated does not reduce to duration: controlling for word duration, the more frequent a two-word sequence, the less likely it is to be interrupted by a restart (z=35.43, p<.0001). Longer repetitions are produced when a shorter repetition would result in restarting speech production from the middle of a cohesive word sequence.

3pSC42. Inferring velum movements from acoustic features. Hong Zhang (Linguist, Univ. of Pennsylvania, 619 Williams Hall, 255 S 36th St., Philadelphia, PA 19104, zhangho@sas.upenn.edu)

Although acoustical methods have been widely used in the literature on nasality, a direct link between the acoustic features and velum movements during speech production is yet to be established. In this study, we propose a model through which the vertical movements of the velum are inferred from an acoustic feature set. An X-ray Microbeam data set collected at University of Tokyo is used for the modeling. The data recorded the vertical movements of the velum of 11 American English speakers saying both isolated words and sentences. Velum positions are traced by a metal pallet placed on top of the velum. 40 MFCC (Mel-frequency Cepstral Coefficient) features are extracted from the accompanying acoustic signal at each time frame. MFCC features of five frames to both sides of the current frame, together with the current frame, consist of the 440-dimensional feature vector for predicting the velum position of the current frame. The feature vectors are then fitted to a lasso regression model. The inferred velum movements in the validation set are a good fit to the actual observation as judged by the high accuracy in identifying locations of peaks and valleys, and predicting the overall contour shape of the velum trajectory.

3pSC43. Speech adaptation to kinematic recording sensors. Christopher Dromey and Elise Hunter (Commun. Disord., Brigham Young Univ., 133 TLRB, BYU, Provo, UT 84602, dromey@byu.edu)

Measuring articulator movements can yield valuable insights into speech motor control, but attaching sensors to the articulators can affect speech. In order to measure the process of adaptation to the presence of kinematic sensors, 20 native English speakers read a sentence before the attachment of sensors to the tongue, lips, and jaw. They read it immediately afterwards, and then 5, 10, 15, and 20 minutes later. Participants read aloud continuously between recordings to encourage adaptation. Audio recordings were rated by 20 native English listeners using a visual analog scale with the endpoints labeled as *precise* and *imprecise*. Acoustic analysis involved segmenting the fricatives /s/ and /ʃ/ from the longer recording and computing spectral center of gravity and spectral standard deviation as well as fricative duration. Results of both perceptual and acoustic analysis revealed a change in speech precision over time, with all post attachment recordings receiving reduced precision ratings. Spectral mean decreased and standard deviation increased following sensor attachment; durations decreased over time. Precision ratings beyond the ten minute recording remained steady. The results show that speakers reached a height of adaptation after 10 minutes but did not at any point return to pre attachment precision levels.

3pSC44. Effect of tongue position in the simplified vocal tract model of sibilant fricatives /s/ and /ʃ/. Tsukasa Yoshinaga (Graduate School of Eng. Sci., Osaka Univ., 1-3 Machikaneyama, Toyonaka, Osaka 560-8531, Japan, t.yoshinaga@biomech.me.es.osaka-u.ac.jp), Kazunori Nozaki (Div. of Medical Information, Osaka Univ. Dental Hospital, Suita, Osaka, Japan), and Shigeo Wada (Graduate School of Eng. Sci., Osaka Univ., Toyonaka, Osaka, Japan)

In this study, a simplified vocal tract model which reproduces both sibilant fricatives /s/ and /ʃ/ is proposed to investigate the effect of tongue position on flow and sound generation in the vocal tract. The simplified model consists of a tongue with sibilant groove and upper and lower teeth in a rectangular channel. Dimensions of the model were determined based on CT images of the subject pronouncing sibilant /s/. The sound and flow fluctuation generated by the model were measured with a microphone and a hot-wire anemometer, respectively. In addition, the large-eddy simulation was conducted to reveal the characteristics of sound source generated by the turbulent flow in the vocal tract. The acoustic measurement showed that the backward shift of the tongue increased the spectral slope value, indicating that the acoustic characteristics were changed from /s/ to /ʃ/. The large-eddy simulation showed that the position of sound source downstream of teeth was shifted backward and the magnitude of the sound source was decreased by shifting the tongue backward. These results indicate that both the resonance characteristics and the position of the sound source were changed by shifting the tongue which causes the difference in acoustic characteristics between /s/ and /ʃ/.

3pSC45. Tongue root positioning in English voiced obstruents: Effects of manner and vowel context. Suzy Ahn and Lisa Davidson (NYU Dept. of Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, suzy.ahn@nyu.edu)

Previous research on utterance-initial voiced stops in American English (AE) has shown that speakers enlarge the vocal cavity via tongue root advancement whether or not the stop is phonated when compared to voiceless stops (Ahn 2015). The current ultrasound study expands this line of research to examine two further variables: (a) utterance-initial fricatives/affricates and (b) effect of frontness of the following vowel. Participants included both monolingual (n = 4) and simultaneous bilingual (n = 7) AE speakers. Phrase-initial stops (/p,b,t,d,k,g/), fricatives (/f,v,s,z/), and affricates (/tʃ,ʤ/) were followed by /e/ or /u/. Most productions confirmed that both phonated and unphonated voiced stops/affricates had more tongue root advancement than voiceless ones, but a small proportion showed no difference between phonated, unphonated, and voiceless stops/affricates. Fricative productions were divided between no tongue root difference due to either voicing or phonation, and greater advancement for voiced fricatives regardless of phonation. Tongue root advancement may be less prevalent for fricatives because weakening of the frication can also facilitate the conditions for phonation. A prediction that tongue root advancement for voicing before /e/ might be limited by the positioning requirements of the front vowel was weakly supported, especially for fricatives. The potential effects of bilingual speakers are also addressed.

3pSC46. An ultrasound study of tongue position during Hindi laryngeal contrasts. Suzy Ahn (Dept. of Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, suzy.ahn@nyu.edu)

Initiating or maintaining phonation during stop closure involves several adjustments, including tongue root advancement to enlarge oral cavity volume (Westbury 1983). Ultrasound imaging shows that in English, the tongue is more advanced for phonologically voiced stops, whether phonated or unphonated (Ahn 2015). The current study uses ultrasound to examine tongue positioning during Hindi stops. Hindi has a unique four-way laryngeal contrast: voiced, murmured, voiceless unaspirated, and voiceless aspirated. Eight native Hindi speakers recorded phrase-initial stops at three places of articulation (labial, dental, velar) followed by the low vowel /a/. Results show a clear distinction in tongue position between voiced and voiceless unaspirated/aspirated stops. The tongue root is advanced for voiced stops in comparison to voiceless stops. However, there is no difference in tongue root position between voiceless unaspirated and aspirated stops. Murmured stops showed variation among speakers in comparison to other stops, while the majority of speakers show more advanced tongue root compared to voiceless stops. The results suggest that tongue root advancement facilitates phonation in Hindi. Thus, in Hindi, tongue root position corresponds to a phonological distinction in phonation most notably between voiced unaspirated stops and voiceless stops, whereas in English, tongue root position reflects a more abstract phonological distinction in voicing that does not correspond to phonation.

3pSC47. Articulatory and acoustic realization of French and German /R/. Cedric Gendrot, Barbara Kuhnert, and Didier Demolin (Linguist, Université Sorbonne Nouvelle, Lab. of Phonet. and Phonology, 19 rue des bernardins, Paris 75005, France, cgendrot@univ-paris3.fr)

French uvular /ʁ/ is usually considered as problematic due to its variability, especially in positions such as word initial and word final. In this study, an articulatory (EMA) and aerodynamic experiment allowed us to determine its major axes of variation. We show that the degree of constriction between the tongue and the palate is related to the voicing of the consonant and we validate the use of a harmonic-to-noise ratio for the measurement of variation of /ʁ/. Aerodynamic data also show that the variation of Nasal Airflow is not significant but a ratio between Oral Airflow and Intra-Oral Pressure significantly varies according to the different realizations of /ʁ/. Subglottic Pressure only plays a significant role in specific cases. These results allow us to propose a new acoustic measurement for the variation in uvular /ʁ/ production. An acoustic study is then presented on large corpora of continuous speech, so as to test the variability of French /ʁ/ in terms of the aforementioned results, and we show the importance of prosodic factors. Finally, a parallel with a German similar corpus of articulatory and acoustic data is drawn, we suggest that German /ʁ/ shows less variation than French /ʁ/ due to phonological reasons.

3pSC48. Articulatory setting as global coarticulation: Simulation, acoustics, and perception. Bryan Gick, Chenhao Chiu (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T1Z4, Canada, gick@mail.ubc.ca), Francois Roewer-Despres (Comput. Sci., Univ. of SK, Saskatoon, SK, Canada), Murray Schellenberg (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada), and Ian Stavness (Comput. Sci., Univ. of SK, Saskatoon, SK, Canada)

Articulatory settings, language-specific default postures of the speech articulators, have been difficult to distinguish from segmental speech content [see Gick *et al.* 2004, *Phonetica* 61, 220-233]. The simplest construal of articulatory setting is as a constantly maintained set of tonic muscle activations that coarticulates globally with all segmental content. In his early Overlapping Innervation Wave theory, Joos [1948, *Language* Monogr. 23] postulated that all coarticulation can be understood as simple overlap, or superposition [Bizzi *et al.* 1991, *Science* 253, 287-291], of muscle activation patterns. The present paper describes an implementation of Joos' proposals within a modular neuromuscular framework [see Gick & Stavness 2013, *Front. Psych.* 4, 977]. Results of a simulation and perception study will be reported in which muscle activations corresponding to English-like and French-like articulatory settings are simulated and superposed on activations for language-neutral vowels using the ArtiSynth biomechanical modeling

toolset (www.artisynth.org). Simulated visible and acoustic outputs presented to perceivers familiar with both languages speak to the question of whether overlapping muscle activations generate outputs that look and sound language-appropriate to perceivers, testing a unified, context-independent model for both coarticulation and articulatory setting. [Research funded by NIH Grant DC-02717 and NSERC.]

3pSC49. Articulatory settings of English-French bilinguals reanalyzed by SS-ANOVA. Yuki Iguro, Ian Wilson, and Julian Villegas (Univ. of Aizu, Tsuruga, Ikki-machi, Aizuwakamatsu, Fukushima 965-8580, Japan, s1200249@gmail.com)

To improve the skill of speaking a second language (L2), one good way may be to be aware of the underlying tongue position for a language. We focused on such underlying position differences between English and French, particularly when pausing for a short time between speaking; something called inter-speech posture (ISP). In past research, Wilson and Gick investigated ISP between English and French spoken by bilinguals. In that research, bilinguals had distinct articulatory settings for each language, mostly in the lips. However, their tongue data was for only 4 points of articulatory settings: distance from the ultrasound probe to tongue root, tongue dorsum, tongue body, and tongue tip, but not overall shape. Furthermore, to measure tongue tip position, past research relied on the alveolar ridge, which is unclear to see: possibly making the results inaccurate for tongue tip. In this study, we analyzed the whole shape of the tongue and made models of them using SS-ANOVA in R so that we could compare the difference from past research using a different measurement method. Our results showed that bilinguals who are perceived as native in both languages have a different ISP in the posterior half of the tongue.

3pSC50. F3 variability in allophones of /l/: Acoustic-articulatory relations. Anisia Popescu (Univ. Paris Diderot, 54 Ave. d'Italie, Paris 75013, France, anisiapopescu358@gmail.com)

F3 is know to exhibit higher values in darker varieties of /l/. This contrast has been attributed to differences in closure fronting and front cavity configuration. In this paper we propose a possible alternative explanation for higher F3 in dark /l/, based on sensitivity functions. Both allophones have an apical gesture, but differ in their tongue dorsum gestures: clear /l/ has a raising and fronting gesture, while dark /l/ has postdorsum retraction towards the uvular region. We propose that the gestural composition of dark /l/ creates two constriction sites in the vocal tract that correspond to the two maxima of the F3 sensitivity function, thus resulting in higher values for dark /l/. To address these predictions we analyzed both acoustic and articulatory data from 6 speakers of the Wisconsin XRMB database. Articulatory strategies vary both inter and intra-speakers. High F3 values are obtained with both elevated tongue tip and retracted postdorsum, while low F3 values, have small values of postdorsum retraction. Overall results show that, despite inter-speaker variability in vocal tract shape and articulatory strategies, F3 is positively correlated with tongue dorsum retraction.

3pSC51. Gestural coordination and blending among liquid consonants and vowels in American English. Rachel Walker (Dept. of Linguist, Univ. of Southern California, GFS 301, 3601 Watt Way, USC, Los Angeles, CA 90089-1693, rwalker@usc.edu) and Michael Proctor (Dept. of Linguist, Macquarie Univ., Sydney, NSW, Australia)

In General American English (GAE), tense/lax vowel contrasts occur before syllables ending in /l/ (e.g., *peel, pill*), but not those ending in /ɹ/ (e.g., *peer*; no [i]/[ɪ] contrast). No such restrictions occur after a syllable-initial liquid consonant (e.g., *lip*, *leap*, *rip*, *reap*). This study investigates the hypothesis that these phonotactic asymmetries arise from the interaction of differences in gestural coordination between nuclear vowels and liquid consonants in syllable onset versus coda position (Browman & Goldstein 1995) and differences in blending parameter strength for GAE laterals versus rhotics. A study of speech production by four GAE speakers was conducted using real-time Magnetic Resonance Imaging (Narayanan *et al*. 2004). Liquid consonants were produced in various vocalic environments and syllable positions (syllable-final, syllable-initial: word-initial and intervocalic). Analysis using a center of gravity metric defined by midsagittal lingual outlines points to greater articulatory stability in rhotics versus laterals across different vocalic contexts and less coarticulatory displacement of pre-lateral vowels than pre-rhotic vowels. Further analysis of spatial and temporal effects in the data will be used to test hypotheses about coordination of vowel and liquid consonant gestures in different syllable positions and their relation to phonotactic patterns. [Research supported by NIH and ARC.]

3pSC52. Revisiting articulatory positions of Japanese vowels as a function of duration on the basis of analysis of large-scale speech corpora. Takayuki Kagomiya (Ctr. for Frontier Medical Eng., Chiba Univ., 1-33, Yayoicho, Chiba, Chiba 263-8522, Japan, kagomiya@chiba-u.jp)

The purpose of this study is to illustrate certain features of the place of articulation of Japanese vowels on the basis of formant frequency. For this purpose, two types of relationship between vowel duration and formant frequency were examined. The first of these relationships was the correlation between speaking rate and formant frequency. Speech materials are obtained from two large-scale speech corpora: Electrotechnical Laboratory Spoken Word database (ETL-WD) and The corpus of spontaneous Japanese (CSJ). The results of analyses showed that as speaking rate slows, articulations get more distinct: open vowels increase in openness, close vowels increase in closeness, front vowels increase in frontness, and back vowels increase in backness. The second relationship was between formant frequencies and vowel length. The result of this analysis showed that place of articulation is more distinct in long vowels than in short vowels. The exception was the back vowel /u/, all of which did not show greater backness in long vowels. This result supports previous arguments that Japanese /u/ is not a typical back vowel.

3pSC53. Articulatory movement in consonant clusters: A contrastive study of Japanese and English. Seiya Funatsu (Prefectural Univ. of Hiroshima, 1-1-71 Ujinahigashi Minami-ku, Hiroshima 734-8558, Japan, funatsu@pu-hiroshima.ac.jp) and Masako Fujimoto (Waseda Univ., Tokyo, Japan)

We investigated the articulation of consonant clusters with Japanese and English speakers using the WAVE system (NDI Corp.). Until now we measured the displacement of tongue tip only, but in this study we also measured the displacement of the mandible. The subjects were three Japanese and two English speakers. The speech samples were 4 words: blat, bnat, plat, pnat. All these second consonants, /l, n/ are articulated with tongue tip placed behind alveolar ridge. These words were embedded in a carrier sentence, "Say X.," and these sentences were uttered 5 times each in random order. The articulation time from first consonants, /b/, /p/, to second consonants, /l/, /n/, were measured. Specifically, the time from the highest point of the mandible to that of the tongue tip was measured. Overall average time was 131.2 ms (Japanese) and 48.1 ms (English). Two-way ANOVA (Language vs. Cluster) showed a significant main effect in Language ($p<0.0001$). This result suggested that for English speakers, the first and second consonants are strongly co-articulated, while in Japanese speakers, they are not co-articulated or weakly co-articulated. The measurement of mandibular displacement more clearly showed the difference of the strength of co-articulation between Japanese and English.

3pSC54. Co-articulation of unreleased final stops to preceding vowels in Cantonese: An ultrasound study. Suki Yiu (Linguist, Univ. of Hong Kong, School of Humanities (Linguistics), University of Hong Kong, Pokfulam Rd., Hong Kong, Hong Kong, syutji@hku.hk)

Unreleased final stops are more susceptible to reduction than their released counterparts, thus co-articulatory information on the preceding vowel is important to signal place contrasts of post-vocalic stops. This ultrasound study examines the gestural co-ordination of a VC sequence of monosyllabic words in Cantonese, a language which has preserved phonemic contrasts of six unreleased final stops /k, t, p, ng, n, m/ to form a VC sequence with three vowels /a, ɐ, ɔ/. We expect properties of /t, k/ to be (i) only immediately present at the vowel offsets (i.e. minimal co-articulation), or (ii) present from vowel midpoints to offsets (i.e. gradual co-articulation). If the /t, k/ contrast is (iii) consistently present throughout the vowel from

onset to offset, it indicates phonologization of the /t,k/ contrast from the stops to the preceding vowels. Preliminary SS ANOVA results show that the lingual contours at vowel midpoints have early lingual raising into final stop constrictions, and the vowel offsets look more like the upcoming stops than the preceding vowels. This agrees with Khouw & Ciocca's 2006 acoustic study, that unreleased final stops use C-to-V co-articulation to strengthen the place contrast.

3pSC55. Source properties of dorsal fricatives. Charles Redmon and Allard Jongman (Linguist, Univ. of Kansas, 1541 Lilac Ln., Rm. 427, Lawrence, KS 66046, redmon@ku.edu)

Present models of voiceless fricative acoustics assume excitation of the vocal cavity anterior to a narrow constriction by an aperiodic source generated from turbulence within that constriction (or in the case of dipole sources from interaction of the jet with an obstacle downstream). This model has been applied successfully to anterior places of articulation (labial to palatal) but for dorsal fricatives, i.e., velar /x/ and uvular /χ/, there is evidence to suggest the assumption of a line source may not hold. Rather, preliminary data from our lab and from studies such as Zeroual (ICPhS, 2003) and Shosted & Chikovani (JIPA, 2006) indicates intermittent presence of a mixed source due to passive vibration of the uvula, yet no study to date has directly addressed this element of the sound source in dorsal fricative production. Acoustic and aerodynamic data on dorsal fricatives produced by four speakers each of Arabic, Persian, and Spanish are presented to validate the nature of the dorsal fricative source component and the degree of cross-linguistic variation in production of these sounds. Measures of periodicity duration and cycle amplitude/shape are employed to estimate the oscillatory pattern of the uvula and model its contribution to the radiated acoustic signal.

3pSC56. Investigating patterns of vowel production using real-time magnetic resonance imaging. Michael I. Proctor (Linguist, Macquarie Univ., 16 University Ave., NSW 2019, Australia, michael.proctor@mq.edu.au)

Characteristic tongue postures of 21,478 vowels produced by four speakers of General American English were compared in lexically stressed and unstressed positions to examine patterns of articulation and reduction. Image frames capturing midsagittal vocal tract configurations were reconstructed at the acoustic midpoint of each vowel produced in a 3,450 word forced-aligned transcribed speech corpus (Narayanan *et al*. 2014). Sixty image sets were constructed, consisting of all exemplars of each contrastive vowel quality of American English, as produced by each speaker: up to 1049 tokens of the most frequent vowel /i/, and 48 tokens of the least frequent vowel /ɔɪ/. Image frames were cropped to a region of interest centred on the tongue dorsum, and linearized into one-dimensional vectors in which pixel intensity corresponds to presence of soft tissue. Principle components analysis was applied to each set of image vectors to examine the distribution and variability of tongue tissue associated with each vowel posture. "Eigentongues" were constructed from the most significant components in each image set (Sirovich & Kirby 1987; Hueber *et al*. 2007), resulting in a set of images characterizing each speakers' vowel space, produced in both lexically stressed and unstressed positions.

3pSC57. Physiological features of geminate depending on the speaking rate. Ken-Ichi Sakakibara (Health Sci. Univ. Hokkaido, 2jo-5chome, Ainosato, Kita-ku, Sapporo 002-8072, Japan, kis@hoku-iryo-u.ac.jp), Kimiko Yamakawa (Shokei Univ., Kumamoto, Japan), Hiroshi Imagawa, Takao Goto (Univ. Tokyo, Tokyo, Japan), Akihito Yamauchi (National Ctr. for Global Health and Medicine, Tokyo, Japan), Katsuhiro Maki, and Shigeaki Amano (Aichi Shukutoku Univ., Nagakute, Japan)

In this study, we examined the physiological and acoustic characteristics of plosive- and fricative-type geminates, three each of minimal pairs for the plosive- and fricative-type geminates were recorded in different speaking rates, with there ranges are from 3 to 13 mora /s. Physiological analysis showed that (1) an opened duration of vocal folds depends on a speaking rate; (2) a closing duration of vocal folds depends on a speaking rate but an opening duration does not. Acoustical analysis showed that, when a speaking rate increases, (1) logarithm of closure and fricative durations linearly decrease; and (2) a ratio of the duration of a singleton to a geminate decreases.

3pSC58. A preliminary acoustic analysis of three-dimensional shape of the human nasal cavity and paranasal sinuses extracted from cone-beam CT. Tatsuya Kitamura (Faculty of Intelligence and Informatics, Konan Univ., 8-9-1, Okamoto, Higashinada, Kobe, Hyogo 658-8501, Japan, t-kitamu@konan-u.ac.jp), Hironori Takemoto (Chiba Inst. of Technol., Narashino, Chiba, Japan), Hisanori Makinae (National Res. Inst. of Police Sci., Kashiwa, Chiba, Japan), Tetsutaro Yamaguchi, and Kotaro Maki (Showa Univ., Shinagawa, Tokyo, Japan)

The shape of the nasal cavity and paranasal sinuses is complex and varies between individuals. Because the shape is almost stable during speech, the acoustic properties could constantly provide speaker specific information to speech sounds, that is, speaker individuality. In this preliminary analysis, the shape was extracted from cone-beam CT data for a subject using a machine learning technique and its acoustic properties were examined using finite-difference time-domain simulation. The transfer function from the glottis to the nostrils was calculated and the distribution pattern of the pressure anti-nodes was visualized at frequencies of major spectral peaks and dips. In addition, transfer functions were calculated when each of the paranasal sinuses other than the ethmoidal ones was occluded to identify which sinus caused which dip. As a result, the longitudinal resonance in the right or left half of the nasal cavity generated each peak, while the transverse resonance in the pharyngeal cavity caused the major dips. The right maxillary, sphenoidal, and frontal sinuses generated dips below 1 kHz. The left maxillary sinus and ethmoidal sinuses, however, contributed to generating not only dips but also peaks. [This research was partly supported by JSPS KAKENHI Grant Numbers 15K00263, 25280066, and 25240026.]

3pSC59. Modeling the effect of palate shape on the articulatory-acoustics mapping. Sarah Bakst and Keith Johnson (Linguist, Univ. of California Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, bakst@berkeley.edu)

Previous research [Brunner *et al*. (2009, JASA 125(6), 3936-3949) and Bakst & Lin (ICPhS 2015)] shows that articulatory variability is reduced for people with flatter palates. Brunner *et al*. (2009) hypothesized that this is because the mapping between articulation and acoustics is more linear for flatter than for more domed palates; in order to maintain similar levels of acoustic consistency, people with flatter palates must be more articulatorily precise than people with more domed palates. A revised version of the Maeda synthesizer was used to model how vocal tract anatomy influences the mapping of articulation onto acoustics, using American English /r/ as a test case. A retroflex-able tongue tip was added to the synthesizer. Two additional palate shapes were also added to the synthesizer, and a script was used to search the articulatory-acoustic space for vocal tract configurations that resulted in a low F3 (the hallmark acoustic cue for /r/) for each palate. Palate shape influences not only the overall sensitivity of the articulatory-acoustic mapping, but also the effect of each individual articulatory parameter on F3.

3pSC60. Acoustic effects of phonetic conditions and laryngeal specification on phonation in English voiceless obstruents. Lisa Davidson (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, lisa.davidson@nyu.edu)

A detailed acoustic analysis of phonation in voiceless obstruents in American English (AE) is presented to investigate potential acoustic outcomes of the laryngeal coordination relationships that have been reported in the literature. The current study examines the realization of phonation in voiceless obstruents in a corpus of read speech (37 AE speakers). Similar to phonation in voiced obstruents (Davidson 2016), linguistic factors such as phrase and word position, stress, and the preceding phoneme condition the presence and degree of phonation during the constriction period of stops and fricatives. The amount of phonation present is further analyzed by characterizing where in the constriction interval phonation appears. Carryover

phonation (or "bleed") from a preceding sonorant is most common for stops, while a "trough" pattern (phonation that dies out and then begins again before the end of the closure) is more prevalent for fricatives. The linguistic conditioning factors, and the overall lower rate of partial voicing and proportion of phonation in the constriction as compared to voiced obstruents, illustrate the acoustic and aerodynamic consequences of the various laryngeal coordination patterns that have been proposed for English voiceless obstruents in different prosodic environments.

3pSC61. Visuomotor pursuit tracking accuracy for intraoral tongue movement. Vedad Fazel and William F. Katz (Commun. Sci. & Disord., The Univ. of Texas at Dallas, The University of Texas at Dallas/Callier Ctr., 1966 Inwood Rd., Dallas, TX 75235, vxf130030@utdallas.edu)

Visuomotor pursuit tracking (VMT) tasks are used to study the movement of the articulators in order to better understand the processes involved in speech motor planning and execution. Because most of this research has focused on lip/jaw tracking of sinusoids visually presented on a screen, little is known about the tracking capabilities of the tongue, or for visual targets placed in the oral cavity. The present study used a novel technique for measuring tongue (and jaw) VMT based on a 3D electromagnetic articulography (EMA) system. Streaming EMA data from oral sensors were used to construct a real-time avatar of the tongue which was shown to subjects on a computer monitor. Subjects also viewed a (virtual) intra-oral spherical target that they were required to "hit" with the tongue tip sensor. The target was programmed to move in a sinusoidal direction and was varied in frequency (Hz), direction, and predictability. Preliminary results suggest that tracking accuracy is inversely related to target frequency and is higher for vertical and horizontal motion than for lateral motion. The role of target predictability will also be described.

3pSC62. Performance and function meet structure: A white matter brain connection tuned for motor speech control. John J. Sidtis (Nathan Kline Inst., 6 Prairie Ave., Suffern, NY 10901, john.sidtis@nyu.edu), Muhammad A. Mubeen, Ali Asaei, Babak Ardekani (Nathan Kline Inst., Orangeburg, NY), and Diana Sidtis (New York Univ., Orangeburg, NY)

Despite bilateral brain activation during speech tasks, we have shown that performance is predicted by a blood flow increase in the left inferior frontal region and decrease in the right caudate, consistent with classic lesion studies. This study characterized the structural connections between these brain areas using diffusion tensor imaging and examined their relationships with measures of motor speech control. Probabilistic tractography estimated the connection strength between these two brain structures, both ipsilaterally and contralaterally, in 25 normal subjects. Speech was recorded at a separate evaluation. The majority of fiber connections were ipsilateral, but contralateral connections were also present. The relative connection strength between the right caudate and the left inferior frontal region was significantly associated with acoustic measures of stability for frequency (correlations for the repetition of /pa/ = 0.46; /ta/ = 0.5; /ka/ = 0.46; /pataka/ = 0.52) and amplitude (/pa/ = 0.43; /ta/ = 0.5; /ka/ = 0.49; /pataka/ = 0.49), echoing the predictive value of blood flow in these regions during speech. This was not observed for other connections. These results suggest that white matter connections share functional specialization with the structures they connect in the motor-speech system.

3pSC63. On the link between the ability to reproduce rhythm and reading speed: Effects of visual grouping marks. Antonin Rossier-Bisaillon and Victor Boucher (Linguistique et traduction, Université de Montréal, 3150, rue Jean-Brillant, Montréal, QC H3T 1N8, Canada, antonin.rossier-bisaillon@umontreal.ca)

Studies have revealed a link between a deficient ability to reproduce rhythm and dyslexia. Moreover, a normal ability to reproduce rhythm has been shown to correlate with reading speed. Some associate these findings to effects of neural oscillations and a visual parsing of text input (Vidyasagar, 2013). The present study aimed to clarify the properties of the visual stimuli that support a correlation between rhythm reproduction and reading speed. The experiments were partly based on Tierney and Kraus (2014). Thirty participants were asked to reproduce heard regular (2 beats/sec) and irregular (1-3 beats/sec) rhythms by tapping on a keyboard. Then, the participants had to read out loud, at a fast rate, visually displayed sequences of words and non-words. The sequences contained either no spaces between items (baseline condition) or spaces marking regular and irregular groups. Mean reading speeds were calculated on accurately decoded sequences. Among the significant findings, strong correlations were observed between mean reading speeds of texts containing spaces and the reproduction of certain rhythm patterns. However, no significant correlations appeared for text where there were no spaces separating words and non-words. This supports the view that the ability to reproduce rhythm can link to a visual parsing.

3pSC64. Post-focus compression: All or nothing? Ting Huang (Dept. of Linguist & Philosophy, Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, funting.huang@gmail.com) and Feng-fan Hsieh (Graduate Inst. of Linguist, National Tsing Hua Univ., Hsinchu, Taiwan)

Post-focus compression (PFC) has been claimed to be a "hard-to-evolve," inherent prosodic feature that may have a single historical origin (Proto-Nostratic). This study explored the distribution of PFC in two sub-dialects of Malaysian Hokkien (Southern Min Chinese), i.e., Penang Hokkien (PH) and Melaka Hokkien (MH), using novel experimental designs and statistical techniques (SS ANOVA). Specifically, in addition to the conventional nasal onset-only contexts, stop and fricative onset-only contexts were taken into consideration. We analyzed F0 and duration of the constituents in question under three focus conditions (Initial vs. Medial vs. Final focus in the Subject-Verb-Object frame). The results are (a) the presence of PFC may be dependent on different segmental contexts: in MH, PFC is attested in the stop onset-only environments but absent elsewhere and (b) the absence of PFC "induces" Post-focus shortening in the nasal and fricative-only contexts in MH and in all data in PH. The findings suggest that PFC may not be an "inherent" feature because MH and PH are closely related sub-dialects and both have had language contact with the same language, Malay. Finally, global F0 raising is found for the first time in Final focus (Object) in both MH and PH.

3pSC65. Tone sequences in lexical processing of Beijing Mandarin. Isabelle Lin (Dept. of Linguist, Univ. of California Los Angeles, 3125 Campbell Hall, UCLA, Los Angeles, CA 90095-1543, isabellelin@ucla.edu)

Previous research on the role of tone information in lexical access in Mandarin Chinese has focused on monosyllabic words. Using corpus data, we show that tone is more informative on Mandarin disyllables than monosyllables. Next, we investigate the role of tone sequences in disambiguating two segmentally identical disyllabic word candidates that differed only in tone. Using a priming paradigm, we show that tone frequency biases lexical decisions. Listeners were presented with a disyllabic sequence that is tonally ambiguous between two lexical entries. When word frequency and tone frequency did not favour the same candidate, a tonally-matched prime reduced the likelihood of picking the candidate with the matching tone. Preliminary results suggest that listeners are sensitive to the overall likelihood of encountering a given sequence of tones in running speech, instead of just tone sequences within disyllabic words.

3pSC66. Surface phonetic or underlying phonological representations: A mismatch negativity study of Mandarin tone assimilation. Yu-Fu Chien (Linguist, Shanghai Int. Studies Univ., Number 550, DaLian West Rd., HongKou District, Shanghai, China, y078c178@ku.edu), Robert Fiorentino, Xiao Yang, and Joan A. Sereno (Linguist, The Univ. of Kansas, Lawrence, KS)

Phonological alternation, in which a sound changes depending on its phonological environment, poses challenges to spoken word recognition models. Mandarin T3 sandhi is such a phenomenon in which a tone 3 changes into a tone 2 when followed by another T3, which raises questions regarding whether the human brain processes the surface acoustic-phonetic representation or the underlying linguistic representation of Mandarin T3 sandhi words. We conducted a mismatch negativity (MMN) study examining this issue. Participants passively listened to a T2 word [tʂu2 je4] /tʂu2 je4/, a T3 word [tʂu3 ʐən4] /tʂu3 ʐən4/, a Sandhi word [tʂu2 jen3] /tʂu3

jen3/, or a mix of T3 and sandhi words in an odd-ball paradigm. All were interspersed with a T2 word [tʂu2] (deviant). Results showed an MMN only in the T2 and T3 condition but not in the Sandhi or Mix condition. The fact that the surface acoustic information in the T2 and Sandhi conditions was identical, yet yielding disparate MMN results suggests that Mandarin speakers process T2 and Sandhi words differently. These data provide evidence for a stored linguistic representation, either an underspecified or underlying tone 3 syllable, which must be operative when listeners map the signal during spoken word processing.

3pSC67. Effects of aero-tactile stimuli on continuous speech perception.
Donald Derrick, Greg A. O'beirne, Tom De Rybel, Jennifer Hay, and Romain Fiasson (NZILBB, Univ. of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, donald.derrick@canterbury.ac.nz)

We follow up on research demonstrating that aerotactile information can enhance accurate identification of stop- and fricative-onset syllables in two-way forced-choice experiments (Derrick, *et al*., 2014) to include open-set identification tasks. We recorded audio and speech airflow simultaneously from the lips of two New Zealand English (NZE) speakers (one female, one male), and used these recordings to produce an auditory/aero-tactile matrix sentence test. The airflow signal is used to drive a piezoelectric air pump that delivers airflow to the right temple simultaneously with presentation of noise-degraded auditory recordings. Participants (including native NZE speakers with and without hearing impairment, and normal-hearing native non-NZE and non-native English speakers) listen to and repeat 5-word sentences presented in noise with and without simultaneous airflow. Their open-set responses are scored by the researchers. Custom-written software identifies the SNRs for 20% and 80% word identification accuracy using a dual-track adaptive algorithm, and these data are fitted to psychometric curves relating SNR to speech intelligibility. Psychometric curves for airflow and no-airflow conditions will be compared in order to identify the influence of airflow on continuous speech perception. Data collection is in progress, and the results will be presented at the ASA conference.

3pSC68. Just noticeable differences for pitch height and pitch contour for Chinese and American listeners.
Allard Jongman, Zhen Qin, Jie Zhang, and Joan Sereno (Linguist, Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, jongman@ku.edu)

Previous studies suggest that Chinese listeners may be more sensitive to pitch direction while American listeners primarily attend to pitch height. The present study sought to establish JNDs for pitch discrimination using a three-interval, forced-choice procedure with a two-down, one-up staircase design (cf. Liu, JASA 2013). We manipulated a high rising and a high falling Mandarin tone produced by a female speaker. Specifically, for pitch height, a standard contour rising from 200 Hz at tone onset to 230 Hz at tone offset was paired with deviants that were parallel to the standard but started (and ended) at lower and higher f0 values (onsets ranging from 185 to 215 Hz in 1 Hz steps). Likewise, a standard contour falling from 200 Hz at tone onset to 170 Hz at tone offset was paired with parallel deviants whose onsets ranged from 215 to 185 Hz. For pitch direction, standard rising and falling contours were paired with deviants with either steeper or shallower slopes, again varying in 1 Hz steps. Results indicate that, overall, Chinese listeners are more sensitive to pitch direction and American listeners to pitch height. However, these effects are modulated by both the direction (falling/rising) and steepness of the contours.

3pSC69. The articulation of /ɹ/ in New Zealand English.
Matthias Heyne, Xuan Wang, Kieran Dorreen, Donald Derrick, and Kevin Watson (NZILBB, Univ. of Canterbury, Private Bag 4800, Christchurch, 8140, New Zealand, donald.derrick@canterbury.ac.nz)

A large number of studies have investigated the articulation of approximant /ɹ/ in American English (AE) (e.g., Delattre & Freeman, 1968). This research has found that a low third formant (F3), the main acoustic cue signaling rhoticity, can be achieved using many different tongue configurations; the two main tongue shapes used for /ɹ/ are "tip-down" ("bunched") and "tip-up" ("retroflex") (cf. Hagiwara, 1994). While speakers likely employ various "trading relationships" to maintain a constantly low F3 across production strategies (Guenther *et al*., 1999), they have access to a pool of variation, which some use to form complex and idiosyncratic patterns of allophony (Mielke *et al*., 2016). Such patterns may arise during speech acquisition (Magloughlin, 2016). This study focuses on a non-rhotic dialect, New Zealand English (NZE), to test whether dialect rhoticity constrains idiosyncratic allophony. Ultrasound video was collected for 63 speakers articulating 13 words containing tokens of /ɹ/ in different phonetic environments. Analysis aims to determine whether NZE speakers utilize the same tongue gestures as seen in AE, and whether they display similar patterns of allophonic variation. The data include productions from 12 children (under 10) and 13 youth (11-18), allowing examination of /ɹ/ during childhood development.

3pSC70. Spatial congruence in multimodal speech perception.
Megan Keough, Murray Schellenberg, and Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, mkeough@alumni.ubc.ca)

A growing literature provides evidence for the importance of synchronicity of cross-modal information in speech perception [e.g., audio-visual, Munhall *et al*. 1996, *Perception & Psychophysics 58:* 351-362; audio-aerotactile, Gick *et al*. 2010, *JASA 128*: 342-346; visual-aerotactile, Bicevskis *et al*. submitted ms]. While considerable work has investigated this role of temporal congruence, no research has directly explored the role of spatial congruence (i.e., co-directionality) of stimulus sources. If perceivers are picking up a localized distal speech event [e.g., Fowler 1986, *Status Report of Speech Research*: 139-169] cross-modal sources of information are predicted to be more likely to integrate when presented codirectionally than contradirectionally. An audio-aerotactile pairing lends itself well to this question as both modalities can easily be presented laterally. The current study draws on methodology from previous work [Gick & Derrick 2009, *Nature 462*: 502-504] to ask whether cross-modal integration persists when cross-modal cues are spatially incongruent. Native English perceivers were presented with syllables contrasting in aspiration and embedded in noise, with some tokens accompanied by inaudible air puffs applied to the neck; aerotactile source locations either matched or opposed the spatial direction of the acoustic signal. Implications of results for multimodal integration theories will be discussed. [Funded by NIH Grant DC-02717 and NSERC.]

3pSC71. Cross-modal association between auditory and visual-spatial information in Mandarin tone perception.
Beverly Hannah, Yue Wang (Linguist, Simon Fraser Univ., 9213 Robert C. Brown Bldg., 8888 University Dr., Burnaby, BC, Canada, yuew@sfu.ca), Allard Jongman, and Joan A. Sereno (Linguist, Univ. of Kansas, Lawrence, KS)

Speech perception involves multiple input modalities. Research has indicated that perceivers may establish a cross-modal association between auditory and visual-spatial events to aid perception. Such intermodal relations can be particularly beneficial for non-native perceivers who need additional resources to process challenging new sounds. This study examines how co-speech hand gestures mimicking pitch contours in space affect non-native Mandarin tone perception. Native English as well as Mandarin perceivers identified tones with either congruent or incongruent auditory-facial and gestural (AF/G) input. Perceivers also identified congruent and incongruent auditory-facial (A/F) stimuli. Native Mandarin results showed the expected ceiling-level performance in the congruent A/F and AF/G conditions. In the incongruent conditions, while A/F identification was primarily auditory-based, AF/G identification was partially based on gestures, demonstrating the use of gestures as valid cues in tone identification. The English perceivers' performance was poor in the congruent A/F condition, but improved significantly in AF/G. While the incongruent A/F identification showed some reliance on facial information, incongruent AF/G identification relied more on gestural than auditory-facial information. These results indicate positive effects of facial and especially gestural input on non-native tone perception, suggesting that cross-modal (visual-spatial) resources can be recruited to aid auditory perception when phonetic demands are high.

3pSC72. Worse than the dentist: Effects of simultaneous acoustic and somatosensory feedback degradation on speech. Elliot A. Pollack, Mary C. Rowley, and Elizabeth D. Casserly (Psych., Trinity College, 300 Summit St., Life Sci. Ctr., Hartford, CT 06106, elizabeth.casserly@trincoll.edu)

Speakers use their sensory feedback to monitor real-time speech production; when unexpected changes to that feedback occur, they alter their articulation in immediate, perturbation-specific ways. However, it is not currently known how the feedback from different sensory systems is integrated across contexts. This study varied the reliability of feedback in the auditory and somatosensory domains and examined vowel acoustic phonetics for changes as a result of the manipulation. Acoustic feedback was degraded via eight-channel real-time cochlear implant simulation, while an over-the-counter oral anesthetic (benzocaine) was applied to speakers' tongues and lips to degrade somatosensory feedback. Speakers (N = 18) produced 139 isolated English words under both baseline and feedback-degraded conditions. F1 and F2 measurements were taken from stressed tokens of eight vowels [i, ɪ, ɛ, æ, ɑ, ʌ, ʊ, u]. Significant differences in speakers' responses to feedback degradation were observed both across vowels and across individuals. [u]-fronting was the most consistent response observed, along with isolated classes of vowels (commonly, front or back) being hyper- or hypo-articulated. The variability suggests that the speech motor control system lacks a single, automatic consequence for global feedback degradation.

3pSC73. Plasticity of the internal model for speech sounds: Articulatory changes during intensive visual biofeedback treatment. Jennell C. Vick, Rebecca L. Mental (Psychol. Sci., Case Western Reserve Univ., 11635 Euclid Ave., Cleveland, OH 44106, jennell@case.edu), Holle Carey (Vulintus, Dallas, TX), Nolan T. Schreiber (Elec. Eng. Comput. Sci., Case Western Reserve Univ., Cleveland, OH), Andrew Barnes (Psychol. Sci., Case Western Reserve Univ., Cleveland, OH), and Gregory S. Lee (Elec. Eng. Comput. Sci., Case Western Reserve Univ., Cleveland, OH)

Visual biofeedback, commonly delivered by a display of ultrasound of tongue position, has demonstrated effectiveness for treating residual speech errors in older children and adults. It can be challenging to make kinematic measures during treatment without the use of a head stabilizing system, however, and it is likewise not possible to measure the changes in speech motor control that accompany improvements in speech sound production. Opti-Speech, a visual biofeedback treatment software that uses EMA to provide positional data, has the benefit of providing a steady stream of position data throughout baseline and treatment sessions. In this study, Opti-Speech treatment was provided with an intensive schedule (2x/day for 5 days) to two adolescent males with persistent speech errors (i.e., lateralized /s/). Marked improvements in /s/ accuracy and quality were noted over the course of treatment and are reported in another paper (Mental *et al.*, this meeting). Kinematic measures were made of tongue position throughout the course of the three baseline and 10 treatment sessions, including peak velocity, duration, and measures of token-to-token variability. The covariance of changes in these measures with increasing accuracy and quality of /s/ productions will be reported.

3pSC74. Acoustic modification across speaking modes in English and Korean. Yunjung Kim and Hyunju Chung (Louisiana State Univ., 86 Hatcher Hall, Baton Rouge, LA 70803, ykim6@lsu.edu)

Previous studies have reported temporal and spectral modification of acoustic signals of speech according to varying speaking modes (i.e. clear, loud speech) in several languages (Krause & Braida, 2004; Lam & Tjaden, 2015). Interestingly, some acoustic characteristics such as vowel contrasts, inferred from the size of the corner vowel space, have been discussed to exhibit common, equivalent change between languages despite the different inventory sizes (Smiljanic & Bradlow, 2005). The current study, as an expansion of this line of research, seeks to examine the effect of native language on selected acoustic measures when speakers voluntarily switch speaking modes among conversational, clear, and slow speech. 24 female adults (12 English-speaking, 12 Korean-speaking) were asked to read CVCV structures in the three aforementioned conditions. English and Korean were chosen mainly because of their differences in vowel and stop inventories. Vowel duration, formant structure (F1-F2), and VOT were measured from the target syllable (CVCV), while intensity and F_0 were analyzed for the entire utterance to extend the observations of the non-segmental aspects of speaking modification. Findings are expected to reflect both language-independent and -dependent effects as well as to call for the need to consider the given language in behavioral management of diverse speech disorders.

3pSC75. Fricatives in conversational vs. read speech in mid-Western American English. Daniel Brenner (Univ. of AB, Edmonton, AB, Canada), Viktor Kharlamov (Florida Atlantic Univ., 777 Glades Rd., CU-97, Ste 280, Boca Raton, FL 85721, vkharlamov@fau.edu), and Benjamin V. Tucker (Univ. of AB, Edmonton, AB, Canada)

The present study examines the acoustic characteristics of fricatives across two corpora from mid-Western American English. Our goal is to determine (i) the range of acoustic variation in the production of fricatives in conversational vs. read speech, and (ii) the type and extent of individual variation schemes demonstrated by the speakers. The study surveys over 100,000 fricative tokens from the conversational Buckeye Corpus (Pitt *et al.* 2005) and the read speech TIMIT Corpus (Zue and Seneff 1988). We test an extensive list of acoustic measures compiled on the basis of prior work with laboratory data (incl. duration, center of gravity, skew, kurtosis, peak power, etc.) and explore the implications of the findings for our understanding of the group and individual patterning of speech production processes across the two different speech styles.

3pSC76. Effects of phonetic reduction and regional variation on lexical access. Ellen Dossey and Cynthia G. Clopper (The Ohio State Univ., 108A Ohio Stadium East, 1961 Tuttle Park Pl., Columbus, OH 43210, dossey.1@osu.edu)

This study investigates the impact of acoustic variability on speech processing using a cross-modal priming experiment. Participants heard an audio prime consisting of a single lexical item followed by a printed word on the computer screen and were asked to indicate whether or not the word they heard was the same as the word on the screen. Auditory primes were balanced for linguistic factors that contribute to phonetic reduction, including lexical neighborhood density and whether the token was the first or second mention of the word in the larger context from which it was extracted. Primes were also balanced for social factors which contribute to phonetic variation, including regional dialect. To analyze the effect of these factors on speech processing, reaction times to matching prime-target pairs were analyzed. The results show that reaction times for Midland listeners were significantly faster following Midland primes than Northern primes, suggesting a native dialect benefit. However, the effect of dialect was smaller for words with a lower neighborhood density and for second mention primes relative to first mention primes. Together, these results suggest that dialect variation may have a smaller impact on speech processing in contexts that favor phonetic reduction.

3pSC77. Assessment of areas of the English-Japanese bilingual brain by using functional MRI. Keiko Asano (School of Medicine, Juntendo Univ., 1-11-2905, Kinko-cho, Kanagawa-ku, Yokohama-city 221-0056, Japan, kiasano@uu.em-net.ne.jp), Hidenori Sugano, Hajime Arai, Madoka Nakajima, and Shigeki Aoki (School of Medicine, Juntendo Univ., Tokyo, Japan)

Bilinguals are often determined by how early they are exposed to L2 and the length of their immersion in the L2 environment, both of which affect factors that are important for acquiring language. To analyze the relationship between these, the language function of English-Japanese bilinguals was assessed using fMRI. In order to aim to observe differences in the brain, language function areas were activated by tasks specific to L2 usage. There were 24 English-Japanese bilingual speakers, average age of 25, categorized into four groups according to the age of exposure (before or after age 8), and the length of exposure (longer or shorter than 5years):Early Long (EL), Early Short (ES), Late Long (LL) and Late Short (LS). A PHILIPS Achieva 3.0T TX MRI machine was used to scan the subjects. Tasks consisted of hearing a short story in Japanese (Issunboshi) and English (Peter Rabbit) and their reverse plays by comparison of BOLD oxygenation levels. Focus was given to Broca's area, Wernicke's area and the premotor cortex. This study showed that broad areas of the premotor cortex were activated in ES and LS. EL and LL scans showed localization in Wernicke's area. Since function in Broca's area and the premotor cortex were consistent in ES and LS, it was assumed that these areas are necessary for proficiency and plasticity of language.

3pSC78. Effects of attention on lexically-informed perceptual learning for speech and letter perception. Julia R. Drouin, Nicholas R. Monto, Stephen Graham, Jacqueline Ose, and Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT, nicholas.monto@uconn.edu)

Listeners use lexical information to adjust the mapping to prelexical representations in both the auditory and visual modalities, a phenomenon referred to as lexically-informed perceptual learning (LIPL). Recent research points to attention mechanisms as potential constraints on this type of learning; learning is attenuated when listeners explicitly note anomalies in the speech input, and learning is negatively correlated with performance on tasks of attention-switching control. Here we examine whether graded lexical recruitment as modulated by attention influences LIPL in speech (Experiment 1) and print (Experiment 2). All participants completed a training phase where they were exposed to an ambiguous sound (midway between /s/ and /ʃ/) or an ambiguous grapheme (midway between H and N). Attention was manipulated during exposure by asking participants to make a lexical decision, a syntactic decision, or a decision regarding surface variation (i.e., amplitude, color). All participants then completed a categorization task to assess LIPL. The results to date indicate that LIPL is influenced by attention; the magnitude of the learning effect was not equivalent among the exposure conditions for either modality. However, differences between the two modalities were observed, which are considered with respect to the neural systems that support spoken and written language.

3pSC79. Bi-dialectal homophone effects in Kansai Japanese lexical decision tasks. Karen Tsai, Nicholas Lester, and Fermín M. del Prado Martín (Dept. of Linguist, Univ. of California, Santa Barbara, Santa Barbara, CA 93106, karentsai@umail.ucsb.edu)

Many Japanese varieties, including Tokyo Japanese, have lexical pitch accent patterns that can result in homophones that vary across dialects, such as *ame* HL 'rain (Tokyo)' and *ame* LH 'candy (Tokyo), rain (Kansai)'. Experimental research on lexical pitch accent in Japanese has been limited, however, to Tokyo Japanese and accentless varieties (Utsugi, Koizumi & Mazuka, 2011). This study explores homophone effects of minimal accent pairs for speakers of Kansai Japanese, a dialect that, unlike Tokyo Japanese, has base tone melodies, HL or LHL (Haraguchi, 1977), in addition to lexical pitch accent. Unlike previous studies on processing effects of Japanese homophones (Cutler & Otake, 1999; Minematsu & Hirose, 1995) that focus on visual recognition in Tokyo Japanese (Hino, Kusunose, Lupker & Jared, 2013; Sekiguchi, 2006), the current experiment measures reaction times to auditory stimuli in a lexical decision task comparing Tokyo Japanese with Kansai Japanese, whose speakers are typically bi-dialectal speakers of Tokyo Japanese as well. Results of the study will shed light on the issue of how pitch accent affects spoken word recognition in two different pitch accent varieties of Japanese.

3pSC80. The native language benefit for voice recognition is not contingent on lexical access. Nicholas R. Monto, Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT, nicholas.monto@uconn.edu), Adriel J. Orena, and Linda Polka (School of Commun. Sci. and Disord., McGill, Montreal, QC, Canada)

Listeners show heightened talker recognition for native compared to nonnative speech, formalized as the language familiarity effect (LFE) for voice recognition. Some findings suggest that language comprehension is the locus of the LFE, while others implicate expertise with the linguistic sound structure. These hypotheses yield different predictions for the LFE with time-reversed speech, a manipulation that precludes lexical access but preserves some indexical and phonetic properties. Research to date shows discrepant results for the LFE with this impoverished signal. Here we reconcile this discrepancy by examining how the amount of exposure to talkers' voices influences the LFE for time-reversed speech. Three experiments were conducted. In all, two groups of English monolinguals were trained and then tested on the identification of four English talkers and four French talkers; one group heard natural speech and the other group heard time-reversed speech. Across the experiments, we manipulated exposure to the voices in terms of number of training trials and duration of the talkers' sentences. A robust LFE emerged in all cases, though the magnitude was attenuated as the amount of exposure decreased. These results are consistent with the account that the LFE for talker identification is linked to the sound structure of language.

3pSC81. Language familiarity mediates identification of bilingual talkers across languages. Adriel John Orena, Linda Polka (School of Commun. Sci. and Disord., McGill Univ., 2001 McGill College, 8th Fl., Montréal, QC H3A 1G1, Canada, adriel.orena@mail.mcgill.ca), and Rachel M. Theodore (Dept. of Speech, Lang., and Hearing Sci., Univ. of Connecticut, Storrs, CT)

Many studies show that listeners are more accurate at identifying talkers in their native language than in an unfamiliar language; yet, little is known about the nature of this *language familiarity effect* in bilingual speech. Here, we investigate the links between language and talker processing further by assessing listeners' ability to identify bilingual talkers across languages. Two groups were recruited: English monolinguals and English-French bilinguals. Participants learned to identify bilinguals speaking in only one language (English); they were then tested on their ability to identify the same talkers speaking in the trained language (*same language context*: English) and in their other language (*different language context*: French). Both monolinguals and bilinguals showed above chance performance in identifying talkers in both language contexts at test, confirming that there is sufficient information in bilingual speech to generalize across languages. Moreover, the results showed a language context effect that was facilitated by language familiarity: monolinguals showed a substantial decrease in performance between the same and different language contexts, whereas the bilinguals—who understood both languages at test—showed only a slight decrease in performance between the two language contexts. These results indicate that language familiarity affects talker encoding and retrieval, even for bilingual speech.

3pSC82. Interspeech posture in Spanish-English bilingual adults. Merrily R. Shary, Kyna S. Betancourt, Stefan A. Frisch, and Nathan D. Maxfield (Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, merrily@usf.edu)

Interspeech posture (ISP) is a term used to define the position of a person's articulators when preparing to speak. Wilson and Gick (2014) found French-English bilinguals used different tongue posture during ISP depending on which language they were speaking. The authors suggested that ISP may be representative of a speaker's phonological knowledge in a particular language but recommended other languages be examined. The purpose of this study was to replicate Wilson and Gick (2014) using Spanish-English bilingual speakers. To this end, bilingual Spanish-English adults were asked to produce sentences while speaking in monolingual and bilingual modes. While they were speaking, ultrasound images of the oral cavity were obtained and tongue tip height during each interutterance pause was measured. Additionally, monolingual English speakers rated the accentedness of each bilingual's speech in English as a behavioral correlate of language proficiency. Overall results of this study supported Wilson and Gick (2014); bilingual Spanish-English speakers judged to be native sounding in English only utilized similar postures in monolingual and bilingual speaking modes. Different measures of ISP and speaker variability will also be completed to determine if there is a better method for measuring ISP in Spanish-English bilinguals.

3pSC83. A preliminary investigation of velar-vowel coarticulation in Spanish-English Bilinguals. Stefan Frisch, Marsha Allen, Kyna Betancourt, and Nathan Maxfield (Commun. Sci. and Disord., Univ. of South Florida, 4202 E Fowler Ave., PCD1017, Tampa, FL 33620, sfrisch@usf.edu)

Coarticulation, the influence of one speech sound on another in running speech, has been suggested as an index of speech motor control. Therefore, proficient and stable coarticulation could be a diagnostic tool in language assessment and speech-language pathology. In the present study coarticulation was investigated in Spanish-English bilingual adults to determine whether velar-vowel coarticulation patterns differ in English versus Spanish. Velar-vowel coarticulation was measured in six Spanish-English bilingual university students. Participants recited sentences in blocks of English, Spanish, and a mix of both languages. Each sentence included a target word starting with /k/ + vowel. Tongue movement was recorded using a sub-mental ultrasound probe held in a stabilizing helmet. Ultrasound images were analyzed to identify tongue posture following procedures of Frisch & Wodzinski (2016, Journal of Phonetics, doi: 10.1016/j.wocn.2016.01.001). The results indicate a pattern similar to that found for Interspeech Posture in French-English bilinguals (Wilson & Gick, 2014, Journal of Speech, Language, Hearing Research, doi: 10.1044/2013$_{\mathrm{JSLHR}}$-S-12-0345). Some bilinguals showed a single pattern for both languages while others produced distinct velar postures between English and Spanish.

3pSC84. Speech sound naturalness alters compensation in response to transformed auditory feedback. Sadao Hiroya and Takemi Mochida (NTT Commun. Sci. Labs., 3-1 Morinosato-Wakamiya, Atsugi, Kanagawa 2430198, Japan, hiroya.sadao@lab.ntt.co.jp)

Articulatory compensations in response to real-time formant perturbation have revealed that auditory feedback plays an important role in speech production. However, these compensatory responses were at most 40% for formant shifts and varied depending on vowel type and subjects. Although previous formant perturbation studies have been done using linear predictive coding (LPC), it is known that the estimation accuracy for low vowels and female speech would be degraded due to a glottal source-vocal tract interaction. To improve the accuracy, we have developed a real-time robust formant tracking system using phase equalization-based autoregressive exogenous (PEAR) model which utilizes the glottal source signals measured by electroglottography. In this study, we compared compensatory responses to real-time formant perturbation using PEAR and LPC. Eleven Japanese subjects (seven females) read a Japanese mora (/hi/ or /he/) with headphones. The first two formant frequencies were altered. Results showed that compensatory responses using PEAR were significantly larger than LPC. Moreover, naturalness of altered speech sounds was improved by PEAR. This indicates that improving speech sound naturalness by PEAR led to larger compensatory responses. Therefore, our system would be useful to understand the auditory feedback mechanisms in more detail.

3pSC85. Speech stability as a characteristic of the individual. Stefan Frisch, Jessica Massey, and Nathan Maxfield (Commun. Sci. and Disord., Univ. of South Florida, 4202 E Fowler Ave., PCD1017, Tampa, FL 33620, sfrisch@usf.edu)

Coarticulation, the influence of one speech sound on another in running speech, has been suggested as an index of speech motor control. Therefore, proficient and stable coarticulation could be a diagnostic tool in language assessment and speech-language pathology. In the present study, stability in the production of velar-vowel coarticulation was measured within speakers across experimental sessions to determine whether differences in stability can be attributed to individual differences in speech production ability. Velar-vowel coarticulation was recorded in children aged 8-12 in three sessions roughly one month apart. Participants recited sentences that included a target word starting with /k/ + vowel. Ultrasound images were analyzed to identify tongue posture and stability in posture across repetitions following procedures of Frisch, Maxfield, & Belmont (2016, Clinical Linguistics & Phonetics, doi:10.3109/02699206.2015.1137632). Preliminary analysis of two participants suggests there is less variation in stability within a participant across sessions than there is between participants. Analysis of the remaining participants is ongoing. This suggests that speech stability may indeed be a property of an individual speaker. If this finding holds, there are implications for models of speech production that include individual differences, including differences in language proficiency or clinical diagnosis.

3pSC86. Event related speech readiness potentials to spoken words and CV's. Silas Smith, Robert Sears, and Al Yonovitz (The Univ. of Montana, Dept. of Communicative Sci. and Disord., Missoula, MT 59812, al.yonovitz@umontana.edu)

This study investigated the temporal and dynamics of brain activity to speech and motor preparation. Previous EEG studies have identified a slow negativity wave that occurs previous to the onset of speech. Brain electrical potentials were obtained in subjects prior to initiation of speech in a using a single vertex electrode. Both a repeated single syllable word and CV's were used. The sample rate permitted an analysis of both slow negative waves and faster neurogenic signals. The purpose of this research was to establish a real-time event related brain electrical potential system. This research uses a vocal signal as the marking point, and displays in real time the event-related potential. Each time epoch was sampled at 25600 samples/sec. One second of these signals were averaged for 100 trials just prior to initiation of the word "pool" or a randomly presented CV. The inter-trial interval was approximately 25 seconds. For the 100 trials, each trial was saved. Separate odd and even event potentials were averaged. The obtained waveform (pre-speech) was consistent and reliable between and within subjects. The micro-structure of the waveform was also obtained showing specific waveform morphology. The reliability was greater for the CV productions.

3pSC87. What the f*: An acoustic-pragmatic analysis of meaning in The Wire.** Erica Gold and Dan McIntyre (Dept. of Linguist and Modern Lang., Univ. of Huddersfield, Queensgate, Huddersfield HD1 3DH, United Kingdom, e.gold@hud.ac.uk)

The award-winning television drama, The Wire, contains a famous scene in which Detectives Moreland and McNulty discuss the crime scene they are investigating using only variations of a single expletive—fuck. Despite limited vocabulary, viewers are able to extract meaning and interpret the scene. This study considers all of the expletives produced in the scene, and carries out an acoustic analysis of / ʌ /—F1~F3 (midpoint and averages) and vowel duration. These measurements are then used in combination with Gricean (Grice 1975) and neo-Gricean (e.g. Horn 1984) pragmatic analysis in an attempt to categorize the intended meaning of fuck as: intensification, confusion, dissatisfaction or suspicion (Fairman 2009). Vowel measurements are considered for Moreland and McNulty individually, and vowel normalization is carried out in order to determine pragmatic categorizations across both detectives. This paper argues that acoustic-phonetic analysis can augment Gricean and neo-Gricean pragmatic analysis by providing a means of determining whether what appears at a lexical level to be conversationally implicated meaning is in fact conventional implicature conveyed at the phonetic level.

3pSC88. Historical, current, and legal bases for forensic speaker identification. Al Yonovitz (Dept. of Communicative Sci. and Disord., The Univ. of Montana, Missoula, MT 59812, al.yonovitz@umontana.edu), Herbert Joe, and Joshua Yonovitz (Yonovitz & Joe, LLP, Dallas, TX)

It is of great interest to the legal system to identify individuals by their voice. Controversy in this area has continued for nearly seven decades with states divided with regard to the merits and standards used for forensic voice identification. Forensic speaker identification and admissibility of expert testimony has always been questionable, and is meeting more legal and scientific criticism. A number of methodologies have evolved that provide different approaches to speaker identification. These methods will be contrasted with a systematic review. In addition, the legal qualifications to testify as an expert witness in state or Federal courts—regardless of specialty, including in civil or criminal cases have evolved since the landmark 1923 *Frye* case, as well as the more sweeping 1993 *Daubert* case, and continue to evolve. This presentation discusses the history of the qualifications, an overview of different jurisdictional requirements, and the legal requirements of a speaker identification voice expert. The enigma of speaker identification for use in forensic applications will be illustrated by utilizing actual cases applying various methods of speaker identification or elimination.

Session 3pSP

Signal Processing in Acoustics: Acoustic Communications

Grace A. Clark, Cochair
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Henry A. Scarton, Cochair
Mechanical, Aerpspace, and Nuclear Engineering, Rensselaer Polytechnic Institute, Rensselaer Polytechnic Institute, 110 Eight Street, JEC 4008, Troy, NY 12180

Tadashi Ebihara, Cochair
Faculty of Engineering, Information and Systems, University of Tsukuba, 1-1-1 Tennodai, Tsukuba 3058573, Japan

Chair's Introduction—1:30

Invited Papers

1:35

3pSP1. Digital acoustic communications through solids. Henry A. Scarton (Mech., Aerpspace, and Nuclear Eng., Rensselaer Polytechnic Inst., Rensselaer Polytechnic Inst., 110 Eight St., JEC 4008, Troy, NY 12180, scarton@rpi.edu), Gary Saulnier (Elec. Comput. and Systems Eng., Rensselaer Polytechnic Inst., Troy, NY), Kyle Wilt (Mech., Aerpspace, and Nuclear Eng., Rensselaer Polytechnic Inst., Troy, NY), and Michael Cunningham (Elec. Comput. and Systems Eng., Rensselaer Polytechnic Inst., Troy, NY)

Digital acoustic communication solid channels make it possible to send data through metallic barriers with up to very high rates while also transferring power and with lower complexity, lower size and power. This paper presents different techniques for achieving this including data transfer rates up to sever kilobits per second (kbps) using simple switching technique to many megabits per second through thick steel. Modulation techniques include Chirp-On-Off Keying (C-O-OK) to Orthogonal Frequency Division Multiplexing (OFDM) and even Multiple-Input Multiple -Output (MIMO) approaching 700 Mbps. Frequencies up to 4 MHz are used with piezoelectric elements propagating longitudinal waves or transversely polarized shear waves serving as the sending and receiver sensors.

1:55

3pSP2. Characterizing ocean acoustic channel through databases: Issues, requirements, and methods. Tsih C. Yang (College of Information Sci. and Electron. Eng., Zhejiang Univ., 38 Zhe Da Rd., Bldg. of Information Sci. & Elec. Eng., Xihu District, Hangzhou 310058, China, tsihyang@gmail.com) and San Ho Huang (Ocean College, Zhejiang Univ., Zhoushan, Zhejiang, China)

There is a growing interest in underwater acoustic telemetry. This calls for methods to evaluate the performance of various (proposed) modulation and equalization algorithms under the same channel conditions. An ideal solution is to develop acoustic channel simulators. However, this effort has not been fruitful since acoustic channel simulations (at practical communication frequencies) so far have failed to capture the characteristics of real ocean channels. Currently, acoustic communications still rely on field experiments for evaluation of the algorithm performance. The problem with this approach is that each experiment is usually tailored to a particular modulation scheme and the cost prohibits testing of many different algorithms under the same ocean conditions. Some data-based channel simulators have been proposed where realizations of the channel impulse response (CIR) functions are generated based on the scattering function estimated from real data but it is still not adequate. A scheme to build ocean channel database is proposed in this paper, where CIR is deduced from data on a fractional symbol-by-symbol basis with negligibly small channel estimation error (using the surrogate signal predication error as an indicator) so that the developers can test the performance of different algorithms under same, realistic ocean conditions.

2:15

3pSP3. Harbor trial of underwater acoustic communication using Doppler-resilient orthogonal signal division multiplexing. Tadashi Ebihara (Faculty of Eng., Information and Systems, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8573, Japan, ebihara@iit.tsukuba.ac.jp), Geert Leus (Faculty of Elec. Eng., Mathematics and Comput. Sci., Delft Univ. of Technol., Delft, Netherlands), and Hanako Ogasawara (Dept. of Earth and Ocean Sci., National Defense Acad., Yokosuka, Kanagawa, Japan)

Underwater acoustic channels are characterized by a severe spread in time and frequency. To provide a highly reliable communication environment in doubly spread channels for UWA communication, we have proposed Doppler-resilient orthogonal signal-division multiplexing (D-OSDM). D-OSDM is a combination of the OSDM technique and orthogonal multiple access, and it preserves orthogonality among data streams even after propagation through doubly spread channels. We have conducted test tank experiments and simulations, and have found that D-OSDM could cope well with time- and frequency-spread channels and that it achieves a good communication quality [1]. In this talk, we will show some results of our first sea trial of D-OSDM, that was performed on 21 June, 2016 at Hashirimizu port, Kanagawa, Japan. In this trial, the transmitter is fixed on the harbor quay, and the receiver is fixed on a remotely-operated survey boat. Acoustic communication was performed by moving the boat to generate a severe spread in frequency. The experiments show that D-OSDM achieves a good communication quality even if the boat moves at a speed of about 1.5 kt. [1] Tadashi Ebihara and Geert Leus: Doppler-Resilient Orthogonal Signal-Division Multiplexing for Underwater Acoustic Communication, IEEE J. Ocean. Eng. 41 (2016) 408-427.

2:35

3pSP4. Multiuser and multiple-input-multiple-output underwater acoustic communication using time reversal. Takuya Shimura, Yukihiro Kida, Mitsuyasu Deguchi, Kohji Meguro, Yoshitaka Watanabe, and Hiroshi Ochi (Maine Tech. Development Dept., JAMSTEC, 2-15 Natsushima-cho, Yokosuka, Kanagawa 237-0061, Japan, shimurat@jamstec.go.jp)

Time reversal is an effective method of channel equalization for underwater acoustic communication. Its temporal and spatial focusing effect collects long time spread signals due to a rich multipath environment inherent in underwater acoustics. Additionally, it is easy to compensate Doppler frequency shift after time reversal focusing. Recently, demands for multiuser or multiple-input-multiple-output (MIMO) communication have increased widely for multiple autonomous underwater vehicles (AUVs) or high rate underwater acoustic network communication. Similarly, in Japan Agency Marine-Earth Science and Technology (JAMSTEC), a plan of multiple AUVs operation is being progressed. Time reversal is a promising solution also for such multiuser or MIMO communication by realizing space division multiplexing (SDM) based on its spatial focusing and nulling effect without sacrificing data transmission rate. In this study, the performance of time reversal is analyzed with simulation or real data comparing with orthogonal frequency-division multiplexing (OFDM) as a conventional or commonly used method. As results, the ability of time reversal to suppress multiuser or multistream interferences is much better than those of maximum ratio combining (MRC), zero forcing (ZF) or minimum mean-square error (MMSE) OFDM, in both vertical and horizontal communication.

2:55

3pSP5. Differential frequency hopping performance in doubly spread underwater acoustic communication channels. Geoffrey S. Edelson (Maritime Systems & Technol., BAE Systems, MER15-2350, P.O. Box 868, Nashua, NH 03061-0868, geoffrey.s.edelson@baesystems.com) and Luca Cazzanti (Machine Learning and Data Sci., Ctr. for Maritime Res. and Experimentation, La Spezia, Italy)

Underwater acoustic communication requires waveforms that are robust to the signal distortions caused by the channel. Many waveforms used for this purpose require the transmission of training symbols that span the inter-symbol interference (ISI) to compensate for these channel effects. Differential frequency hopping (DFH) is a fast frequency hopping, digital signaling technology that requires minimal information at the transmitter to communicate in the underwater channel. The relationships between the parameters of Doppler spread-inducing underwater environments and DFH bit-error performance are characterized. Wind speed determines the nature of the effect that the water surface imposes on acoustic DFH waveforms with low wind speeds resulting in an essentially flat, low-absorption sea surface. In this case, strong surface reflections and little frequency spreading make ISI the dominant effect on the received waveforms. At high wind speeds, air bubbles in the surface layer absorb almost all energy incident on the surface, resulting in no surface reflections reaching the receiver so that the surface has little effect on the received signal. The intermediate ranges of wind speed, with a mix of ISI and surface-induced Doppler spread, pose the greatest challenge. Simulations and at-sea experiments show that DFH performance is robust across environmental conditions.

3:15–3:30 Break

Contributed Papers

3:30

3pSP6. Waveform designs for computationally efficient acoustic response estimation. Jacob L. Silva (ECE Dept., Univ. of Massachusetts Dartmouth, Darmouth, MA) and Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, pgendron@umassd.edu)

Broadband source waveforms are constructed that allow for the minimum mean square error (MMSE) estimation of an observed acoustic response function with minimal computational effort at the receiver. The waveforms offers near perfect auto and cross correlation properties over the limited propagation delay spread of the acoustic response allowing for least squares estimation of the whitened response with simple matched filtering. MMSE estimation of the whitened response can be accomplished in many cases from this least squares estimate. We compare the approach to conventional maximal length sequences as well as Gold sequence to determine the merits of the approach we further demonstrate the approach for communications at low signal to noise ratios.

3:45

3pSP7. A covert underwater acoustic communication method based on spread spectrum digital watermarking. Shaofan Yang, Zhongyuan Guo, Qunyan Ren, and Shengming Guo (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North Fourth Ring Rd., Beijing 100190, China, 1172966054@qq.com)

The covert underwater acoustic communication is necessary under some circumstances when the contents of transmission or location of transmitter are required to be concealed. In this paper, the signal transmitter and the information detector are designed, which hides the spread spectrum digital watermark (SSDW) into the log-spectrum of dolphin whistles that are as a carrier and detects SSDW by matched filtering in the same log-spectrum. Various factors of noise, carrier and multipath in channels may degrade the performance of the method. In order to analyze their effects, corresponding channel models in log-spectrum domain are derived from the additive Gaussian noise channel and the underwater time-varying acoustic multipath channel. Results suggests that the effects of channel noise and the carrier are equivalent to that of two additive Gaussian noise terms attached to the correct decision and the proposed method also has a good multipath extension tolerance performance. Preliminary results obtained form at-sea experiments showed that the data rate can achieve 6 bps and bit error rate (BER) under 10^{-3} in the distance of 5 km.

4:00

3pSP8. Efficient estimation and compensation of Doppler shift for multicarrier signals in underwater acoustic communications. Saraswathi Karanam and Ravi Shankar S (Electronics & Commun., R.V college of Eng., Dept. of ECE, RVCE, Mysore Rd., Bangalore, Karnataka 560059, India, ksaraswathi@rvce.edu.in)

Orthogonal frequency division multiplexed (OFDM) carriers are well studied in literature for their ability to tolerate multipath effects in the presence of ambient noise in an underwater acoustic (UWA) channel. However UWA Doppler destroys the orthogonality of subcarriers, resulting in severely degraded Signal to Noise ratio (SNR). To mitigate this effect, we present a novel method for Doppler estimation and compensation over this channel. This method involves transmitting a training preamble of up to three OFDM symbols. Doppler estimation is carried out based on a sliding window correlation over an OFDM symbol period between the received and local copy of time samples. After rejecting spurious peaks, the correlation values are normalized and existence of Doppler detected when the correlation value exceeds a set threshold. The Doppler shift is then estimated post FFT. Specifically the phase angle of received pilot samples is averaged over three OFDM symbols. This estimated shift is used to compensate the Doppler shift in the following received data symbols. Simulations have been performed for different Doppler shifts over an UWA multipath channel, with BER as a parameter. Significantly, the approach avoids resampling and the need for buffering received data packets. This method also reduces computational complexity significantly.

4:15

3pSP9. Multichannel equalization method based on vector time-reversal and decision feedback equalizer technology. Xueli Sheng, Yewu Ruan, Dian Lu, JIngwei Yin, and Longxiang Guo (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, shengxueli@hrbeu.edu.cn)

A kind of underwater communication receiver combined with Vector Time-Reversal algorithm and multichannel decision feedback equalizer (VTR-MDFE) is studied, which can address the problem of inter-symbol interference and waveform distortion, resulting from time-varying multipath channel. Impacts of environmental noise can be suppressed by use of vector sensor's combined directivity and Corresponding vector signal processing technology. Meanwhile, error rate is reduced while system capacity is enhanced by time-reversal algorithm's anti-multipath ability. The performance of VTR-MDFE are proved by simulated results in the communication channel provided by a ray-based acoustic model, for different ocean conditions and source-receiver geometries.

WEDNESDAY AFTERNOON, 30 NOVEMBER 2016 NAUTILUS, 1:00 P.M. TO 2:45 P.M.

Session 3pUWa

Underwater Acoustics: Inversion, Beam-forming, and Calibration II

Steven A. Stotts, Chair

Environmental Sciences Laboratory, Applied Research Labs/The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78759

Contributed Papers

1:00

3pUWa1. Application of maximum entropy to statistical inference for inversion of data from a single track segment. Steven A. Stotts and Robert A. Koch (Environ. Sci. Lab., Appl. Res. Labs/The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78759, stotts@arlut.utexas.edu)

An approach is presented for statistical inference, based on maximum entropy (ME) with inversion data from a single source track segment, to account for model mismatch. A previous approach requires data from multiple track segments to set the, otherwise undetermined, ME constraint value specifying the posterior probability density (PPD). One effect of mismatch is that the lowest cost inversion solutions for some parameter values, e.g., source track parameter values obtained from GPS measurements or source levels from towed sources, may be well outside a relatively well known, narrow, uncertainty interval. The basis for the new approach is that the ME constraint value is determined by requiring for such a parameter value that the inferred uncertainty interval encompass the entire uncertainty interval comprising its prior information. Motivating this approach is the hypothesis that the proposed constraint determination, applied to the PPD for a model space broader than that parameter value's prior, could account for the effect of mismatch on the inferred, but *a priori* less well defined, values for other parameters.

1:15

3pUWa2. Wave physics of the frequency difference autoproduct: Helmholtz equation analysis. David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI, drd@umich.edu) and Brian M. Worthmann (Appl. Phys., Univ. of Michigan, Ann Arbor, MI)

The frequency-difference autoproduct (Δf-AP) is a quadratic product of solutions of the inhomogenous Helmholtz equation that differ only in frequency, and may be easily constructed from the Fourier transform of measured time-domain signals. Recent findings involving beamforming (Abadi *et al.*, 2012, JASA 132, 3018-3029) and matched field processing (Worthmann *et al.*, 2015, JASA 138, 3549-3562) suggest that the Δf-AP is similar to an acoustic field at the difference frequency, even when the difference frequency lies below the recorded signal's bandwidth. This presentation provides mathematical analysis that supports this interpretation and indicates its limitations, along with examples from a Lloyd's mirror environment where a frequency sweep from 1-2kHz was broadcast from 100m below the reflecting surface out to a range of 2km. In particular, the Δf-AP in time-independent, inhomogeneous, multipath environments should be locally indistinguishable from an acoustic field at the difference frequency when: (*i*) the in-band acoustic propagation is well described by a ray-path sum, (*ii*) the in-band field gradient is dominated by phase (not amplitude) variations, and

(*iii*) the products of the signal bandwidth and ray travel-time differences are large enough so that a signal-bandwidth average suppresses ray-path cross terms from the quadratic product. [Sponsored by ONR and NSF.]

1:30

3pUWa3. Wave physics of the frequency difference autoproduct: Bilinear time-frequency analysis. Brian M. Worthmann (Appl. Phys., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, bworthma@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Frequency difference beamforming (Abadi *et al.*, 2012, JASA, 132, 3018-3029) and matched field processing (Worthmann *et al.*, 2015, JASA, 138, 3549-3562) are array signal processing techniques that determine acoustic ray-path directions and source locations, respectively, from sparse array recordings by shifting the signal processing to a lower, out-of-band, difference-frequency bandwidth. This frequency down-shift is accomplished through the use of the frequency difference autoproduct (Δf-AP), a quadratic product of complex frequency-domain acoustic field amplitudes at two different frequencies. In this presentation, the physical and mathematical underpinnings of this quadratic product are discussed. Specifically, in the branch of applied mathematics termed bilinear time-frequency analysis, this Δf-AP is actually one of four quadratic time-frequency functions, each interrelated through forward and inverse Fourier transforms. The other three functions are the time-domain autoproduct, the Wigner distribution, and the ambiguity function. These four functions and several of their interesting properties are reviewed, including their relationship to the Wiener-Khintchine Theorem, and the spatial and spectral coherence and correlation functions. Additionally, the behavior of these functions when evaluated from simulated acoustic recordings from ideal single-path and multi-path acoustic environments is shown and compared. [Sponsored by ONR and NSF.]

1:45

3pUWa4. Low frequency beamforming in shallow water environments. Shima Abadi, Cole Amaratunge, Gavin Boyd, and Matthew Daniels (Univ. of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011, abadi@uw.edu)

Ray theory is a high frequency approximation of the wave equation solution. Beamforming is a spatial filtering method that uses the ray theory to estimate the arrival angle from the array recordings. However, the ray theory is restricted to situations where water column is much deeper than the propagated wavelength and fails when water depth is smaller compared to the acoustic wavelength (i.e., in shallow-water low-frequency propagation). In this situation, the underwater sound propagation is modeled by normal-modes theory, the exact solution of the wave equation. In the theory of normal modes, each mode propagates at a different incident angle and the total field is composed of a discrete sum of the propagation modes excited at the broadcast frequency. In this situation, the beamforming performance degrades because of the interference between individual mode incident angles. In this presentation, the performance of the conventional beamforming technique is investigated when low frequency signals are propagated in shallow water environments and recorded by either a vertical or a horizontal linear array of hydrophones. Simulations are undertaken to understand how the resolved angle from the conventional beamforming method is related to the bearing angle and the incident angle of each mode.

2:00

3pUWa5. The investigation of the method for measuring the low-frequency radiated sound power in a reverberation tank. Yi-ming Zhang, Rui Tang, Qi Li, and Da-jing Shang (Harbin Eng. Univ., Nantong Str. Nangang Dist. No.145, Harbin 150001, China, tangrui@hrbeu.edu.cn)

In order to measure the low-frequency radiated sound power of the underwater sound source in a reverberation tank, the measuring method based on the characteristic acquisition of the testing sound field is proposed. Based on the normal-mode theory, the spatial averaging properties of the reverberation sound field with the ideal boundaries is firstly derived, of which the sound field transfer relationship from the reverberation field to the free field is obtained. In addition, for the purpose of obtaining the characteristic of the testing sound field with the general boundaries, the numerical method based on the ACTRAN software is utilized. According to obtaining the characteristic of the testing sound field in advance, the low-frequency radiated sound power can be got by measuring the square pressure in a reverberation tank. The proposed method is checked by the experiment measurement. The test results show that by utilizing the proposed method, the low-frequency radiated sound power of the underwater sound source has been accurately measured in the glass tank below the Schroeder cut-off frequency, of which the radiated sound power level deviation is less than 2 dB compared with the free field.

2:15

3pUWa6. Calibration of the matched filter beam output from the triplet portion of the Five Octave Research Array. Dale D. Ellis (Phys., Mount Allison Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com) and Jie Yang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The Five Octave Research Array (FORA) [K.M. Becker and J. R. Preston, Proceedings Oceans 2003, 2607-2610] includes a triplet section in which the elements are three omnidirectional sensors. The elements can be combined as a standard linear array to produce conical beam patterns, or the triplets can be used to form directional beams that allow for "left-right" discrimination. The beam output can then be matched filtered to produce high-temporal-resolution beams over the full 360° azimuth, and identify scattering features in reverberation data. To extract scattering strengths, the calibration of the beams and the matched filter output must be known accurately. Here we look at results from the 2013 Target and Reverberation Experiment (TREX13), where data were received on the FORA array, then beamformed and matched filtered by various procedures that had different normalizations. The broadside beam and several natural beams of the linear array can readily be calibrated against a simple "exact" procedure. No simple procedure was obtained for the directional beams, but comparisons of the peaks of the features in the left and right triplet beams compare well with the peaks of the linear beams.

2:30

3pUWa7. Source localization in underwater waveguides using machine learning. Haiqiang Niu and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., San Diego, CA 92093-0238, hniu@ucsd.edu)

The problem of underwater acoustic source localization is solved using an artificial neural network (ANN) under the machine learning framework. Source localization in a waveguide is posed as a classification problem by training a feed-forward neural network (FFN) with one hidden layer. In this paper, the acoustic pressure signals received by a vertical linear array are preprocessed by constructing the normalized cross-spectral density matrices (CSDM), which are used as input of the FFN. Each unit of the output layer represents one possible range (Here, for simplicity, source range is the only parameter that needs to be determined). Different from model-based localization methods such as matched field processing (MFP), ANN as a data-driven method can learn features from real acoustic data, thereby bypassing the sound propagation modeling step completely. Simulations show that FFN achieves a good performance in determining the source ranges even with deficient training data samples and low signal-to-noise ratios (SNRs). The validity of FFN in source localization is further demonstrated with vertical array data from Noise09 experiment where the ship is located successfully using FFN.

Session 3pUWb

Underwater Acoustics: Seabed and Sea Surface Interactions

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Contributed Papers

3:00

3pUWb1. A comparison between direct measurements and inference results from an acoustic propagation experiment in Currituck Sound. Megan S. Ballard, Jason D. Sagers, Kevin M. Lee, Andrew R. McNeese (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Thomas G. Muir (Appl. Res. Labs. at the Univ. of Texas at Austin, Austin, TX), and R. D. Costley (GeoTech. and Structures Lab., U.S. Army Engineer Res. & Development Ctr., Vicksburg, MS)

An acoustic propagation experiment was conducted in Currituck Sound to characterize low frequency propagation in a very shallow-water environment with water depths of only a few meters. Signals from a Combustive Sound Source (CSS) were recorded on bottom mounted geophones and a vertical array of hydrophones. The CSS produces a broadband signal with significant low frequency energy, and the analysis presented in this paper focuses on frequencies below 2.5 kHz. A statistical inference method was applied to obtain an estimate of the sediment sound-speed profile as a function of depth in the seabed. During the experiment, i*n situ* measurements of compressional and shear wave speed and attenuation were also collected 30 cm below the sediment-water interface. Bimorph bender elements were employed to generate and receive horizontally polarized shear waves in the frequency range from 200 Hz to 1 kHz. Compressional waves were measured using cylindrically shaped piezoelectric elements operated in the 5 kHz to 100 kHz frequency band. Sediment acoustics models were fit to the *in situ* wave speed and attenuation measurements to enable comparison with the inferred low frequency sound speeds. [Work supported by ERDC, ARL:UT, and ONR.]

3:15

3pUWb2. On the dependence of the nonlinearity coefficient of marine silica sands on grain size, shape, and packing density. William Sanders, Allen Reed, and Warren Wood (Naval Res. Lab., Code 7433 Bldg. 1005, Stennis Space Ctr., MS 39529, wsanders@nrlssc.navy.mil)

Several applications of parametric sonars rely on utilizing nonlinear propagation of sound in marine sediments. Whereas it has been speculated that the nonlinearity coefficient for granular materials is expected to be relatively high, few measurements have been made to support the notion, nor has any theory been used to satisfactorily predict the parameter. Moreover, unlike fluids, the nonlinearity coefficient of granular materials does not follow from an equation of state, but is likely the composite effect of multiple mechanisms. Hence, we aim to compile a comprehensive set of measurements of the nonlinearity coefficient for a variety of sediments and conditions. This effort reports on the initial set of observations on unconsolidated quartz silica sands in water with grain sizes ranging from 0.05 to 1 mm, a variety of shapes (angular, sub-rounded, and rounded), and a variation in packing density. [This work was supported by ONR.]

3:30

3pUWb3. Near-field measurements of a parametric source. William Sanders (Naval Res. Lab., Code 7433 Bldg. 1005, Stennis Space Ctr., LA 39529, wsanders@nrlssc.navy.mil), Allen Reed, and Warren Wood (Naval Res. Lab., Stennis Space Ctr., MS)

In order to characterize nonlinear interactions within granular materials, measurements were made in the near-field of a 50 kHz transducer (an ITC 5270). High-resolution beam patterns were observed which reveal significant diffraction effects. Observed beam patterns were used to predict the difference frequency signal and this resulted in more than a 10 dB difference from that predicted by assuming constant amplitudes. This points to an inadequacy of the typical treatment of the parametric field as that due to collimated beams of constant amplitude in the near field, e.g. as in the Moffet and Mellen model. [This work was supported by ONR.]

3:45

3pUWb4. Estimating seabed structure from perturbed layer resonance fringes. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu)

Waves reflecting from layered media give rise to constructive and destructive interference fringes. These well-known fringes occur at integer multiples of half- and quarter-wavelength conditions respectively. When any of the layer boundaries are rough, the fringe pattern is perturbed. One class of observed perturbations is when the when the fringe exhibits a stair-step pattern in frequency and angle, instead of the smooth behavior predicted by theory. These perturbations can lead to difficulties for statistical inference when using the flat layer assumption. First, the physics behind the stair-step perturbations are described. Then, from the physics we show how to estimate or bound small-scale roughness parameters from the stair-step perturbations. Both simulations and measured data are employed to probe the information content of reflection data for small-scale roughness in layered media. [Research supported by the ONR Ocean Acoustics Program.]

4:00

3pUWb5. Volume scattering from turbidite sequences. Derek R. Olson (Appl. Res. Lab., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, olson.derek.r@gmail.com) and Charles W. Holland (Appl. Res. Lab., Penn State Univ., State College, PA)

The seabed environment near continental margins is commonly composed of turbidite sequences, which are alternating layers of silt/sand and pelagic clay (mud). Scattering can occur both from interface roughness structure, as well as volume heterogeneities within each layer. A modeling technique is detailed that takes into account the layering structure of a turbidite for propagation into and out of the sediment, and uses the Born approximation to model the scattering. This model properly accounts for effects of the layering structure on the incident field (instead of using an effective

medium approximation), which is expected to be important particularly at high frequencies (relative to the scales of the layering). Model results from several types of environments will be shown, and also compared to bottom scattering data collected in the Ionian sea.

4:15

3pUWb6. Numerical study of three-dimensional sound reflection from corrugated surface. Youngmin Choo (Res. Inst. of Marine System Eng., Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul 151 - 744, South Korea, sks655@snu.ac.kr), Heechun Song (Scripps Inst. of Oceanogr., San Diego, CA), and Woojae Seong (Naval Architecture and Ocean Eng. and Res. Inst. of Marine Systems Eng., Seoul National Univ., Seoul, South Korea)

When a sound wave propagates in a water medium bounded by a smooth surface wave, reflection from a wave crest can lead to focusing and result in rapid variation of the received waveform as the surface wave moves [J. Acoust. Soc. Am. **125**, 66-72 (2009)]. In prior work, propagation paths have been constrained to be in a plane parallel to the direction of corrugated surface waves, i.e., a two-dimensional (2-D) propagation problem. In this paper, the azimuthal dependence of sound propagation as a three-dimensional (3-D) problem is investigated using an efficient, time-domain Helmholtz-Kirchhoff integral formulation. When the source and receiver are in the plane orthogonal to the surface wave direction, the surface wave curvature vanishes in conventional 2-D treatments and the flat surface simply moves up and down, resulting in minimal temporal variation of the reflected signal intensity. On the other hand, the 3-D propagation analysis reveals that a focusing phenomenon occurs in the reflected signal due to the surface wave curvature formed along the orthogonal plane, i.e., out-of-plane scattering.

4:30

3pUWb7. Measurements and predictions of surface Doppler and arrival time spread in a variety of wind conditions. Sean Walstead (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, sean.walstead@navy.mil)

Measurements from a bistatic surface scattering experiment are presented. Acoustic waveforms in the 200 kHz—400 kHz (VHF) frequency regime were transmitted and received in a variable speed wind wave channel at the Scripps Institution of Oceanography. Physics based analytic predictions for Doppler shift and arrival time spread are compared to experimental measurements. The effect of source/receiver depth and surface wave shape on acoustic properties of the surface multipath are quantified as a function of wind speed. The importance of transmit waveform Doppler sensitivity of common underwater communications signals is quantified for various surface scattering regimes. This work has direct application to the improved performance of phase coherent underwater acoustic communications systems and the remote sensing of gravity-capillary waves.

4:45

3pUWb8. Perturbation theory for mode coupling over a sloping penetrable bottom. Feilong Zhu (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No.21, Bei-Si-Huan-Xi Rd., Haidian District, Beijing 100190, China, zhufl@mail.ioa.ac.cn) and Eric I. Thorsos (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

In environments with water depth variations, one-way modal solutions involve mode coupling. One approach is to model the depth variation as a series of steps, and use a new set of modes at each step (Evans, JASA, 1983). Higham and Tindle (JASA, 2003) developed an accurate and faster approach using perturbation theory to locally determine the change in mode functions at steps and limit the required number of new sets of mode functions. The method of Higham and Tindle is limited to low frequency (< 250 Hz). We have extended the coupled mode perturbation method so it can apply to higher frequencies. The approach will be described and some examples shown.

5:00

3pUWb9. Toward a test of the existence of the Biot slow wave in sand using layer resonances. Anthony L. Bonomo, Gabriel R. Venegas, Marcia J. Isakson, and Preston S. Wilson (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713, abonomo@arlut.utexas.edu)

Geoacoustic models based on Biot's theory of poroelasticity have found some success in predicting the interaction of underwater sound with sandy sediments. One of the unique features of Biot theory is its prediction that poroelastic media can support two types of compressional waves—the conventional compressional wave, referred to as the "fast wave," and an additional, slower compressional wave due to the out-of-phase motion of the saturating fluid with respect to the solid frame dubbed the "slow wave." While Biot-based sand models have been shown to accurately fit measured compressional sound speed and attenuation data, shear wave sound speed and attenuation data, bottom loss data, and backscattering strength data, the so-called slow wave has not been directly measured in sand. Rather than attempt to directly measure the slow wave, it is the goal of this work to confirm or disprove the slow wave's existence from the resonances it causes in thin sand layers. It is expected that these resonances would be apparent in measurements of normal-incidence bottom loss. Finite element models are used to demonstrate the possible effectiveness of this approach. Experimental design and preliminary measurements are also considered. [Work supported by ONR, Ocean Acoustics.]

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, and Wednesday. See the list below for the exact schedule.

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

Committees meeting on Tuesday, 29 November

Committee	Start Time	Room
Acoustical Oceanography	7:30 p.m.	Kahili
Animal Bioacoustics	7:30 p.m.	Coral 2
Architectural Acoustics	7:30 p.m.	Lehua
Engineering Acoustics	7:30 p.m.	South Pacific 4
Physical Acoustics	7:30 p.m.	South Pacific 2
Psychological and Physiological Acoustics	7:30 p.m.	South Pacific 3
Structural Acoustics and Vibration	7:30 p.m.	South Pacific 1

Committees meeting on Wednesday, 30 November

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Coral 1
Musical Acoustics	7:30 p.m.	Kahili
Noise	7:30 p.m.	South Pacific 4
Signal Processing in Acoustics	7:30 p.m.	South Pacific 1
Speech Communication	7:30 p.m.	Coral 4
Underwater Acoustics	7:30 p.m.	Nautilus