Session 1aID

Interdisciplinary: Opening Ceremonies, Plenary Lectures

Whitlow Au, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

1aID1. Adventures of an expeditionary biologist: Neuroethology of ultrasonic communication in amphibians. Peter M. Narins (Integrative Biology & Physiol., UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095-1606, pnarins@ucla.edu)

Animal communication occurs when a signal generated by one individual is transmitted through an appropriate channel and results in a behavioral change in a second individual. We have explored specific morphological, physiological, and behavioral adaptations in a wide variety of taxa that appear to have evolved specifically to tailor and sculpt intraspecific communication systems. In this lecture, I will review one of these adaptational studies that involves two distantly related organisms: the concave-eared torrent frog (*Odorrana tormota*), calling near fast-flowing mountain streams of Anhui Province, Central China, and the endemic Bornean frog, *Huia cavitympanum*, living in a very similar riverine habitat in Sarawak, Malaysia. In addition to the high-pitched audible components, these species' calls contain previously unreported ultrasonic harmonics. Our studies of these two Asian frogs revealed that they communicate acoustically using ultrasound and that their auditory systems are sensitive up to 34-38 kHz. This extraordinary upward extension into the ultrasonic range of both the harmonic content of the advertisement calls and the frogs' hearing sensitivity is likely to have coevolved in response to the intense, predominately low-frequency ambient noise from local streams.

Chair's Introduction—9:00

9:05

1aID2. Cartilage conduction hearing-The third sound conduction pathway. Hiroshi Hosoi (Nara Medical Univ., 840 Shijo-cho, Kashihara, Nara 6348522, Japan, hosoi@naramed-u.ac.jp)

Hosoi found in 2004 that a clear sound can be heard when a vibration signal is delivered to the aural cartilage from a transducer. This form of signal transmission is referred to as cartilage conduction (CC). Then, we have examined its mechanism, application, and superiority over the well-known air or bone conduction (AC or BC). We have published 11 English articles including JASA by 2016. AC, BC, and CC are distinguishable according to two points, namely, transmission media and movement of the skull bone. The transmission media are sound in AC, vibration in BC and CC. The movements of the bone are immobile in AC and CC, vibrated in BC. The world first CC hearing aids have been completed. It is very useful for the patients who cannot wear regular hearing aids because of atresia of external auditory canals, for example. More useful applications are cell phones and robots. The CC cell phone has a lot of advantage such as 1) easy volume control without button operation, 2) insulation from outer noise and obtaining the best S/N instantly, 3) no sound leakage to outward, 4) clean liquid crystal display, 5) easy to use even with hearing aids, 6) better suited for very small body type cell phone, and all that. There is no disadvantage compared with the regular phone. Various types of cell phone can be realized using CC. For example, wrist watch type, pen type, eye glass type, finger ring type, etc. Also, CC will give the good method to communication between robots and human beings.

Session 1aAA

Architectural Acoustics: Acoustics for Children and Pupils I

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

David Lubman, Cochair
DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683-4514

Invited Papers

10:35

1aAA1. Overview of acoustic environment of child daycare facilities in Japan: Current situation and recent researches. Keiji Kawai (Kumamoto Univ., 2-39-1 Kurokami, Kumamoto 860-8555, Japan, kkawai@kumamoto-u.ac.jp), Kanako Ueno, Saki Noguchi (Meiji Univ., Kawasaki, Kanagawa, Japan), and Hisao Funaba (Iwate Prefectural Univ., Takizawa, Iwate, Japan)

Demands for child daycare facilities are continuously increasing in Japan as the Japanese society is changing with respect to women's social advancement in the workforce. However, since there are virtually no guidelines for the acoustic design of daycare facilities in Japan, many daycare classrooms have been built without considering acoustics or sound insulation. As a result, the rooms tend to be too noisy and too reverberant for children in the process of language and listening acquisition. This is a report both on the present situation and architectural trend of daycare facilities and on the existing researches for the acoustic environment of daycare classrooms, aiming to share the information about daycare facilities in Japan for the discussions to develop guidelines of acoustic design.

10:55

1aAA2. Statistically defining the construct of "acoustic quality" in K-12 classrooms. Joonhee Lee, Laura C. Brill (Durham School of Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, joonhee.lee@huskers.unl.edu), Houston Lester, James Bovaird (The Dept. of Educational Psych., Univ. of Nebraska - Lincoln, Lincoln, NE), and Lily M. Wang (Durham School of Architectural Eng., Univ. of Nebraska - Lincoln, Omaha, NE)

This paper presents preliminary statistical analyses of acoustic measurements taken in over a hundred K-12 classrooms in Nebraska during the 2015-2016 academic year. Noise levels were continuously logged over two consecutive school days, three times seasonally. Other measurements included unoccupied background noise levels and room impulse responses. Equivalent sound pressure levels, percentile levels, reverberation times, and other assorted room acoustic metrics have been calculated from the measured data. Preliminary statistical analyses have been performed to investigate how each metric is related to each other and how the metrics are distributed across classrooms. These quantities have also been statistically analyzed to understand which are most pertinent to a comprehensive construct of "acoustic quality." This "acoustic quality" construct will subsequently be used to investigate how its components are related to the academic performance of students, and its inter-relationships with other environmental conditions such as indoor air quality, thermal, and lighting conditions. [Work supported by the United States Environmental Protection Agency Grant Number R835633.]

11:15

1aAA3. Exploring correlation between sound levels in active occupied classrooms and unoccupied classrooms. Laura C. Brill, Joonhee Lee, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 6349 Cedar Plz, #222, Omaha, NE 68106, lbrill@huskers.unl.edu)

A large-scale in-situ survey of environmental conditions, including acoustics, is being conducted in over 200 K-12 classrooms in multiple school districts by researchers at the University of Nebraska-Lincoln. Continuing analyses, conducted on logged acoustic sound levels collected seasonally over two occupied class days through Fall 2016, will be presented to determine how unoccupied noise levels are related to levels in the occupied active classroom. Do increased unoccupied noise levels correlate to increased occupied levels, and if so, to what degree? A number of methods are used to explore this relationship including spectral analyses of the logged data, further percentile level analyses, and linking the type of activity in the classroom more clearly to the logged sound levels. [Work supported by the United States Environmental Protection Agency Grant Number R835633.]

11:35

1aAA4. Teachers' accumulation of silence and voicing periods of continuous speech in classroom with different reverberation times. Pasquale Bottalico (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Lansing, MI 48910, pb@msu.edu)

The relationship between the distribution of continuous silence and voicing periods accumulated (duration of continuous voiced and unvoiced segments of 0.05 s) by teachers teaching in classrooms with different reverberation time (RT) was examined. Understanding the distributions of these accumulations and the relationship with the RT has implications on potential vocal fatigue risk factors and it can be a useful information for clinicians. Twenty-two primary school teachers were monitored over 1 or 2 four-hour workdays with the Ambulatory Phonation Monitor. The RT ranged between 0.58 s and 1.58 s, with a median equal to 0.9 s. The classrooms were classified as low RT (0.58 s $<$ RT $<$ 0.90 s) and high RT (0.90 s $\leq$ RT $<$ 1.58 s). The values of silence accumulation in low RT classrooms were significantly shorter than the accumulation in high RT classrooms in almost the whole range of accumulation periods (0.2 s—10 s). The difference was negligible for time shorter than 1.5 s and longer than 8 s. The values of voice accumulation in low RT classrooms are significantly shorter than the accumulation in high RT classrooms in almost the whole range of accumulation periods (0.2 s—1.31 s).

MONDAY MORNING, 28 NOVEMBER 2016 — SOUTH PACIFIC 4, 10:30 A.M. TO 11:55 A.M.

Session 1aAB

Animal Bioacoustics: Bat Echolocation: New Insights on Biosonar Production, Processing, and Performance from Field and Laboratory Investigations I

Hiroshi Riquimaroux, Cochair
Life and Medical Sciences, Doshisha University, Shandong University, 27 Shanda Nanlu, Jinan 250100, China

Laura Kloepper, Cochair
Biology, Saint Mary's College, 262 Science Hall, Saint Mary's College, Notre Dame, IN 46556

Chair's Introduction—10:30

Invited Papers

10:35

1aAB1. The model for echolocation of insects by using bat-like sounds. Ikuo Matsuo (Dept. of Information Sci., Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@mail.tohoku-gakuin.ac.jp)

Bats use frequency-modulated echolocation to identify and capture flying insects in real three dimensional space. Experimental evidence indicates that bats are capable of locating static objects with a range accuracy of less than 1 μs. A previously introduced model could localize objects in two-dimensional space by accurately estimating the object's range at each timepoint by extracting temporal changes from the time-frequency pattern and interaural range differences while echoes were measured at two receiving points by intermittently emitting LFM sounds. In addition, this model was shown to localize moving objects in two-dimensional space by accurately estimating the object's range under the influences of the Doppler shift, which was dependent on the movements. However, it was not evaluated whether this model could detect and localize small sized objects, e.g., small moths, whose echoes were weak. In this presentation, the echoes were measured from two kinds of moth, *Spodoptera litura* and *Helicoverpa armigera*. The accuracies of estimated range and direction were examined while echoes were measured from flying moths by intermittently emitting bat-like sounds.

10:55

1aAB2. Information-theoretic assessment of the peripheral dynamics in the biosonar system of horseshoe bats. Rolf Müller, Anupam Gupta (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061, rolf.mueller@vt.edu), Hongxiao Zhu (Statistics, Virginia Tech, Blacksburg, VA), Mittu Pannala (Mech. Eng., Virginia Tech, Blacksburg, VA), Uzair S. Gillani (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Yanqing Fu (Biomedical Eng. and Mech., Virginia Tech, Blacksburg, VA), Philip Caspers (Mech. Eng., Virginia Tech, Blacksburg, VA), and John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Dartmouth, MA)

Horseshoe bat biosonar is characterized by a pronounced dynamics at the interfaces of ultrasound emission and reception: The bats can alter the shapes of their noseleaves and outer ears (pinnae) through muscular action while they are emitting or receiving biosonar signals. An information-theoretic analysis has been conducted to assess whether these non-rigid baffle motions could affect the ability of

the bats' biosonar system to encode sensory information. This analysis was based on acoustic characterizations (beampatterns) obtained either through numerical predictions or from measurements of physical models. In total, four time-variant baffle shapes, for emission and reception, numerical and physical each, have been analyzed. The similarity of beampattern across different baffle shape conformations was quantified using normalized mutual information that was found to be $<20\%$ for distant stages. Hence, the changes in the baffle shapes were capable of providing different views of a biosonar environment. The information added by these views was found to enhance performance bounds for sonar estimation tasks related to target direction. In particular, it was found that the number of resolvable directions and the accuracy of direction-finding (as measured by the Cramer Rao lower bound) increased when beampatterns from across a sequence of shape conformation were combined.

11:15

1aAB3. Specialized facial muscles support sonar beam-forming by free-tailed bats. Samantha Trent and Michael Smotherman (Biology, Texas A&M Univ., 3258 TAMU, College Station, TX 77843-3258, smotherman@tamu.edu)

Echolocating bats may be able to manipulate the acoustic projection pattern of their sonar pulse emissions, but the mechanism(s) for this and potential constraints are unexplored. The Mexican free-tailed bat (*Tadarida brasiliensis*) appears to achieve this by finely adjusting the shape of its mouth cavity (beam-forming) in a behavior akin to supralaryngeal speech motor control by humans. Flying *Tadarida* raise their nose and lips preceding each echolocation pulse with a hypertrophied set of specialized facial muscles (levator labii complex). We investigated 1) muscle complex activity patterns during sonar performance, 2) whether the muscles tissue displayed necessary fast-twitch specializations to accommodate echolocation, and 3) how manipulations of mouth shape altered 3D beam patterns. First, EMG recordings from awake echolocating bats confirmed the muscles were activated in precise temporal coordination with pulse emissions. Second, we describe the anatomical connectivity of the muscle complex, origin and insertions, and innervation pattern. Histochemical analyses confirmed they were highly aerobic, fast-twitch muscles. Last, direct measures revealed that raising the nose tip alone created a small aperture and wide-angle beam, and simultaneously raising the front and side lips created a wider aperture with narrower beam. These results indicate that *Tadarida* possesses a specialized neuromuscular apparatus for sonar beam-forming.

11:35

1aAB4. Neural representation of sonar emissions and echoes in the auditory midbrain of the echolocating bat, *Eptesicus fuscus*. Michaela Warnecke (Johns Hopkins Univ., Baltimore, MD 21218, warnecke@jhu.edu), Brandon C. Casper (Naval Undersea Warfare Ctr., Newport, RI), Andrea M. Simmons, and James A. Simmons (Brown Univ., Providence, RI)

Local field potentials and multi-unit responses were recorded from the auditory brainstem of the anesthetized big brown bat, *Eptesicus fuscus*, in response to FM sweeps modeled on those the bat would receive from its broadcasts and also their echoes. Tungsten microelectrodes were inserted at an angle into the inferior colliculus (IC), and responses were recorded at progressively deeper sites in the IC and into the area of the cochlear nucleus (CN). Waveforms of the averaged field potentials varied with depth and recording site, with amplitude and latency of the largest peak in each response consistent with transmission time from the cochlea to the CN and then to the IC. For these responses, values of amplitude-latency trading are consistent with those from previous work. Simulated broadcasts followed by sequences of echoes that mimic reflections from complex sonar scenes evoked sequences of distinct responses. Their latencies were affected by amplitude and also by intersound intervals, creating a dynamic pattern of peaks across time. Latencies tracked the shortening of delay that simulates movement through the scene, providing a data set that can be used as steering inputs to a guidance model. [Work supported by ONR, Capita Foundation.]

Session 1aAO

Acoustical Oceanography: Topics in Acoustical Oceanography I

Darrell Jackson, Cochair
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Carolyn Binder, Cochair
Oceanography Department, Dalhousie University, LSC Ocean Wing, 1355 Oxford St., PO Box 15000, Halifax, NS B3H 4R2, Canada

Contributed Papers

10:30

1aAO1. Impacts of environment-dependent acoustic propagation on passive acoustic monitoring of cetaceans. Carolyn Binder (Defence Res. & Development Canada, LSC Ocean Wing, 1355 Oxford St., PO Box 15000, Halifax, NS B3H 4R2, Canada, carolyn.binder@dal.ca) and Paul C. Hines (Dept. of Elec. and Comput. Eng., Dalhousie Univ., Halifax, NS, Canada)

Significant effort has been made over the last few decades to develop automated passive acoustic monitoring (PAM) systems capable of classifying cetaceans at the species level; however, these systems often require tuning when deployed in different environments. Anecdotal evidence suggests that this requirement to adjust a PAM system's parameters is partially due to differences in the propagation characteristics. The environment-dependent propagation characteristics create variation in how a cetacean vocalization is distorted after it is emitted. If this is not accounted for, it could reduce the accuracy and precision of automated PAM systems. An aural classifier developed at Defence R&D Canada (DRDC) has successfully been used for inter-species discrimination of cetaceans. It achieves accurate results by using perceptual signal features that model the features employed by the human auditory system. To quantify the impacts of propagation on the perceptual features, an experiment was conducted in which bowhead and humpback whale vocalizations were transmitted over 1-20 km ranges during a two-day sea trial in the Gulf of Mexico. The experimental results will be presented and strategies will be discussed for training the classifier so that it is capable of operating in numerous acoustic environments with minimal adjustment of the classifier's parameters.

10:45

1aAO2. Measurements of the acoustic properties of crude oil. Scott Loranger (Earth Sci., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 01950, scottloranger@gmail.com), Christopher Bassett (Alaska Fisheries Sci. Ctr., National Marine Fisheries Service, National Oceanic and Atmospheric Administration, Seattle, WA), and Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

The acoustic response of crude oil is dependent on the density and sound speed contrast between the oil and the surrounding media. To detect and quantify crude oil in the marine environment, it is crucial to have accurate measures of the density and sound speed at oceanographically relevant temperatures and pressures. A meta-analysis of currently available sound speed data has found a paucity of sound speed data for crude oil at relevant temperatures (<298 K) and pressures (0.1-30 MPa). Typical sound speed measurements are confined to reservoir temperatures (>300K) and pressures (>100 MPa). In order to evaluate the use of currently available data, time-of-flight measurements in a few different types of crude oil were performed in a purpose-built chamber that simulated oceanographic conditions. These test data were compared with extrapolated sound speed curves from currently available data.

11:00

1aAO3. Efficient sequential Monte Carlo estimation of range-dependent seabed properties. Eric Mandolesi (School of Earth and Ocean Sci., UVIC, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, emand@uvic.ca), Jan Dettmer (GeoSci. Dept., Univ. of Calgary, Calgary, AB, Canada), Stan E. Dosso (School of Earth and Ocean Sci., UVIC, Victoria, BC, Canada), Charles W. Holland, and Sheri Martinelli (Appl. Res. Lab., The Penn State Univ., State College, PA)

Seabed geoacoustic properties play a crucial role in shallow-water sonar applications, including the detection of unexploded ordnance. Our goal is improved efficiency of Bayesian seabed parameter and uncertainty estimation for large data volumes. While Bayesian uncertainty estimation provides important information for sonar applications, the approach is computationally expensive which limits utility for large surveys, where an abundance of range-dependent data can be collected. This work considers the efficiency of a particle filter to quantify information content of multiple data sets along the survey track by considering results from previous data along the track to inform the importance sampling at the current point. Efficiency is improved by tempering the likelihood function of particle subsets and including exchange moves (parallel tempering), and by adapting the proposal distribution for the Markov-chain steps. In particular, perturbations are proposed in principal-component space, with the rotation matrix computed via eigenvector decomposition of the unit-lag parameter covariance matrix. The algorithm is applied to 350 data sets collected along a 13-km track on the Malta Plateau, Central Mediterranean Sea. Improved efficiency from parallel tempering and principal-component proposal densities are studied. [Work supported by the Strategic Environmental Research and Development Program, U.S. Department of Defence.]

11:15

1aAO4. Toward estimating ocean temperature through passive acoustics. Ana Bela Santos, Paulo Felisberto, and Sérgio M. Jesus (LARSyS, Univ. of Algarve, Campus de Gambelas, Faro P8005-139 Faro, Portugal, Portugal, absantos@ualg.pt)

In this paper, we analyze the noise recorded in two drifting vertical line arrays (VLAs) deployed 1 km apart, in the area of the Setubal's underwater canyon off the west coast of Portugal during RADAR'07 sea trial. Automatic Information System (AIS) recordings for this period reveal a major tanker passing by the area, whose acoustic signature is clearly seen in the spectrogram. The influence of the underwater canyon in the recorded spectrograms is discussed by simulations. By cross-correlating and spatially filtering (beamforming) the signals received at the VLAs, a wave front structure is obtained with coherent information from propagating paths traversing both VLAs and respective arrival times. This work is a contribution for application in a passive ocean acoustics framework to the estimation of sound speed perturbations in the water column.

11:30

1aAO5. Computing where ocean fluctuations affect the acoustic impulse response. John Spiesberger (Earth and Environ. Sci., Univ. of Pennsylvania, 240 S. 33rd St., Philadelphia, PA 19104-6316, johnsr@sas.upenn.edu) and Dmitry Mikhin (Acacia Res. Pty Ltd., Hendon, SA, Australia)

We compute accurate maps of oceanic fluctuations affecting transient acoustic signals propagating between a source and receiver. The technological advance involves coupling the One-Way Wave Equation (OWWE) propagation model with the theory for the Differential Measure of Influence (DMI). This theory uses two finite-frequency solutions of the linear acoustic wave equation: from source to receiver and vice versa. They must obey reciprocity. OWWE satisfies reciprocity with great accuracy even at basin-scales when the field of sound speed varies horizontally and vertically. At infinite frequency, maps of the DMI collapse into infinitely thin rays. Mapping the DMI is useful for understanding measurements of the amplitude and travel time changes of sound at finite frequencies. [Work supported by ONR contracts N00014-12-C0230 and N62909-12-1-7033.]

11:45

1aAO6. Applying double-difference methods for fine-scale acoustic tracking using a short aperture vertical array. Ludovic Tenorio-Hallé (ludovictenorio@gmail.com), Aaron M. Thode, Jit Sarkar, Chris Verlinden, Jeffrey Tippmann, William S. Hodgkiss, and William A. Kuperman (Scripps Inst. of Oceanogr., 1044 Loring St., San Diego, CA)

Ray tracing can estimate an acoustic source's depth and range in a waveguide by exploiting multipath arrival information on a vertical array. However, environmental mismatch in the model or array tilt can yield highly scattered trajectories when ray tracing multiple events. "Double-difference" methods have been used to localize earthquakes (Waldhauser and Ellsworth, 2000) and fin whales (Wilcock, 2012) by determining the relative locations of multiple events, rather than their absolute positions. This approach, which exploits changes in relative travel times between events, has recently been reformulated to recover the dive trajectory of a source using a single multi-hydrophone vertical array, whenever three acoustic rays are available for each event. Here, the method is expanded such that changes in ray elevation angles between events can be used to reduce the number of rays required. This technique is tested on data recorded on a short aperture vertical array off the coast of Southern California in 4 km deep water. Trajectories from both a controlled towed source and sperm whale dives are examined. [Work supported by Office of Naval Research—Marine Mammals and Biology and Ocean Acoustics Program.]

MONDAY MORNING, 28 NOVEMBER 2016 — CORAL 1, 10:30 A.M. TO 11:50 A.M.

Session 1aBA

Biomedical Acoustics and Physical Acoustics: Photoacoustics: Light and Sound I

Kang Kim, Cochair
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Parag V. Chitnis, Cochair
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Yoshifumi Saijo, Cochair
Tohoku University, 4-1 Seiryomachi, Aoba-ku, Sendai 980-8575, Japan

Invited Papers

10:30

1aBA1. Real-time three-dimensional visualization of blood vessels and its condition, based on photoacoustic technology. Takayuki Yagi (ImPACT Program, Japan Sci. and Technol. Agency, K's Gobancho, 7 Gobancho, Chiyoda-ku, Tokyo 102-0076, Japan, takayuki.yagi@jst.go.jp)

In 2014, the Japanese Government created the new innovation funding program, called "ImPACT; Impulsing PAradigm Change through disruptive Technologies" and established in the Council for Science, Technology and Innovation (CSTI), the Cabinet Office. ImPACT consists of 16 programs and its period is for 5 years (JFY2014-2018). One of ImPACT is this program; Innovative Visualization Technology to Lead to Creation of a New Growth Industry. The goal of this program is to develop the technologies for real-time 3D photoacoustic imaging of blood vessels in the human body and the physical properties of substances, and then demonstrate the value of applications in medical, healthcare, and cosmetic fields. The configuration of the program consists of six projects: "Visualization and measurement technology," "Tunable laser technology," "Ultrasound sensor technology," "Wide-field visualization system" and "Micro-visualization system" for the achievement of real-time 3D visualization, and "Demonstration of value"for demonstrating the value of applications. Six research institutions and six companies will work together for a common goal. I will present the targets and strategies of the program and show recent research results.

10:50

1aBA2. Ultrasound imaging of extravascular molecular targets with light and optically triggered nanoscale contrast agents. Stanislav Emelianov (School of Elec. and Comput. Eng. and Dept. of Biomedical Eng., Georgia Inst. of Technol., 777 Atlanti Dr., Atlanta, GA 30332-0250, stas@gatech.edu)

To obtain high sensitivity, specificity, and signal/contrast-to-noise ratio imaging of disease, detection of biomarkers or biochemical and cellular processes is often needed. Typically, such information is gathered using imaging contrast agents. Unfortunately, microbubbles—most common ultrasound contrast agent, are too large to escape vascular compartments. Thus, contrast-enhanced molecular ultrasound imaging is often limited to molecular markers on the vascular endothelium and other intravascular targets. Recently, we introduced several high-resolution, high-sensitivity, depth-resolved ultrasound-based molecular imaging techniques augmented with nanometer scale imaging contrast agents capable of visualizing molecular signatures of cells in context of structural and functional properties of tissue. In this presentation, several examples of molecular ultrasound imaging based on synergistic combination of light and sound will be given. Specifically, clinical problems including detection and phenotyping of primary tumor, assessment of micrometastatic lesions in sentinel lymph node, and image-guided cancer cell therapy will be described. Design and synthesis of clinically relevant contrast nanoagents with properties desired for cellular/molecular ultrasound imaging will be discussed. Finally, the presentation will conclude with the analysis of how molecular ultrasound can change both fundamental medical science and the utility of clinical ultrasound imaging in diagnostic and therapeutic applications.

11:10

1aBA3. Photoacoustic imaging for ultrasound tissue characterization and treatment monitoring. Michael C. Kolios (Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B2K3, Canada, mkolios@ryerson.ca)

Photoacoustic imaging relies on generation of ultrasound waves from optically absorbing structures. The ultrasound produced in photoacoustic imaging can be analyzed by methods developed to analyze ultrasound backscatter signals for ultrasound tissue characterization, but the interpretation of the analysis is based on the physics of photoacoustic wave generation. In the absence of exogenous absorbers, blood is one of the dominant optically absorbing tissues. Hemoglobin in red blood cells is the main endogenous chromophore in blood. The spatial distribution of red blood cells in tissue determines the spectral characteristics of the ultrasound signals produced. We are interested in cancer treatment monitoring. Tumor blood vessels have distinct organizational structure compared to normal blood vessels: normal vessel networks are hierarchically organized, with vessels that are evenly distributed to ensure adequate oxygen and nutrient delivery. Tumor vessels are structurally different: they are torturous and hyperpermeable. Therapies that target the vasculature can induce changes in the vascular networks that in principle should be detected using photoacoustic imaging. In this work, we will show how the frequency content of the photoacoustic signals encodes information about the size, concentration, and spatial distribution of non-resolvable blood vessels that can be used to assess treatment response.

11:30

1aBA4. Dual-wavelength photoacoustic microscopy for cell tracking. Pai-Chi Li (National Taiwan Univ., Dept. of Elec. Eng., National Taiwan Univ., NTU, No.1, Sec. 4, Roosevelt Rd., Taipei 106, Taiwan, paichi@ntu.edu.tw)

Understanding cell interactions can help to develop cell-based therapeutic strategies such as immunotherapy. The ability to track cells thus becomes an important task. 3D *in vitro* models have become a promising platform for such cell studies. However, conventional optical imaging methods are no longer adequate in 3D models due to limited penetration. To overcome this problem, we developed a dual-wavelength optical-resolution photoacoustic microscope (2L-OR-PAM) to study CD8 + cytotoxic T lymphocytes (CTLs) trafficking in a 3D tumor microenvironment. With the aid of molecular probes, the 2L-OR-PAM system can track both the CTLs and the tumor cells. An Nd:YLF laser was used as a pump laser at 523nm passing into a dye laser to generate near-infrared light at 813-nm. Two types of AuNPs were used to target the cancer cells and the CTLs. An x-y galvanometer scanner was employed to obtain a high scanning rate. The lateral and the axial resolutions were measured at 1.6 μm and 5 μm, respectively. We successfully showed that the 2L-OR-PAM can map the distribution of CTLs and 3D structures of Hepta1-6 tumor spheres. The 2L-OR-PAM can provide cellular and molecular information and it has the potential for preclinical screening requiring cell tracking.

Session 1aED

Education in Acoustics: Attractive Educational Methods and Tools in Acoustics I

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

Invited Papers

10:30

1aED1. Structuring an introductory acoustics course to be a vehicle for improving pre-calculus math skills and recruiting students to technical careers. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

"Why do I have to take this math course—when will I ever use this stuff?" and "I could never be a scientist—I'm terrible at math." These statements are familiar to educators the world over, much to our frustration... and to the detriment of society's goals of being scientifically and mathematically literate and having a large and diverse population of STEM professionals. The present paper describes elements of a general education course in acoustics at Central Washington University ("Physics of Musical Sound") that serve vital secondary functions of improving math skills and fostering an appreciation of acoustics-related careers, appropriate for the typical demographic of students from diverse academic and social backgrounds, many of whom have not declared a major, that are attracted to the interdisciplinary nature of the course. These secondary goals, as well as the subject matter, are also highly appropriate for high school students. The content, the skills, and the applications are closely linked, leveraging students' native interest in, and familiarity with, music and sound to provide the incentive to develop and hone math concepts. Specific content, applications, and pedagogical approaches will be discussed.

10:50

1aED2. Using wine glasses, strings, and bars to teach resonance at the undergraduate and graduate levels. Brian E. Anderson (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, bea@byu.edu)

At Brigham Young University, two of the acoustics courses taught in the Physics and Astronomy Department focus on resonance topics. One of the undergraduate courses is a general education, descriptive acoustics course on musical instruments, speech, hearing, and audio. One of the graduate level courses using a differential equations approach to wave motion and resonances on strings, bars, membranes, and plates. In both of these courses, taught at very different levels, the concept of resonance is explored and analyzed. This presentation will discuss various demonstrations used in these classrooms and discuss how they are used in the learning development of the students. Demonstrations include wine glasses, strings, and bars.

11:10

1aED3. Tools for education in acoustics and phonetics. Takayuki Arai (Dept. of Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp)

We have developed a series of tools for education in acoustics and phonetics. Some of them are in electronic form, while some are physical models. A set of demonstrations and explanations in acoustic phonetics is now available as "Acoustic-Phonetics Demonstrations (APD)" through our WebSite: www.splab.net/APD/. We are adding content and have made YouTube video clips using physical models available on our WebPages. In particular, we use the physical models of the human vocal tract to demonstrate acoustic phenomena and theories. They are useful tools for explaining acoustics in speech production, such as the source-filter theory and the relationship between vocal-tract configurations and vowel quality. Furthermore, the models are effective when teaching articulation/pronunciation of native and non-native languages. Due to the high demand for vocal-tract models of various sounds in multiple languages, we are expanding the range of sounds to include more than Japanese sounds, but also additional vowels and consonants in different languages, including English. We also discuss how to apply three-dimensional printing.

11:30

1aED4. A project-oriented graduate course on ultrasonics. Michael R. Haberman (Appl. Res. Labs. & Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

The study of ultrasonics encompasses a wide range of topics in acoustics. Some topics, like dispersion in elastic waveguides, require mathematical sophistication, while others, like transducer selection and common signal processing techniques, demand detailed technical knowledge. The variety of topics germane to the subject of ultrasonics presents an instructor with a number of challenges that must be addressed through the course design. This talk reports on the current graduate course on ultrasonics in the graduate acoustics program at The University of Texas at Austin. The structure of the course is inspired by the experiential learning model, which emphasizes experimentation and experiences to supplement in-class lectures. This is achieved using numerous in-class demonstrations and an extended team project. Whenever possible, data from in-class demonstrations are used in homework assignments to provide continuity between theory and practice. Specific example course projects will be discussed which illustrate the breadth and depth of course projects resulting from this approach, and perspectives are provided for avenues for improvement using this teaching method for instruction in ultrasonics.

Contributed Paper

11:50

1aED5. Comparison of acoustic education in the architecture departments between Turkey and different universities in the world. Filiz B. Kocyigit (Architecture, Acoust. Society of America, Atilim Univ., Faculty of Fine Art Design and Architecture, Kizilcasar Mah. Incek, Ankara 06830, Turkey, filizbk@gmail.com) and Sevgi Lokce (Architecture, TMMOB, Ankara, Turkey)

The aim of this study is to determine the "when and what extent" do architectural acoustics is included in architecture curricula. Additionally, to find out the state of research facilities across academic institutions between Turkey and different countries including Architectural Acoustic. In Turkey, in many architecture programs do not include "Experiences of the Acoustics Laboratory or courses" and also education is not among the objectives of architectural acoustics for their future. On comparing of teaching activities, education objective is to allow professional figures to interpret, define, and govern the processes of maintenance and management with high quality checking in the life cycle of the built systems are researched. It seems that searching a new formation is being established in order to integrate with the EU standards such as European Credit Transfer and Accumulation system (ECTS) and many of the schools have revised their programs according to these new standards. As a consequence of this revision, duration and content of the courses are being re-handled. Curriculum of Architecture, course Group, Percentage of course are compared.

MONDAY MORNING, 28 NOVEMBER 2016 SOUTH PACIFIC 3, 10:30 A.M. TO 12:00 NOON

Session 1aNS

Noise: Sound Design I

André Fiebig, Cochair
HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

Takeshi Toi, Cochair

Patricia Davies, Cochair
Ray W. Herrick Labs., School of Mechanical Engineering, Purdue University, 177 South Russell Street, West Lafayette, IN 47907-2099

Chair's Introduction—10:30

Invited Papers

10:35

1aNS1. Sharpness perception and modeling of stationary and time-varying sounds. Roland Sottek (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

Psychoacoustic parameters like loudness and sharpness can be used for sound quality assessment and sound design applications. While loudness of stationary sounds has been standardized for decades, standards for sharpness of stationary sounds and loudness of time-varying sounds have only been published since 2009 (DIN 45692:2009-08) and 2010 (DIN 45631/A1:2010-03), respectively. Present calculation methods of sharpness consider the "center of gravity" of the weighted specific loudness patterns as a measure for sharpness; thus, their results do not depend on loudness. Only the Aures model includes, to a small extent, the influence of loudness on sharpness by an additional loudness-dependent factor. Until now, mostly stationary synthetic sounds (filtered noise and pure tones) have been used for the derivation of sharpness models. However, when these calculation procedures are applied to generally time-varying technical sounds, the results deviate significantly from the listening test results. This study describes new listening tests performed to investigate systematically the influence of loudness as well as temporal structures on sharpness. The findings from the listening tests should serve as a basis for an improved model of sharpness perception for stationary and time-varying sounds.

10:55

1aNS2. The role of cross-validation approaches in sound quality metric development. André Fiebig and Fabian Kamp (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

The sound of a product is an important feature influencing the perceived quality impression. Thus, product sound is deliberately designed to stand out against competitors. To determine meaningful design targets guaranteeing a perceived sound quality, jury tests are performed and sound quality metrics are developed. Ideally, by means of a developed sound quality metric, the prediction of perceived sound quality of comparable products is possible without the need for further jury testing. However, several aspects influence the reliability and validity of jurors' ratings deteriorating the prediction accuracy of the derived sound quality metric. Moreover, if parameters of a sound quality metric are only estimated in a way to give a best fit with the rating data, the external validity of the metric is possibly reduced. In order to estimate the robustness of a sound quality metric and to avoid overfitting, cross-validation methods and different error measures should be considered. The paper will illustrate the impact of missing robustness analyses on the validity of sound quality metrics and how systematic approaches can be applied to improve the process of metric development. Moreover, general limitations and risks of sound quality metrics based on jury tests will be discussed.

Contributed Papers

11:15

1aNS3. Consideration of radiated sound quality for high-grade washing machine through paired comparison test. Junji Yoshida, Rei Yamashita, Shota Yoshida (Mech. Eng., Osaka Inst. of Technol., 5-16-1 Omiya, Asahi-Ku, Osaka 535-8585, Japan, yoshida@med.oit.ac.jp), Tomohiro Fujii, and Akihiro Hosokawa (Panasonic, Kusatsu, Japan)

In this study, we conducted a sound quality evaluation test to obtain a guide realizing a radiated noise suitable for high-grade washing machines. Twelve sounds at spin drying of drum type washing machines were prepared for paired comparison experiments. Forty eight participants evaluated the presented sound quality about "noisiness," "discomfort," and "luxuriousness." The results indicated that noisiness and discomfort had very high correlation between each other. In addition, loudness values calculated from the recorded sound pressure signal had very high linear relationship with them. On the other hand, evaluation result of luxuriousness showed a different tendency. The relationship between the calculated loudness value and luxuriousness score was high; however, the relationship was not linear but hyperbolic curve. When the loudness of the washing machine was over 10 sone, the luxury feeling of the radiated sound was rarely improved in spite of the loudness reduction. On the contrary, the luxuriousness was found to be expected to increase largely by the noise reduction when the loudness was less than 10 sone. Consequently, decreasing the loudness less than 10 sone was found to be important in any cases to realize a noise suitable for high-grade washing machine.

11:30

1aNS4. Musical mind control: Acoustic convergence to background music in speech production. Ryan G. Podlubny (Linguist, Univ. of Canterbury, 214, Locke Bldg., Christchurch 8011, New Zealand, ryan.podlubny@pg.canterbury.ac.nz)

Talkers adjust their speech to resemble that of their interlocutors—a phenomenon known as speech *convergence*. Broadly defined, convergence describes automatic synchronization to some external source, and this effect has been recognized in various linguistic domains (e.g., syntactic structuring, morphological choices, and vowel production). Using a speech-in-noise reading-task, this paper explores whether or not speech production also converges with non-linguistic signals: Specifically, might a speaker's *rhythm*, *pitch* or *intensity* be influenced by fluctuations in background music? Participants read passages aloud while music was presented at ~45 dB(A) via headphones. Music was composed with relatively flat envelopes in the domains of interest, and in test-conditions only Pitch, Tempo, or Intensity was altered. Manipulation was gradual (linear function) and would similarly return to the point of origin after reaching a target (Pitch = -200 cents; Amp = +6dB; Tempo = -20 beats per minute). Control conditions (i.e., no acoustic manipulation) broke up test conditions. Analyzing data from 25 English speakers identifies significant convergence effects in the Pitch condition (tempo and intensity analyses have not yet been conducted). Using Generalized Additive Mixed Models, the predicted Time by Condition interactions were observed, suggesting that environmental noise may constantly be influencing speech production.

11:45

1aNS5. Controlling smartphone vibration and noise. Inman Jang, Taeyoung Park, Donghyun Kim, Sangbeom Woo, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, kgpbjim@yonsei.ac.kr), and Heungkil Park (Samsung Electro Mech. Co.,Ltd, Seoul, Korea (the Republic of))

Vibration of the internal electronics components and its transmission to the casing has been known as a source of noise radiation that degrades the call quality of a smartphone. Here, we expand upon our previous work [J. Acoust. Soc. Am. **137**, 2293 (2015)] that employed the nearfield acoustic holography (NAH) for characterization of the smartphone radiated noise. The NAH, when used in conjunction with a vibration model of the smartphone, is shown to be a powerful tool for controlling the smartphone vibration and noise at the initial design phase. The vibration model is largely based on the classical plate theory and yields concrete guidelines on how to reduce vibration and the subsequent noise. The effectiveness of the technique is demonstrated using some of the smartphones currently available on the market.

Session 1aPA

Physical Acoustics: Thermoacoustics

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

Contributed Papers

10:30

1aPA1. On-off intermittency in coupled chaotic thermoacoustic systems. Rémi Delage, Yusuke Takayama, and Tetsushi Biwa (Mech. Eng., Adv. Mech. Systems and Design, Graduate School of Eng., Tohoku Univ. 6-6-01, Aoba, Aramaki, Aoba-ku, Sendai 980-8579, Japan, delageremi@hotmail.fr)

We present chaos-chaos synchronization in coupled thermally induced acoustic gas oscillation systems. By introducing a local cross-sectional change in a gas-filled tube subjected to a temperature gradient, chaotic oscillations are found to be achieved through the mode competition. The chaos-chaos synchronization is then obtained by connecting the two thermoacoustic oscillators via a rigid plate with an orifice. Theoretical statistical scaling laws regarding the laminar phases, spectral density, and amplitude probability distribution are found to be satisfied in the coupled thermoacoustic oscillators, as presented from the analysis of pressure fluctuations when the thermoacoustic chaos-chaos synchronization breaks down with the decreased orifice size.

10:45

1aPA2. Development of a prototype thermoacoustic refrigerator with coaxial geometry. Ayumu Kakuda (Faculty of Sci. and Eng., Doshisha Univ., 1-3 Tatara-miyakodani, Kyotanabe-city, Kyoto, 610-0321 Japan, dup0318@mail4.doshisha.ac.jp), Shin-ichi Sakamoto (Univ. of Shiga Prefecture, Hikone, Shiga, Japan), and Yoshiaki Watanabe (Doshisha Univ., Kyotanabe, Kyoto, Japan)

The thermoacoustic system with coaxial geometry (coaxial-type) is discussed. The coaxial-type has high efficiency because it performs the energy conversion with the traveling wave mode as the same of the looped tube type thermoacoustic system (loop-tube-type). Additionally, the coaxial-type can be realized more downsizing in comparison with the loop-tube-type, and it is also expected to expand the practical use. In this report, the experimental studies of the coaxial-type refrigerator were carried out. A pair of stacks was placed in the experimental setup. Those stacks were employed as a prime mover and a heat pump. In the case of the loop-tube-type, those are usually placed symmetrically. However, it is not practical for the coaxial-type to place stacks symmetrically because of the difficulties for the taking off cold energy; therefore, both of stacks were placed in annular path. As a result, temperature decreases were confirmed, that is, amount of 5 decreases from room temperature were observed at the cooling point of a heat pump. It is thought that the energy conversion with the traveling wave mode is occurring in stacks. These results show that the coaxial-type can be possible to become a new design to take the place of the loop-tube-type.

11:00

1aPA3. Increasing of work flow on the thermoacoustic system by control of the temperature gradient within the stack. Seiya Fukuda (Elec. Eng. and Electronics, Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe, Kyoto 610-0321, Japan, dup0305@mail4.doshisha.ac.jp), Shin-ichi Sakamoto (Univ. of Shiga Prefectural, Hikone, Shiga, Japan), and Yosiaki Watanabe (Doshisha Univ., Kyotanabe, Japan)

A gas column in a tube with a temperature-gradient spontaneously begins to oscillate when air turbulence occurs. The work flow produced in the stack, ΔI, is important for the thermoacoustic oscillation. The amount of ΔI is determined by the shape of temperature-gradient. Experimental studies on thermoacoustic systems are previously assessed by measurement of temperature ratio, T_H/T_C, of the stack; however, investigation for distribution of temperature within the stack is less discussed. In order to examine the influence of a temperature-gradient within the stack on the thermoacoustic oscillation, we propose that control method for the temperature-gradient under the same T_H/T_C. In particular, the control of the temperature distribution is implemented in actively changing the internal temperature by additional nichrome heater in middle of the stack. The sound pressure in the standing-wave thermoacoustic system is measured at several temperature distribution by the proposed method. Furthermore, ΔI was calculated based on the measured sound pressure. As a result, the work flow by the control of a temperature distribution have increased, and it was qualitatively estimated by the slope of heat flux within the stack from temperature distribution.

11:15

1aPA4. Amplitude death in delay-coupled thermoacoustic oscillators. Hiroaki Hyodo, Itaru Shinkai, and Tetsushi Biwa (Dept. of Mech. System and Design, Tohoku Univ., 6-6-04 Aoba, Aramaki, Aobaku, Sendai 980-8579, Japan, hyodo@amsd.mech.tohoku.ac.jp)

A phenomenon of thermoacoustic oscillation is seen as an undesirable vibration in combustion engineering as it can cause serious damage to the combustor. Lots of methods such as controlling a secondary fuel supply and addition of Helmholtz resonators are proposed to suppress the oscillations, but they often become complicated. As a simple and reliable method, we try to make use of mutual interaction of the coupled thermoacoustic oscillators to stop oscillations completely. It has been demonstrated in the system of coupled electrical circuits that "amplitude death," meaning the stabilization of zero amplitude states, is induced by time-delay coupling even when the oscillators have the same frequency. In this study, we introduce the time-delay coupling in thermoacoustic oscillators by using hollow tubes. In the tube coupling, the sound propagation time through the tube corresponds to the delay time, while the tube diameter controls the coupling strength. The condition of amplitude death is experimentally found for various tube lengths, diameters, and number of tubes. Thus, the tube coupling provides the way to annihilate thermoacoustic oscillations. The results are compared with those of linear stability analysis based on basic equations of hydrodynamics. Both results are qualitatively consistent with each other.

11:30

1aPA5. Thermoacoustics of a T-burner: How to better estimate the combustion response function from a T-burner experiment. Taeyoung Park, Hunki Lee, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., Eng. Bldg. A391, 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, pty0948@yonsei.ac.kr), and Dohyung Lee (Agency for Defense Development, Seoul, Korea (the Republic of))

The combustion response function, representing the coupling between the pressure oscillation and the combustion rate of a solid propellant, is a key indicator of combustion instability (large unwanted pressure oscillations in a combustion chamber) of a solid rocket motor. To obtain the response function experimentally, a T-burner is often used, in which the lowest longitudinal acoustic resonance of the cylindrical enclosure is excited by the combustion of propellant samples (acting like speakers) at both ends. The existing formula used to derive the combustion response function from a T-burner experiment is based on a series of rather crude assumptions. For example, acoustically rigid boundaries are assumed at both ends, whereas impedance boundary conditions may be more appropriate to reflect the introduction of combustion products from the propellant. In this talk, we propose a few modifications to the current T-burner analysis regime, which could improve the accuracy of the measured response function. The new method is demonstrated using an actual T-burner test. [This work was supported by Defense Acquisition Program Administration and Agency for Defense Development under the contract UD140024GD.]

11:45

1aPA6. Numerical study of heat transfer in parallel-plate heat exchangers in thermoacoustic engine. Kazuto Kuzuu and Shinya Hasegawa (Tokai Univ., 4-1-1 Kitakaname, Hiratsuka, Kanagawa 2591292, Japan, kuzuu@tokai-u.jp)

For designing thermoacoustic devices, heat transfer problem of heat exchangers is a key point and is not solved completely yet. In the present study, both computational fluid dynamics (CFD) simulations and numerical analysis based on linear thermoacoustic theory are performed, and heat transfer characteristics, heat fluxes, and Nusselt numbers are investigated from both numerical results. Especially, through parametric studies by linear analysis, the effect of displacement amplitude and thermal penetration depth on heat transfer is discussed. In this simulation, a regenerator is located between hot and cold heat exchangers, and the effect of each channel size is also investigated. Furthermore, simulated results are compared with the conventional heat transfer models for oscillatory flow, and the verifications of those models or simulations are executed. On the other hand, by comparing the CFD results with linear analysis, non-linearity of heat transfer included in thermoacoustic phenomena is also discussed. For this discussion, unsteady behavior of temperature field in heat exchangers is observed in the CFD simulation. Finally, by combining the parametric studies of linear analysis and non-linear effect obtained from the CFD simulation, the heat transfer characteristics for heat exchangers in thermoacoustic engine are comprehensively discussed.

MONDAY MORNING, 28 NOVEMBER 2016 — CORAL 2, 10:30 A.M. TO 11:50 A.M.

Session 1aPP

Psychological and Physiological Acoustics: Spatial Hearing and Its Applications I

Kazuhiro Iida, Cochair
Chiba Institute of Technology, 2-17-1, Tsudanuma, Narashino 275-0016, Japan

Griffin D. Romigh, Cochair
Air Force Research Labs, 2610 Seventh Street, Area B, Building 441, Wright Patterson AFB, OH 45433

Yôiti Suzuki, Cochair
Research Inst. Electrical Communication, Tohoku University, Katahira 2-1-1, Aoba-ku, Sendai 981-0942, Japan

Douglas Brungart, Cochair
Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889

Yukio Iwaya, Cochair
Faculty of Engineering, Tohoku Gakuin University, 1-13-1 Chuo, Tagajo 985-8537, Japan

Invited Papers

10:30

1aPP1. Comparison of localization performance with individualized and non-individualized head-related transfer functions for dynamic listeners. Ravish Mehra, Aaron Nicholls (Res., Oculus, 8747 148th Ave. NE, Redmond, WA 98052, ravish.mehra@oculus.com), Durand Begault (NASA Ames, Moffett Field, CA), and Marina Zannoli (Res., Oculus, Redmond, WA)

The head-related transfer function (HRTF) depends on head, torso, and ear geometry of an individual and typically measured in an anechoic chamber using an HRTF measurement apparatus. This is referred to as the *individualized* HRTF. In many applications, an HRTF corresponding to an average human head is measured on dummy also called *generic*HRTFs. Prior research has shown that

personalized HRTF gives significantly better performance over the generic HRTF for a static listener. However, these studies were limited to only static listeners where the listener's head is fixed during stimuli presentation. In case of dynamic listeners, the head motion enables the listeners to perceive additional spectral cues resulting due to head motion. In the present study, we compare the performance of personalized vs. generic HRTFs for a dynamic listener. We quantify the localization performance of each listener with both individualized and non-individualized HRTFs (virtual conditions) and compare it against their ground truth localization performance. Our data reveals the benefits of personalized HRTF over the generic HRTF for dynamic listeners in virtual environments.

10:50

1aPP2. A dipole model for estimating frequency versus elevation-angle trajectory of the first spectral notch of head-related transfer functions in the median plane. Hironori Takemoto, Parham Mokhtari, Hiroaki Kato, Ryouichi Nishimura (National Inst. of Information and Communications Technol., 3-5 Hikaridai, Seika-cho, Soraku-gun, Kyoto 619-0289, Japan, hironori.takemoto@p.chibakoudai.jp), and Kazuhiro Iida (Chiba Inst. of Technol., Narashino, Japan)

The first (lowest) spectral notch (N1) of head-related transfer functions is known as a cue for sound localization in the median plane. This may be due to the fact that N1 frequency gradually increases as the sound source approaches the direction above the head. The mechanism of this phenomenon, however, is still unclear. To clarify the mechanism, using finite-difference time-domain simulation, the normalized pressure amplitude patterns were visualized around the head for 12 subjects when the analysis field was excited at various frequencies from a point just outside the blocked meatus. As a result, strong radiations with different phases occurred in the upper and lower directions, while in the intermediate narrow zone, the pressure amplitude became minimum. This zone corresponded to N1 and bent upward with increasing excitation frequency. These facts implied that the two radiated sound waves canceled each other in the zone, where the path difference amounted to a phase difference of half a wavelength. Thus, the shape of the intermediate zone could be modeled by a hyperboloid whose foci were two hypothetical points on the pinna. The model could roughly estimate, for each individual, the frequency versus elevation-angle trajectory of N1 in the median plane.

11:10

1aPP3. Improved understanding of head-related transfer functions through sagittal-plane spectral decomposition. Griffin D. Romigh (Air Force Res. Labs, 2610 Seventh St., Area B, Bldg. 441, Wright Patterson AFB, OH 45433, griffin.romigh@us.af.mil), Douglas S. Brungart (Walter Reed NMMC, Bethesda, MD), and Brian D. Simpson (Air Force Res. Labs, WPAFB, OH)

Head-related transfer functions (HRTFs) capture the listener-specific modifications that are imparted on a sound as it travels from a specific location in space, interacts with the listener's head, shoulders, and outer ears, and arrives at the ear drums. HRTFs have often been decomposed into a combination of two parts to aid modeling and analysis: a diffuse-field transfer function (DFTF) that captures non-directional but still listener-specific spectral details of the HRTF, and a directional transfer function (DTF) that captures all of the listener-specific directional information. Because the DTF contains spectral variation along all spatial dimensions, it does not provide a way to separate the spectral cues responsible for lateral and sagittal plane localization judgments, which are largely independent. As an alternative approach, decomposing an HRTF along each sagittal plane (aka along the "cones-of-confusion") provides similar benefits to DTF analysis while providing additional insights into the spectral cues responsible for lateral and vertical localization, independently. This talk will provide a detailed overview of sagittal-plane spectral decomposition along with experimental results showing how the technique can be applied to improve localization with non-individualized HRTFs, separate general and idiosyncratic HRTF features, and cluster listeners' HRTFs based on spectral similarity.

11:30

1aPP4. Roles of spectral peaks and notches in the head-related transfer functions in the upper median plane for vertical localization. Kazuhiro Iida and Yohji Ishii (Chiba Inst. of Technol., 2-17-1, Tsudanuma, Narashino 275-0016, Japan, kazuhiro.iida@it-chiba.ac.jp)

The parametric head-related transfer function (HRTF) recomposed of only a spectral peak (P1) generated by the first resonance of the pinna and two spectral notches (N1 and N2) generated by the first and the second anti-resonances of the pinna has been reported to provide approximately the same localization performance as the measured HRTF for the front and rear directions. However, for the upper directions, the localization performance for some of the subjects decreased. In the present study, we carried out two localization tests with seven target vertical angles in the upper median plane to clarify whether adding a spectral peak (P2) generated by the second resonance of the pinna can resolve this performance decrease. The results suggested the following: (1) N1, N2, and P1 play an important role in vertical localization; (2) localization performance was improved by adding P2 to N1N2P1 at the upper directions; (3) however, a sound image was hardly perceived in the upward direction by reproducing only P2, P1, and P1 + P2; and (4) P2 could be utilized to enhance N1 at the upper direction.

Session 1aSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration I (Poster Session)

Benjamin Shafer, Chair
Technical Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

All posters will be on display from 10:30 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 10:30 a.m. to 11:15 a.m. and authors of even-numbered papers will be at their posters from 11:15 a.m. to 12:00 noon.

Contributed Papers

1aSA1. Evaluation of environmental noise by analyzing faint random vibration in structural health monitoring. Yoshinori Takahashi (Tokyo Metropolitan College of Industrial Technol., 8-17-1, Minamisenju, Arakawaku, Tokyo 116-0003, Japan, yoshinori@ieee.org)

A number of big intermittent aftershocks often follow a great earthquake. Even if a building could avoid collapse by the first main quake, there would be no guarantee that the building can survive from the aftershocks. However, if it is possible to monitor the condition of the building constantly using some faint random vibration caused by environmental noise such as wind, weak earthquakes, and traffic noise, the safety of the victims should be more secured. In a field of machinery diagnostics, although several diagnostic methods using vibration based on environmental noise have been proposed, those methods require that the characteristics of the noise are known since the resonance of the object is shadowed by the environmental noise spectrum. The author has proposed an emphasizing method of hidden resonances in a time-invariant transfer function and has also applied the method to structural health monitoring by analyzing faint random vibration caused by environmental noise. This report presents the results of experiment in an actual building. The results showed that rain showers or temporal thunders gave some influences on the estimation for the resonance of the buildings; however, it was possible to avoid the influences by selecting the analysis parameters correctly.

1aSA2. Modelling of dynamic behaviour of viscoelastic materials in the frequency domain. Tamás Pritz (Budapest Univ. of Technol. and Economics, Műegyetem rkp. 3-9, Budapest 1111, Hungary, tampri@eik.bme.hu)

The modern methods of acoustical and vibration calculus require accurate material models to take properly into account the damping and dynamic elastic properties of the viscoelastic materials used for sound and vibration control. The usual modeling methods (spring-dashpot models, fractional calculus) start out of the stress to strain relations defined in the time-domain, which are then transformed into the frequency domain to find the appropriate material functions. In contrast, the damping and elastic properties are known from measurements made in the frequency domain; therefore, it seems to be reasonable to model the material behavior directly in this domain. The aim of this paper is to show that the typical dynamic behavior of solid viscoelastic materials can efficiently be modeled in the frequency domain. This modeling method is based on simple theoretical considerations, the frequency domain experiences and the causality principle. Two fundamental material models are developed in this way, namely one for the damping increasing with frequency, and the other one for the damping peak, which can be either symmetrical or asymmetrical with respect to the logarithmic frequency. The relation of the frequency domain models to the popular fractional derivative models defined in the time-domain is shown and discussed.

1aSA3. Numerical simulation for low frequency noise from viaduct by the vehicle load. Noboru Kamiakito, Masayuki Shimura (Civil Eng. and Eco-Technol. Consultants Co. Ltd., 2-23-2,Higashi-ikebukuro, Toshima-ku, Tokyo 1700013, Japan, kamiakit@kensetsukankyo.co.jp), Hiroshi Iwabuki (Central Nippon Expressway Co., Ltd., Hachioji-shi, Tokyo, Japan), and Toshikazu Osafune (NEXCO Res. Inst. Co., Ltd., Machida-shi, Tokyo, Japan)

There is the case that the low frequency noise generated complaints of neighborhood residents from the road viaduct. We tried to clarify the source of low frequency noise using the FEM model as one approach for such a problem. First, the 3D-FEM model of the vehicle, bridge, and air was constructed. Then, propagating of the acceleration was calculated by the vehicle traveling on viaduct with road irregularities. Then, the vibration velocity of the viaduct is radiated as sound pressure, and propagating of the sound pressure through the air was carried out by 3D-FEM calculation. Actually, a numerical model for the viaduct that caused the complaint was constructed. The low frequency noise generation mechanism was investigated. The effective measures was considered. As a result, it became clear that a high reduction effect of low frequency noise by performing the measures road irregularities can be obtained in the viaduct. It became clear that there is the acoustic radiation of low frequency noise by the sound insulation wall on the viaduct. By carrying out the parameter study using such a method, we can clarify the occurrence factor of the low frequency noise.

1aSA4. On a fire extinguisher using sound winds. Myungjin Bae and EunYoung Yi (Commun. Eng., Soongsil Univ., 369 Sangdoro, Dongjakgu, Seoul 156-743, South Korea, mjbae@ssu.ac.kr)

Sound fire extinguisher is developed based on principle of quenching fire by lowering its temperature with contacting vibration energy from low frequency sound under 100 Hz to fire and then blocking inflow of oxygen with a sound wind. The one researched by American students are at very basic level to be commercialized. It concentrates a sound beam through special sound winds and increases sound energy performance 10 times more to put out the fire. This is because it utilizes basic theory of making sound and as sound has a tendency of dispersing, big sound and energy have to be created to extinguish fire. That is, even if 100 W is produced, it just makes a big noise but only 1/10 of it can be used for putting out fire. This special sound winds has convex shape which concentrates all the sound into one place to increase effectiveness by more than 10 times. Even 1~2 m away, initial control can be done. As a result, size of battery as well as amplifier can be made small to have minimized size and weight, making it more practical.

1aSA5. Skull-shaped antenna enables near-field super-resolution in acoustic source localization using elastic waves. Michael Reinwald (Laboratoire d'Imagerie Biomédicale, Université Pierre et Marie Curie, 15 rue de l'Ecole de Médecine, Laboratoire d'Imagerie Biomédicale, Paris 75006, France, mchlrnwld@gmail.com), Stefan Catheline (LabTAU, Unité U1032, Université Claude Bernard Lyon 1, Lyon, France), Quentin Grimal (Laboratoire d'Imagerie Biomédicale, Université Pierre et Marie Curie, Paris, France), and Lapo Boschi (Institut des Sci. de la Terre de Paris, Université Pierre et Marie Curie, Paris, France)

Internal cues such as bone conducted sound (i.e., long signals involving reverberation within the skull) may be beneficial to acoustic source localization in humans. We investigate the potential of bone conduction for human localization in the near field. Following previous work of the authors, we use a conceptual processing model that translates elastic waves conducted and reverberated in a skull-like object into spatial positioning through a time-reversal analysis. The localization accuracy is tested for one and two piezoelectric receivers glued close to the position of the ears, measuring solely the vibrations of the object. The sound source is placed along the azimuthal and sagittal planes for distances to the skull between 5 and 100 cm. We are able to estimate the source position in the near-field ($<$20cm) with a resolution below the diffraction limit ($1/4$ of a wavelength) for frequencies in the physiological hearing range (~2kHz). We infer that anatomical details of the skull give rise to complex features of the skull's Green's function enabling super-resolution localization. The role of the skull is presumably particularly relevant in marine animals, owing to the effectiveness of energy transfer from water to bones. We are currently investigating this phenomenon on dolphin skulls.

1aSA6. A sound absorbing metasurface with coupled resonators. Junfei Li, Wenqi Wang, Yangbo Xie, Bogdan Popa, and Steven A. Cummer (ECE, Duke Univ., 101 Sci. Dr., Rm. 3417, FCIEMAS Bldg. Duke Univ, Durham, NC 27708, junfei.li@duke.edu)

An impedance matched surface is able, in principle, to totally absorb the incident sound and yield no reaction, and this is desired in many acoustic applications. Here, we demonstrate a new design of impedance matched sound absorbing surface with a simple construction. By coupling diJerent resonators and generating a hybrid resonance mode, we designed and fabricated a metasurface that is impedance-matched to airborne sound at tunable frequencies with subwavelength scale unit cells. With careful design of the coupled resonators, over 99% energy absorption at central frequency of 511 Hz with a 50% absorption bandwidth of 140 Hz is achieved experimentally. The proposed design can be easily fabricated and is mechanically stable. The proposed metasurface can be used in many sound absorption applications such as loudspeaker design and architectural acoustics.

1aSA7. Passive structural health monitoring using cross correlation of ambient vibrations in an aluminum plate. Sun Ah Jung (Seoul National Univ., Bldg. 36 212, 1 Gwanak-ro, Gwanak-gu, Seoul 151-742, South Korea, sunj@snu.ac.kr), Keunhwa Lee (Sejong Univ., Seoul, South Korea), and Woojae Seong (Seoul National Univ., Seoul, South Korea)

Green's function extracted from cross correlating ambient diffuse field is used for structural health monitoring. Experimental study was conducted to detect structural damages in an aluminum plate using two accelerometers. Green's function obtained by the cross correlation in an undamaged plate is compared to the signal directly generated by an impact hammer. A good agreement between active (impact hammer) and passive (cross correlation) processing is demonstrated. Various damage scenarios, including mass addition and hole in the plate, are investigated. Different from damage index definition in a recent study which compares the causal and anti-causal part of the Greens' function using the reciprocity theorem, we computed the damage index by comparing the causal part of Green's function of an undamaged plate to that of a damaged plate. Distributions of damage indices, according to various number of noise sources for each type of damage type, are analyzed. Using the support vector machine, we classified the damage index distributions. Error analysis of these classifications is given; it shows that present damage index definition provides better damage diagnosis than previous definitions even when using small number of sources.

1aSA8. Urban seismology: Numerical results and real data observations on the clustering effect of buildings. Andrea Colombi (Mathematics, Imperial College, South Kensington, London SW7 2AZ, United Kingdom, andree.colombi@gmail.com) and Philippe Gueguen (ISTerre, Grenoble, France)

Seismic design parameters for buildings are traditionally based on the seismicity of the area, the so-called site effects, and structural response decoupling with the underlying surface. It is however well known that buildings strongly affect the wavefield interacting in a non-linear way through the foundations and the surrounding soil. In dense urban areas, buildings are very close to each other, and it is reasonable to expect a collective effect known as site-city interaction (SCI). Initially, observations of this phenomenon have been made difficult by the lack of data. Now, the increased availability of continuous records from permanent seismic networks in urban areas or buildings has made SCI an important topic in urban seismology. Here, after presenting some numerical results of SCI in densely built areas, we discuss a recent study on three residential towers located in the city of Grenoble (France) exhibiting an SCI that depends on the strike angle of the seismic energy. Both numerical simulations and recorded data exhibit dynamic coupling within the building cluster that has a non-negligible impact on the building response. The angle dependence helps interpreting the tenants' answers to a survey made right after the Earthquake in each of the tower.

Session 1aSC

Speech Communication: Speech Technology: Synthesis and Interfaces (Poster Session)

Jonathan C. Yip, Chair

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All posters will be on display from 10:30 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 10:30 a.m. to 11:15 a.m. and authors of even-numbered papers will be at their posters from 11:15 a.m. to 12:00 noon.

Contributed Papers

1aSC1. Emotional voice conversion system for multiple languages based on three-layered model in dimensional space. Yawen Xue (Information Sci., Japan Adv. Inst. of Sci. and Technol., 1-1 Asahidai, Nomi City, Ishikawa, Japan), Yasuhiro Hamada (Meiji Univ., Tokyo, Japan), and Masato Akagi (Information Sci., Japan Adv. Inst. of Sci. and Technol., Nomi City, Ishikawa, Japan, akagi@jaist.ac.jp)

This paper proposes a system to convert neutral speech to emotional one with controlled intensity of emotions for multiple languages. Most of previous researches considering synthesis of emotional voices used statistical or concatenative methods that can synthesize emotions in category. While humans sometimes enhance or relieve emotional states and intensity during daily life, synthesized emotional speech in categories is not enough. A dimensional approach which can represent emotion as a point in a dimensional space can express emotions with continuous intensity. Employing the dimensional approach, a three-layered model to estimate displacement of the acoustic features of the target emotional speech from that of source (neutral) speech and a rule-based conversion method to modify acoustic features of source (neutral) speech to synthesize the target emotional speech are proposed. To convert the source speech freely and easily, Fujisaki model for f0 contour and target prediction model for power envelope are introduced to parameterize dynamic features in prosody. Although this system is trained with Japanese database, neutral speech in two different languages, English and Chinese are tried to convert. Evaluation results show that all converted voices can convey the same impression with satisfactory order of emotional intensity and naturalness as Japanese voices. It means that the emotion voice conversion system can work for other languages with controllable emotion category and intensity although it is built in one language.

1aSC2. Feasibility study on synthesis of English vowels with a vocal tract mapping interface. Riku Aoki (Graduate School of Sci. and Technol., Kumamoto Univ., 2-39-1 Kurokami Chuo-ku Kumamoto, Ogata Lab., Kumamoto 860-8555, Japan, aoki_r@st.cs.kumamoto-u.ac.jp), Kohichi Ogata (Faculty of Sci. and Technol., Kumamoto Univ., Kumamoto, Japan), and Akihiro Taruguchi (Graduate School of Sci. and Technol., Kumamoto Univ., Kumamoto, Japan)

This paper describes the synthesis of English vowels with a vocal tract mapping interface developed in our laboratory. This interface uses five vocal tract shapes that are located at apexes of a pentagonal chart in consideration of Japanese vowels. It allows users to configure the vocal tract shape corresponding to the location of the mouse pointer based on interpolation. Moreover, it has a function of the inverse estimation of the vocal tract from formant frequencies. Although the interface has been developed considering the Japanese vowels, whether English vowels can also be synthesized with this interface is an interesting topic for confirmation of its versatility. Therefore, we investigated the feasibility of producing English vowels ([ɑ], [æ], [ɔ], [ɛ], [ɝ], [i], [I], [u], [ʊ], [ʌ]) with the mapping interface. As the first step, we investigated that the mapping interface could estimate vocal tract shapes for the English vowels synthesized with the VowelEditor in PRAAT as test sounds. Furthermore, listening tests were conducted to examine whether the re-synthesized vowel sounds with the mapping interface were matched with the original vowel sounds by PRAAT. The experimental results showed that the mapping interface has a potential for producing English vowels.

1aSC3. Speech resynthesis from 1/f$^{\beta}$ noise. Robert Fuhrman and Eric Vatikiotis-Bateson (UBC, #402-2588 Alder St., Vancouver, BC V6H4E3, Canada, robert.a.fuhrman@gmail.com)

Speech can be resynthesized from acoustic parameters estimated from the motion of the vocal tract articulators, head, and face. This has shown that speech information is broadly distributed both in space and time across the physiological systems associated with speech production, and correspondingly, it provides a basis for exploring what properties of signal structure are relevant for specifying the relation between acoustics and physiology. In this presentation, we use the resynthesis paradigm to explore the extent to which the acoustic-articulatory relation can be specified in terms of the structure of *fluctuations* in the speech signal. Specifically, we demonstrate that speech acoustics closely matching the original can be obtained by replacing the vocal tract kinematic signals typically used in the re-synthesis paradigm with surrogate signals having *1/f$^{\beta}$fractal* fluctuation structure—a known (Voss & Clarke 1975) statistical property of the speech amplitude envelope that holds for low frequencies corresponding to the rate of articulatory motion (<10 Hz: Dudley, 1939). We also address the hypothesis that quantitative descriptors derived from both the fractal and *multifractal* properties of speech link the *time-varying* properties of speech acoustics to the cooperative motion of physiological structures involved in speech production.

1aSC4. Singing voice synthesizer with non-filter waveform generation using spectral phase reconstruction. Yasuhiro Hamada (Meiji Univ., Nakano, Nakano 1648525, Japan, hamada@meiji.ac.jp), Nobutaka Ono (National Inst. of Informatics / SOKENDAI, Chiyoda, Japan), and Shigeki Sagayama (Meiji Univ., Nakano, Japan)

This paper discusses a singing voice synthesis system based on non-filter waveform generation using spectral phase reconstruction as an alternative method to replace the conventional source-filter model. Source-filter model has been considered as an essential technique in the long history of speech synthesis as it simulates the human process of speech production consisting of excitation and resonances, even after hidden Markov model was introduced in the 90s toward statistically trainable speech synthesis. Filter (particularly, recursive filter), however, may cause serious problems in undesired amplitude and long time decay when sharp resonances in recursive filter match harmonics of F0 of the excitation. As the ultimate purpose of usage of filter is to transfer the spectrogram designed in the TTS system to the listener's hearing organs, an alternative solution can be brought from "phase reconstruction" without using filters. We propose a spectral phase reconstruction, instead of using filter, to generate waveform from the desired

power spectrogram. We applied this framework to singing voice synthesis where fundamental frequency (F0) was modified according to the target melody. The result of the listening tests showed that the proposed method gives higher quality of naturalness than previous methods with recursive filter.

1aSC5. Analysis of driver workload when using speech interfaces. Daiki Hayashi, Chiyomi Miyajima, and Kazuya Takeda (Graduate School of Information Sci., Nagoya Univ., Furo-cho, Chikusa, Nagoya 4648603, Japan, hayashi.daiki@g.sp.m.is.nagoya-u.ac.jp)

Speech interfaces are useful for improving the accessibility of in-vehicle devices as they are considered to be less distracting than visual interfaces. However, there is still considerable cognitive demand when using a speech interface, which may increase the chance of a traffic accident. This study investigates driver workload induced by various tasks, including a speech interface used for music retrieval. Driving data were collected from 523 drivers operating an instrumented vehicle on urban streets and highways. The drivers were given each of the following secondary tasks while driving: a) repeating random strings of four letters they heard through an earphone, b) interacting with a spoken dialogue system to retrieve and play music, c) being directed to an unfamiliar place by a human navigator on a cell phone, and d) reading aloud words seen on signboards while driving. After driving, each driver subjectively rated the induced workload for each secondary task from 1 (lowest) to 5 (highest). Analysis of the results of their subjective evaluations showed that talking to a navigator on a cell phone resulted in the lowest average workload (about 1.6), which was almost the same as the driving without any secondary task. Repeating random strings of four letters and retrieving music using a speech interface produced the highest average workload (about 2.5).

1aSC6. An Indonesian concatenative speech synthesis system. Toshio Hirai and Ivan Setiawan (Arcadia, Inc., Minoh Sun-plaza 1st Bldg. 7F, 6-3-1 Minoh, Minoh, Japan, thirai@arcadia.co.jp)

An Indonesian concatenative speech synthesis system was developed that can (1) differentiate between the pronunciation of [e] and [@] (XSAMPA) by utilizing a pronunciation dictionary and pronunciation rules and (2) automatically create derived words that are then inserted into the pronunciation dictionary, which is used in converting text into phonemes. Speech data were recorded from a female native Indonesian speaker. Phrases included proper nouns and text samples from Indonesian textbooks, newspaper articles, and TV scripts, which covered almost all possible Indonesian syllables. There were 4,490 phrases lasting a total of 5.1 hours. The transcription was constructed considering pauses during narration. A forced alignment technique was used to obtain the phoneme boundary of the speech, and each phoneme and its acoustic/linguistic features were added to a database. In the synthesis step, an input text was converted into phonemes. An appropriate segment sequence, which has features similar to the phonemes should have and shows the least concatenative distortion, was selected from the database and was concatenated into a waveform. The synthesized speech was of acceptable quality, but we discovered some problems, such as sound discontinuity. The parameter weights of cost function to evaluate the similarity and the distortion should be optimized further.

1aSC7. Robust text-to-speech duration modelling with a deep neural network. Gustav E. Henter (National Inst. of Informatics, 13F 1313, 2-1-2 Hitotsubashi, Tokyo, Chiyoda-ku 101-8430, Japan, ghenter@inf.ed.ac.uk), Srikanth Ronanki, Oliver Watts, Mirjam Wester, Zhizheng Wu, and Simon King (The Univ. of Edinburgh, Edinburgh, United Kingdom)

Accurate modeling and prediction of speech-sound durations is important for generating more natural synthetic speech. Deep neural networks (DNNs) offer powerful models, and large, found corpora of natural speech are easily acquired for training them. Unfortunately, poor quality control (e.g., transcription errors) and phenomena such as reductions and filled pauses complicate duration modelling from found speech data. To mitigate issues caused by these idiosyncrasies, we propose to incorporate methods from *robust statistics* into speech synthesis. Robust methods can disregard ill-fitting training-data points—errors or other outliers—to describe the typical case better. For instance, parameter estimation can be made robust by replacing maximum likelihood with a robust estimation criterion based on the density power divergence (a.k.a. the β-divergence). Alternatively, a standard approximation for output generation with mixture density networks (MDNs) can be interpreted as a robust output generation heuristic. To evaluate the potential benefits of robust techniques, we adapted data from a free online audiobook to build several DNN-based text-to-speech systems, with either conventional or robust duration prediction. Our objective results indicate that robust methods described typical durations better than the baselines. Additionally, listeners significantly preferred synthetic speech generated using the robust methods in a subjective evaluation.

1aSC8. Speech synthesizers controlled by the movements of non-vocal organs. Masashi Ito (Tohoku Inst. of Technol., 35-1 Yagiyama-Kasumicho, Sendai 982-8577, Japan, itojin@tohtech.ac.jp)

A speech-like signal can be synthesized without the text-to-speech conversion if speech parameters, such as loudness, pitch, and phonemic information, are successively determined in the utterance. To control the parameters, human body motions have been utilized in this type of speech synthesis. The speech synthesizers controlled by the movement of non-vocal organs (MNVO) are expected to provide effective assistant tools for speech disabilities. Although there have been extensive studies for the MNVO synthesizers, the optimal conversion strategy from body motions to speech parameters has not been clarified yet. This is because that the optimal MNVO speech synthesizer requires both usability and expressiveness while they have a trade-off relationship in general. In order to address this problem, a refined architecture of the MNVO synthesizer is proposed in the present study. This is characterized by an explicit separation between motion detection and speech synthesis processes, which makes it possible to examine a common conversion strategy regardless of the input devices. Based on this architecture, several MNVO speech synthesizers were realized using different types of the motion sensors. Evaluating the performances of these systems, the effect and limitation of the proposed architecture is discussed.

1aSC9. The UCLA voice synthesizer, version 2.0. Jody Kreiman, Norma Antonanzas-Barroso, and Bruce R. Gerratt (Dept. of Head and Neck Surgery, UCLA School of Medicine, 31-24 Rehab Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu)

Based on our psychoacoustic model of voice quality, the UCLA voice synthesizer allows users to copy synthesize nearly any steady-state voice sample or to create stimuli that systematically vary in specific acoustic dimensions. This new release contains a number of significant improvements from earlier versions. The vocal tract model now includes three spectral zeros with adjustable bandwidths. The precise spectral shape of the harmonic and inharmonic sources can be modified at will, either by adjusting the heights of the harmonics or the noise amplitude in a selected frequency range or by specifying the desired spectral or noise slope in a range. Any number of ranges of any size can be specified. Synthesizer variables can be individually saved and copied between cases. Additional changes increase control of analysis parameters and the ease with which series of stimuli can be created. The synthesizer is fully documented and is freely available for download from headandnecksurgery.ucla.edu/glottalaffairs. Copies will also be available at the conference. [Research supported by NIH.]

1aSC10. Improving Spanish speech synthesis intelligibility under noisy environments. Jaime Lorenzo-Trueba (Speech Technol. Group / NII, 2-1-2 Hitotsubashi, Chiyoda-ku, Tokyo, Tokyo 101-8430, Japan, jaime@nii.ac.jp), Roberto Barra-Chicote (Speech Technol. Group, Madrid, Spain), Junichi Yamagishi (National Inst. of Informatics, Tokyo, Japan), and Juan M. Montero (Speech Technol. Group, Madrid, Spain)

In this paper, we evaluate a newly recorded Spanish Lombard Speech database. This database has been recorded with expressive speech synthesis in mind, and more particularly adaptability to the environment. Real stationary noise recorded inside of a car was used to produce the Lombard speech response in the speaker by means of a headphone. Four different noise levels were used, in steps of 5 dB and also clean speech to set the clean speech baseline. Finally, a pair of intelligibility evaluations were carried out, one with natural speech that proves the validity of the recorded database by

showing a 37% absolute increase in intelligibility in a -10 dB SNR condition when compared to non-Lombard speech. The second evaluation was carried out with synthetic speech, which showed a 10% absolute increase in intelligibility for both the -10 and -15 dB SNR condition.

1aSC11. Example-based spoken chat system which can be customized for each user. Eichi Seto and Norihide Kitaoka (Tokushima Univ., 2-1 Minami-Johsanjima-cho, Tokushima-shi, Tokushima 770-8506, Japan, c501637006@tokushima-u.ac.jp)

There is a need to develop automated dialog systems, such as chatbots, which are capable of engaging in natural conversations with elderly users. We propose an example-based dialog system featuring an adaptation method which customizes the dialog for each user. After retrieving information about the user from the Web using the user's profile, morpheme analysis is applied to the retrieval results and only proper nouns are extracted. Words which appear with high frequencies are considered to be related to the user. We then calculate the similarity between words related to the user and words in example phrases in the dialog system. Cosine similarity between distributed representations of words is calculated using Word2vec. We then generate a phrase which has been adapted to include a topic related to the user by substituting user-related words for highly similar words in the original example phrase. We manually evaluated the naturalness of the generated phrases and found that the system did in fact generate natural-sounding phrases. However, Word2vec sometimes mistook words which were grammatically different parts of speech (POS) for similar words, so we added a POS constraint to the substitution process. As a result, the system achieved higher phrase generation precision.

1aSC12. A DNN-based text-to-speech synthesis system using speaker, gender, and age codes. Hieu Thi Luong (Comput. Sci., VNU - HCM - Univ. of Sci., Ho Chi Minh, Viet Nam), Shinji Takaki (Digital Content and Media Sci. Res. Div., National Inst. of Informatics, 2-1-2, Hitotsubashi-cho, Chiyoda-ku, Tokyo 101-8430, Japan, takaki@nii.ac.jp), SangJin Kim (Naver Labs, Naver Corp., Seongnam, South Korea), and Junichi Yamagishi (Digital Content and Media Sci. Res. Div., National Inst. of Informatics, Chiyoda-ku, Tokyo, Japan)

Adaptation and manipulation techniques for creating various characteristics of synthetic speech are important research topics in the speech synthesis field. In this work, we investigate the performance of a DNN-based text-to-speech synthesis system that uses speaker, gender, and age codes as well as the text inputs (1) for modeling speaker-independent models called "average voice models," (2) for performing speaker adaptation using a small amount of adaptation data, and also (3) for manipulating characteristics of synthetic speech based on the codes. For these purposes, we extracted a set of studio-quality speech data uttered by 68 males and 70 females, whose age vary between 10 and 80, from our large-scale Japanese corpus and carried out the three experiments: (1) We constructed a DNN-based speaker-independent model using one-hot vectors representing a set of the above speakers. (2) We performed speaker adaptation by estimating a code vector for a new speaker via the back-propagation. (3) We performed manual manipulation of the code vector to modify perceived characteristics, gender, and/or age of synthetic speech. Experimental results showed that high-performance speaker-independent models can be constructed using the proposed code vectors and additionally that adaptation and manipulation using the codes can also be performed effectively.

1aSC13. Speech analysis-synthesis system using principal components of vowel spectra. Tomio Takara, Akichika Higa, Syouki Kaneshiro, and Yuuya Oshiro (Eng., Univ. of Ryukyus, 1 Senbaru, Nishihara-chou, Okinawa-ken 903-0213, Japan, takara@ie.u-ryukyu.ac.jp)

We have studied an effective method using principal components spanning a feature space of isolated vowels. A covariance matrix is calculated from many log-amplitude spectra of isolated vowels uttered by a speaker. An eigen equation of the covariance matrix is solved. The resulting eigenvectors are called principal vectors. In the analysis system, log-amplitude spectrum for each frame of a word uttered by the same speaker is transformed to the components on the principal vectors. In the synthesis system, a log-amplitude spectrum is reconstructed using the components on the principal vectors with the largest eigenvalues and the spoken word is synthesized using the LMA filter. We draw the distribution chart of the first and the second principal components extracted from Japanese vowel data. This figure was very similar to the F1—F2 distribution of vowels and so to the vowel classification map in coordinate axes of the degree of constriction and the tongue hump position. The Listening tests showed that the quality of the word with only 7 parameters per frame synthesized by the proposed method was comparable to the word with 9 to 12 parameters by the cepstral method. Consequently, the effectiveness of the proposed method was shown.

1aSC14. Improved ETSI advanced front-end for automatic speech recognition using quadrature mirror filter synthesis based on robust complex speech analysis. Keita Higa (Toshiba Corp., Tokyo, Japan) and Keiichi Funaki (C&N Ctr., Univ. of the Ryukyus, Senbaru 1, Nishihara, Okinawa 903-0213, Japan, funaki@cc.u-ryukyu.ac.jp)

An automatic speech recognition (ASR) is commonly used in these days. Current ASR systems perform well in ideal environment; however, it does not perform well in realistic noisy environment. As a robust ASR, ETSI has standardized Advanced Front-End (AFE) that adopts two-stage of iterative Wiener filter (IWF) to realize a speech enhancement as the front-end of ASR. In the ETSI AFE, 16 kHz speech is divided uniformly by QMF (Quadrature Mirror Filter) into lower-band and higher-band signal and only lower-band signal is emphasized by the IWF and MFCC (Mel Frequency Cepstral Coefficient) is extracted from both the second-stage of emphasized lower-band signal and non-emphasized higher-band signal. FFT is used to estimate speech spectrum that designs the Wiener filter. On the other hand, we have already proposed robust complex speech analysis for an analytic signal. It can estimate more robust and more accurate speech spectrum due to the introduced robust criterion and nature of analytic signal. This paper proposes an improved AFE using wide-band robust ELS (Extended Least Square) complex analysis and real-valued analysis instead of FFT. Moreover, the QMF synthesis is also introduced for second-stage of wide-band analysis. The experimental results using the CENSREC-2 speech database demonstrate that the performance is improved.

1aSC15. Spectral and pitch modeling with hybrid approach to singing voice synthesis using hidden semi-Markov model and deep neural network. Kouki Hongo, Takashi Nose, and Akinori Ito (Graduate School of Eng., Tohoku Univ., 6-6-05 Aoba, Aramaki, Aoba-ku, Sendai, Miyagi 980-8579, Japan, kouki.hongou.p4@dc.tohoku.ac.jp)

We propose a corpus-based singing voice synthesis system combining the hidden Markov model (HMM) and the Deep Neural Network (DNN). Recently, in the area of text-to-speech synthesis, it was reported that the DNN-based speech synthesis method showed better speech quality than the HMM-based one. However, when we introduced the DNN to statistical singing voice synthesis, it did not improve the synthetic singing voice quality. Thus, we introduced the DNN in the singing voice synthesis in a different way. Instead of modeling the speech spectra, we exploited the DNN to model the difference between the spectra of natural singing voice and synthetic singing voice from the HMM. To do that, we used the DNN to map the input musical information such as lyrics, tones, durations into the difference of output acoustic features between the natural and synthetic singing voice. This allows us to reconstruct the spectral fine structures in singing voice generated by HMMs. Our results proved that the proposed method improved the quality of synthetic singing voice compared to the conventional methods.

1aSC16. Low delay statistical singing voice conversion with direct waveform modification based on spectral differential considering global variance. Kazuhiro Kobayashi (Information Sci., Nara Inst. of Sci. and Technol., Nakatomigaoka1-4162-1-D06-201, Nara, Nara 6310003, Japan, kazuhiro-k@is.naist.jp), Tomoki Toda (Information Technol. Ctr., Nagoya Univ., Nagoya, Japan), and Satoshi Nakamura (Information Sci., Nara Inst. of Sci. and Technol., Nara, Japan)

This paper presents a low-delay statistical singing voice conversion (SVC) method with direct waveform modification based on spectral differential (DIFFSVC) considering global variance (GV). Statistical SVC based on a Gaussian mixture model (GMM) is a technique to convert singer identity of a source singer's voice into that of a target singer converting several acoustic features. We have successfully improved quality of converted singing voices by proposing the DIFFSVC considering the GV to avoid parameterization errors caused by vocoding process and alleviate over-smoothing effects. On the other hand, this method is based on batch-type conversion processing, and therefore, it is difficult to directly apply it to real-time conversion processing that needs a low delay conversion method. In this paper, to develop a high-quality and real-time SVC system, we propose a low-delay DIFFSVC method using GV-based post-filtering process. The experimental results have demonstrated that the proposed method makes it possible to improve speech quality of the converted singing voice in low-delay conversion processing. [This work was supported in part by JSPS KAKENHI Grant Number 26280060, Grant-in-Aid for JSPS Research Fellow Number 16J10726, and by the JST OngaCREST project.]

1aSC17. Improvement of quality of voice conversion based on spectral differential filter using STRAIGHT-based Mel-cepstral coefficients. Harunori Koike, Takashi Nose (Tohoku Univ., 6-6-05, Aoba, Aramaki, Aoba-ku, Sendai 980-8579, Japan, harunori.koike.q5@dc.tohoku.ac.jp), Takahiro Shinozaki (Tokyo Inst. of Technol., Yokohama, Japan), and Akinori Ito (Tohoku Univ., Sendai, Japan)

In our previous research, we proposed a technique for converting the speech individuality of speech of an arbitrary input speaker into a specified speaker's one. This technique used the neural network (NN) for many-to-one mapping, which was trained from the pairs of multiple source speakers and a target speaker. Using the NN, we design a filter based on the difference of the spectra of the source and the target speech to directly convert the waveform of the input speaker into the target speaker's one. An advantage of the proposed method is that the direct waveform conversion alleviates the quality degradation caused by the F0 extraction error. To improve this proposed method, we use mel-cepstral coefficients extracted by STRAIGHT, the high-quality tool of speech analysis and synthesis. The higher order components of the cepstral coefficients determine the detailed shape of the spectral envelope, but the prediction accuracy of the high-order coefficients might be worse than that of the lower-order ones. Thus, we investigated the effect of the order of cepstral components to find the condition that gave better conversion quality. Additionally, we exploited the dynamic features to reduce the discontinuity of the frame and improve the converted speech quality.

1aSC18. Acquisition and evaluation of a human-robot elderly spoken dialog corpus for developing computerized cognitive assessment systems. Hiroaki Kojima (National Inst. of Adv. Industrial Sci. and Technol. (AIST), 1-1-1, Umezono, Tsukuba, Ibaraki 305-8560, Japan, h.kojima@aist.go.jp), Kana Takaeda (National Rehabilitation Ctr. for Persons with Disabilities, Tokorozawa, Japan), Misato Nihei (The Univ. of Tokyo, Tokyo, Japan), Ken Sadohara (National Inst. of Adv. Industrial Sci. and Technol. (AIST), Tsukuba, Japan), Shinichi Ohnaka (NEC Corp., Kawasaki, Japan), and Takenobu Inoue (National Rehabilitation Ctr. for Persons with Disabilities, Tokorozawa, Japan)

Personal communication robots are expected to assist daily living of elderly people. Aiming at developing computerized cognitive assessment systems, we collected human-robot spoken dialog of a cognitive impairment test scenario based on TICS (Telephone Interview for Cognitive Status) and COGNISTAT (Cognitive Status Examination) in Japanese. For the efficient acquisition of the spoken dialog corpus of this scenario, we implemented a WOz (Wizard of Oz) style spoken dialog system on a commercial personal robot, PaPeRo by NEC. By using this system, we collected 147 dialogs spoken by 48 elderlies whose ages varied about 75-85 and MMSE (Mini Mental State Examination) scores varied around 26.5. Each dialog took about 30 min, and contains around 100 human utterances. In order to evaluate feasibility of automatic assessment, we conducted speech recognition experiments with the speech corpus. In the recognition experiments of the elderly speech of answering to the temporal orientation test of asking today's month and date, the accuracies of discriminating correctness of the answers exceeds 80% by using speech dictation engines. This preliminary results show potential feasibilities for computerized cognitive assessment systems.

1aSC19. A method to estimate glottal source waves and vocal tract shapes for widely pronounced types using ARX-LF model. Yongwei Li, Daisuke Morikawa, and Masato Akagi (Japan Adv. Inst. of Sci. and Technol., Asahidai 1-1, Nomishi, Ishikawaken 9231211, Japan, yongwei@jaist.ac.jp)

Accurate estimates of glottal source waves and vocal tract shapes can be used for speech synthesis with high quality. One of source-filter models, ARX-LF model, is widely used to estimate glottal source waves of neutral speech and singing voices. This paper extends the ARX-LF model to accurately estimate glottal source waves and vocal tract shapes with more widely pronounced types. The proposed method mainly contains two steps: the first step is to set initial values of LF parameters and shift glottal closure instant position, and the second step is to save optimum LF parameters and vocal tract coefficients when mean-square equation error is the smallest. The voices are synthesized by different LF parameters and fixed vocal tract filter model, which are used to test whether the proposed method can work well. The results show that the proposed method has ability to estimate LF parameters in more widely pronounced types. The proposed method is further used to estimate glottal source waves and vocal tract shapes of emotional voices. Spectral tilts of estimated glottal waves and vocal tract shapes derived from ARX filters are consistent with other reports and human speech production mechanisms.

1aSC20. Improvement of the noisy speech recognition using method of thresholds and emphasizing wavelet coefficients for clean speech. Yoichi Midorikawa and Masanori Akita (Faculty of Eng., Oita Univ., 700 Dannoharu, Oita, Oita 870-1192, Japan, ymido@oita-u.ac.jp)

In recent years, speech recognition under the noisy environment is one of the very important technologies. This study improves a speech recognition method for the signals under the noisy environment using method of thresholds and emphasizing wavelet coefficients for clean signal data. Noisy speech recognitions are extremely difficult problem. To analyze noise problem, in general, the majority of people have used the Fourier analysis. But the Fourier transform reveals only the frequency information. The general noise filters reduce specific frequency band contained both signal and noise. It is difficult to eliminate only noise component from a signal containing noise components. To overcome this difficulty, we applied the wavelet analysis. In this study, we improve the speech recognition under the noisy environment by bringing it close to reference data and noisy input data by modifying the spectrum by the wavelet transform using thresholds and emphasizing methods for reference clean speech data. We apply this method using wavelet transform to the modification of spectral envelope shape. As a result, noisy speech recognition rate is improved by this method using wavelet transform for clean speech data.

1aSC21. Multimodal control system for autonomous vehicles using speech and gesture recognition. Takuma Nakagawa and Norihide Kitaoka (Tokushima Univ., 2-1 Minami-Johsanjima-cho, Tokushima-shi, Tokushima 770-8506, Japan, c501637005@tokushima-u.ac.jp)

The recent development of autonomous vehicles has attracted much attention, but operating these vehicles may be too complex for average users. Therefore, we propose an intuitive, multimodal interface for the control of autonomous vehicles using speech and gesture recognition to interpret and execute the commands of users. For example, if the user says "turn there" while pointing at a landmark, the vehicle can utilize this behavior to correctly understand and comply with the user's intent. To achieve this, we designed a two-part interface consisting of a multimodal understanding component and a dialog control component. Our multimodal understanding and dialog control components can be seen as a concatenation of two separate transducers. One transducer is used for multimodal understanding and the other for a conventional dialog system. We then construct a combined transducer from these two transducers. We developed various scenarios which might arise while operating an autonomous vehicle and displayed these scenes on a monitor. Subjects were then asked to operate a virtual car using speech commands and pointing gestures to control the vehicle while observing the monitor. The questionnaire results show that subjects felt they were able to easily and naturally operate the autonomous vehicle using utterances and gestures.

1aSC22. Experimental study of impression-based statistical mapping between speakers' faces and their voices. Yasuhito Ohsugi, Daisuke Saito, and Nobuaki Minematsu (the Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo 1130033, Japan, yasuhito.ohsugi@gavo.t.u-tokyo.ac.jp)

Recently, many types of voice-user interface (VUI) have been developed and some of them become prevalent. To realize more natural conversation between a user and a machine, one feasible approach is expected to be enhanced personification of the machine. In this study, we attempt to give an adequate face to the voice quality of the speech synthesizer embedded in the machine. A research question here is what kind of face should be provided to a given voice quality? Also, an inverse question is possible when a face is given in advance: what kind of voice quality should be mapped to a given face? To solve a problem of mapping between a set of elements and another set, a statistical mapping is often applied, where a parallel and pairwise corpus of elements and their corresponding ones is required. In this study, GMM-based mapping is adopted as statistical mapping between voice features and face features. Further, a parallel corpus is prepared by asking subjects to select suited voices (faces) for given faces (voices) based on their impressions on the faces and the voices. Experiments showed that, although adequate voices can be automatically generated for given faces, mapping between them was strongly subject-dependent.

1aSC23. A study on acoustic features affecting speaker similarity between recorded voice and synthesized voice. Kenko Ota (Nippon Inst. of Technol., 4-1 Gakuendai, Miyashiro, Minami-saitama 345-8501, Japan, otakenko@nit.ac.jp) and Kohei Yoshida (Nippon Inst. of Technol., Minamisaitama, Japan)

This study focuses on the difference in the audibility between recorded voice and synthesized voice uttered by the same speaker. Voices whose pitch does not fluctuate in an utterance were recorded. Voices are synthesized by STRAIGHT. In this research, recorded voices and synthesized voices are allocated to a three-dimensional space by INDSCAL (Individual Differences Scaling) which is an analysis method of MDS (Multi-dimensional Scaling) in order to investigate the acoustic features which are related to the difference in the audibility between recorded voice and synthesized voice. Subjects compared following three combinations of voices: two recorded voices, two synthesized voices, a recorded voice and a synthesized voice. The acoustic features considered in this research are shown as follows: fundamental frequency, spectral centroid, spectral roll-off, spectral flux, spectral contrast, and spectral intensity. As a result, spectral roll-off and spectral contrast are related to the discrimination whether speakers of recorded voice and synthesized voice with same pitch are same or different. Moreover, fundamental frequency is also related to the discrimination whether speakers of recorded voice and synthesized voice with different pitch are same or different.

1aSC24. An evaluation of the automatically topology-generated auto-regressive hidden Markov model with regard to an esophageal voice enhancement task. Akira Sasou (National Inst. of Adv. Industrial Sci. and Technol., AIST, Central1,1-1-1 Umezono, Tsukuba, Ibaraki 305-8568, Japan, a-sasou@aist.go.jp)

An Auto-Regressive eXogenous (ARX) model combined with descriptive models of the glottal source waveform has been adopted to more accurately separate the vocal tract and the voicing source. However, these methods cannot be easily applied to the analysis of voices uttered by different speech production methods, such as esophageal voice. We previously proposed the voicing source hidden Markov model and an accompanying parameter estimation method. We refer to the model combining the HMM with an Auto-Regressive (AR) filter as AR-HMM. The proposed method automatically generates the optimum topology for the HMM using the minimum description length-based successive state splitting algorithm in order to simultaneously and accurately estimate the vocal tract and voicing source based on a voice excited by an unknown, aperiodic voicing source such as an esophageal voice. In this paper, we evaluate the perceived quality of the enhanced esophageal voices, which are synthesized by filtering the voicing source extracted from normal speakers with the AR filter extracted from the esophageal voices.

1aSC25. Evaluation of electrolarynx controlled by real-time statistical F0 prediction. Kou Tanaka (NAIST, 8916-5 Takayama-cho, Augmented Human Commun. Lab. Graduate School of Information Sci. Nara Inst. of Sci. and Technol., Nara, Ikoma 630-0192, Japan, ko-t@is.naist.jp), Tomoki Toda (Nagoya Univ., Aichi, Nagoya, Japan), and Satoshi Nakamura (NAIST, Nara, Japan)

One of the major speaking methods for laryngectomees is a speaking method using an electrolarynx to generate artificial excitation sounds, instead of vocal fold vibration. Although electrolaryngeal (EL) speech is relatively intelligible, its naturalness is quite low owing to the artificial excitation sounds. To make it possible to produce more naturally sounding EL speech, we have proposed an automatic control method of the fundamental frequency (F0) patterns of the excitation sounds generated from the electrolarynx based on real-time statistical F0 prediction. In this method, a vibration of the electrolarynx to generate the excitation sounds is controlled not according to additional signals consciously provided by the laryngectomees but using only their produced EL speech signals. In the previous report, we have developed a prototype system by implementing our proposed method to the electrolarynx and have evaluated its performance objectively through a simulation. In this paper, we evaluate its performance subjectively in terms of naturalness and intelligibility. The experimental results demonstrate that the prototype system can produce more natural EL speech while maintaining its high intelligibility compared to the conventional electrolarynx. [This research was supported in part by JSPS KAKENHI Grant Nos. 15J10727 and 26280060.]

1aSC26. Investigation of using the highway network to predict the F0 trajectory for text-to-speech synthesis. Xin Wang, Shinji Takaki, and Junichi Yamagishi (National Inst. of Informatics, National Inst. of Informatics, 2-1-2 Hitotsubashi, Tokyo, Tokyo 1018430, Japan, wangxin@nii.ac.jp)

Generation of the F0 trajectory is critical for Text-to-Speech (TTS) synthesis. In most of the recently proposed TTS systems based on neural networks, F0, together with the spectral features of speech, is predicted by a single neural network. However, our recent work, which utilizes the highway network with trainable gates to bypass the non-linear transformation layers, has shown that more accurate F0 can be generated by separating the network for the F0 and the spectral streams. It is also observed that most of the layers for the F0 stream are inactive. In this work, the inactive network for the F0 stream will be analyzed, and empirical evidence will show that a shallow network as well as a deep network for the F0 stream can perform. Additionally, this work will analyze the sensitivity values of the hidden units in the highway network to identify the uninformative input features for predicting the F0 trajectory, which include the automatically inferred ToBI tags.

1aSC27. Many-to-many voice conversion using hidden Markov model-based speech recognition and synthesis. Yoshitaka Aizawa, Masaharu Kato, and Tetsuo Kosaka (Yamagata Univ., 4-3-16 Jonan, Yonezawa, Yamagata 9928510, Japan, tnm45806@st.yamagata-u.ac.jp)

This paper describes a many-to-many voice conversion (VC) technique that does not require a parallel training set of source and target speakers. In a previous study, we already proposed many-to-one VC method that consists

of decoding and synthesis parts, and it does not require a parallel training set. The basic idea of this system is that an input utterance is recognized utilizing the hidden Markov model (HMM) for speech recognition, and the recognized phoneme sequences are used as labels for speech synthesis. The aim of this work is to extend functionality from many-to-one to many-to-many VC. In particular, we focus on the VC of emotional speech. In order to achieve this, we utilize speaker adaptation techniques to adapt HMMs for speech synthesis. By using adaptation techniques, an arbitrary speaker's voice can be produced. In this work, a combination of constrained structural maximum a posteriori linear regression and maximum a posteriori estimation is used for adaptation. In order to evaluate the proposed system, subjective speech intelligibility tests were conducted. In the experiments, the proposed adaptation with 10 utterances was compared with traditional parameter training with 450 utterances. The results showed that speaker adaptation could be carried out without performance degradation.

1aSC28. The effect of training in producing continuous vowels with a data-glove-driven vocal tract configuration tool. Kohichi Ogata (Faculty of Adv. Sci. and Technol., Kumamoto Univ., 2-39-1 Kurokami, Chuo-ku, Kumamoto 860-8555, Japan, ogata@cs.kumamoto-u.ac.jp) and Yusuke Matsuda (Graduate School of Sci. and Technol., Kumamoto Univ., Kumamoto, Japan)

This paper describes vocal tract configuration for continuous vowels by using a data glove. The authors have proposed data-glove-driven vocal tract configuration methods and showed the usefulness of a method with three fingers for producing vowel sounds. In the method, the location of constriction, the degree of constriction, and the lip area are controlled by bending the thumb, middle finger, and index finger, respectively. Moreover, the effectiveness of training for continuous vowels was shown for sequence /aiueo/. Whether any combination of vowels can be synthesized by this method is an interesting topic for confirmation of the effectiveness. To examine the possibility of successful manipulation of the data glove, an experiment for synthesizing various three consecutive vowels was performed with five beginner subjects. Japanese vowel sequences of 60 patterns were used in the experiment. The results proved that the subjects were able to successfully produce the vowel sequences immediately after showing a target sequence through three training sessions for manipulating the data glove.

1aSC29. Generative model of spectra for a word using Fujisaki's model and genetic algorithm. Tomio Takara and Akichika Higa (Information Eng., Univ. of the Ryukyus, 1 Senbaru, Nishihara-cho, Nakagami-gun, Okinawa-ken 903-0213, Japan, k168577@ie.u-ryukyu.ac.jp)

Fujisaki's model is a generative model of fundamental frequency which approximates original time pattern effectively. In this study, we propose a new generative model of spectral sequence using Fujisaki's model. We have already proposed a speech synthesis method using vowel space parameters. The vowel space parameter is defined as the component on principal axis obtained by PCA for log-amplitude spectra of isolated vowels. Log-amplitude spectrum can be reconstructed from linear combination of a few principal vectors with coefficients of the vowel space parameters. Using time sequence of the vowel space parameters, we can synthesize a spoken word. In this study, we apply Fujisaki's model to the time pattern of vowel space parameter. We adopt parameters of Fujisaki's model as genes of the genetic algorithm. Then, we execute the genetic algorithm with the fitness of similarity between time patterns of Fujisaki's model and the pattern generated from the vowel space parameter. We obtained synthesized speech which can be recognized as a spoken word at the first generation of the genetic algorithm. As a result of listening test for vowels showed 45 to 80% correct rate at the first generation. Consequently, the effectiveness of the proposed method is shown.

1aSC30. Combination method air and bone conducted speech for speaker recognition in i-vector space. Satoru Tsuge (Information Systems, Daido Univ., 10-3 Takiharu-cho, Minami-ku, Nagoya, Aichi 457-8530, Japan, tsuge@daido-it.ac.jp) and Shingo Kuroiwa (Chiba Univ., Chiba, Japan)

Recently, some new sensors, such as bone-conductive microphones, throat microphones, and non-audible murmur (NAM) microphones, besides conventional condenser microphones have been developed for collecting speech data. Accordingly, some researchers began to study speaker and speech recognition using speech data collected by these new sensors. From these new sensors, we focus on a bone-conductive microphone. Recently, the speaker recognition method based on i-vector will be state-of-the-art. Hence, in this paper, first, we report the speaker recognition performance using bone-conducted speech based on i-vector-based speaker recognition system. In addition, we propose a speaker recognition method combined bone-conducted speech with air-conducted speech. In this paper, we investigate three combination methods, which are a distance combination method, an i-vector combination method, and a feature combination method. To evaluate the proposed methods, we conducted the speaker identification experiments. From experimental results, performance of the bone-conducted speech is almost same as that of the air-conducted speech under the condition of the enrolment and the evaluation speech collected on same session. In addition, the experimental results show that all proposed methods are able to improve the speaker recognition performance of air- and bone-conducted speech.

1aSC31. Horseshoe bat inspired reception dynamics embed dynamic features into speech signals. Anupam K. Gupta (IBM T. J. Watson Res. Ctr., 1208 Snyder Ln, Blacksburg, Virginia, anupamkg@vt.edu), Jin P. Han (IBM T. J. Watson Res. Ctr., Yorktown, NY), Philip Caspers (ME, Virginia Tech, Blacksburg, VA), Xiaodong Cui (IBM T. J. Watson Res. Ctr., Yorktown, NY), and Rolf Müller (ME, Virginia Tech, Blacksburg, VA)

Horseshoe bats can alter the shape of noseleaves and outer ears (pinnae) at the time of emission and reception of biosonar pulses. The shape changes are a result of specific muscular action and significantly change the emission and reception beams by a spatial redistribution of energy in mainlobe and sidelobes. Here, we explore the potential of embedding dynamic features into the speech signals by means of bat inspired reception dynamics to improve the speech recognition accuracy and speaker localization. For this analysis, a digit dataset with 11 U.S. English speakers was recorded through a robotic setup that mimics pinna dynamics as observed in bats. The utterances were obtained through open-source speech corpora and amplitude modulated (carrier frequency: 55 kHz) to shift to ultrasonic band. These utterances were then re-recorded through the dynamic pinna set up and later demodulated to extract the original signal. It was found that reception dynamics have been able to enrich the speech signals with dynamic, direction-dependent time-frequency patterns. The next step in this research is to assess whether these patterns can be utilized to improve speech recognition and speaker localization.

1aSC32. Normal-rate to fast-rate speech conversion using non-linear compression maps. Michael D. Fry and Eric Vatikiotis-Bateson (Linguist, Univ. of Br. Columbia, 2329 West Mall, Vancouver, BC V6T 1Z4, Canada, mdfry20@gmail.com)

This paper presents a new technique to convert normal-rate speech into intelligible fast-rate, speeded speech. Speeded speech has long been recognized for its potential to improve spoken media comprehension; however, current tools to significantly speed playback of non-text media are insufficient due to their reliance on inaccurate phoneme analysis. With the ever increasing amount of non-text media online, a method to speed playback that is agnostic of phonemes is needed. Our technique uses spectral and source components of the acoustics to generate a non-linear compression map that characterizes how conversational-rate speech signals are compressed to achieve analogue fast-rate speech signals. A data set containing conversational- and fast-rate speech pairs was processed to determine compression maps corresponding to each pair. A Recursive Neural Network (RNN) was trained on the set of normal-rate speech and the corresponding compression maps. The RNN was then used to generate compression maps for novel normal-rate speech and ultimately output a fast-rate speech signal.

Elicited fast-rate speech and speeded speech conversions technique are now being compared perceptually for intelligibility and naturalness.

1aSC33. Evaluation of natural language understanding based speech dialog interface's effectiveness regarding car navigation system usability performance. Takahiro Kojima (Graduated School of Information Technol., Aoyamagakuin Univ., Fuchinobe, Chuo-ku, Sagamihara-city, Kanagawa Pre., Sagamihara 251-0038, Japan, tkojima@wil-aoyama.jp), Atsunobu Kaminuma (Nissan Motor, Astugi, Japan), Naoya Isoyama, and Lopez Guillaume (Aoyamagakuin Univ., Sagamihara, Japan)

Recently, new cell phone services enable taking input by speaking to a quasi-agent using highly accurate speech recognition technologies. However, there are two problems when equipping a vehicle with these technologies. First, we do not understand yet the effect on driver's usability performance of a car navigation system equipped with spoken dialogue interface. Second, the corresponding design techniques to obtain sufficiently good navigation results has not yet been determined. In this study, we first discuss problems involving speech recognition technologies and the types of speech interface used in this study. Second, we experimentally compare in pseudo-driving conditions the effect on driver's physiology of various speech user interfaces: the Command Input (CI) interface, the Natural Language Understanding (NLU) interface, and the NLU/Spoken Dialogue Management (SDM) interface. According to the analysis of physiological signals, parasympathetic nervous system activity index value was higher when using the NLU/SDM interface than when using the CI interface. Furthermore, both subjective assessment result and time satisfaction degree of NLU/SDM interface were higher than others. Therefore, we can conclude that the NLU/SDM-type interface has better usability performance.

MONDAY MORNING, 28 NOVEMBER 2016 SOUTH PACIFIC 1, 10:30 A.M. TO 12:00 NOON

Session 1aSP

Signal Processing in Acoustics: General Topics in Signal Processing I

R. Lee Culver, Chair
ARL, Penn State University, PO Box 30, State College, PA 16804

Contributed Papers

10:30

1aSP1. Noise reduction for sound capturing by creating spatial null sensitivity point. Akio Ando, Kodai Yamauchi, and Kazuyoshi Onishi (Electric and Electronics Eng., Faculty of Eng., Univ. of Toyama, 3190 Gofuku, Toyama 930-8555, Japan, andio@eng.u-toyama.ac.jp)

We had developed a sound source separation system in which two microphones had been arranged in a line toward sound sources to form null sensitivity points at the source positions by solving the weighting coefficients for microphone outputs. The system achieved the sound source separation by excluding non-target sound sources, instead of extracting target sound sources. In this study, we propose the sound capturing system by which the user can set the null sensitivity point at the arbitrary position. The null sensitivity point is formulated as a function of the weight coefficients. The gain function is also defined as a function of a spatial position. The proposed system uses these functions to capture the target sound while suppressing the non-target sound. A computer simulation showed that, instead of a null sensitivity point, the proposed system generated the circular null sensitivity area whose center was at the microphone position.

10:45

1aSP2. Sound field interpolation in the spatial domain with a rigid spherical microphone array. Cesar D. Salvador, Shuichi Sakamoto, Jorge Trevino, and Yôiti Suzuki (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., 6-6-05, Aza-Aoba, Aramaki, Aoba-ku, Sendai 980-8579, Japan, cdsalv@gmail.com)

Sound field interpolation aims to calculate sound fields at arbitrary points from original measurements at discrete points. Rigid spherical microphone arrays are effective for interpolation because they can capture sound from several directions with uniform resolution. Interpolation from spherical array measurements is typically based on the spherical Fourier transform and assumes no prior knowledge concerning the source positions. The spherical Fourier transform, however, yields results whose accuracy strongly depends on microphone positioning and the inversion method used for its computation. When knowledge of the source positions is available, the pressure generated at any point on the rigid spherical baffle can be estimated with an existing physical model. This model was used in this study to define an analytic transfer function that relates the pressure at two arbitrary points on a rigid sphere. Based on this analytic function, an interpolation method in the spatial domain is presented as an alternative to transformed domain methods. Numerical experiments with a rigid sphere showed that accurate interpolation of magnitude and phase, over the full range of frequencies, is possible on the side of the array ipsilateral to the sound sources. However, on the contralateral side, accuracy systematically decreased as frequency increased.

11:00

1aSP3. Infinite impulse response inverse filters based on system identification with singular value decomposition. Toshiya Samejima (Faculty of Design, Kyushu Univ., 4-9-1 Shiobaru Minami-ku, Fukuoka, Fukuoka 815-8540, Japan, samejima@design.kyushu-u.ac.jp) and Kento Mitsuyasu (Shimano Inc., Sakai, Osaka, Japan)

The authors propose infinite impulse response (IIR) inverse filters for equalizing non-minimum phase acoustic plants through accurate system identification. System identification of an acoustic plant is an essential and important process in designing a controller for the plant, such as an inverse filter. In the previous paper, one of the authors has introduced the singular value decomposition (SVD) method for system identification of an enclosed sound field. The SVD method, which is the state space model-based system identification technique, involves the characteristics of experimental modal analysis. The author has verified that the SVD method has several advantages, such as high precision of identification. The proposed IIR inverse filter is simple inversion of the transfer function of the accurately identified model of an acoustic plant through the SVD method. The poles outside the unit circle in the z-plane, which are originated with the non-minimum phase zeros of the identified plant, are reflected toward the inside of the unit circle. The proposed IIR inverse filter is applied to enclosed sound field equalization. It is demonstrated that the proposed IIR inverse filter can achieve lower equalization error or lower order of the inverse filter than conventional inverse filters using the least squares method.

11:15

1aSP4. Stereo channel music signal separation based on non-negative tensor factorization with cepstrum regularization. Shogo Seki, Kento Ohtani, Tomoki Toda, and Kazuya Takeda (Nagoya Univ., Furo-cho, Chikusa-ku, Nagoya 464-8603, Japan, seki.shogo@g.sp.m.is.nagoya-u.ac.jp)

Music signals are usually generated by mixing many music source signals, such as various instrumental sounds and vocal sounds, and they are often represented as 2-channel signals (i.e., stereo channel signals). Underdetermined source separation for separating the music signals into individual music source signals is a potential technique to develop various applications, such as music transcription, singer discrimination, and vocal extraction. One of the most powerful underdetermined source separation methods is Nonnegative Matrix Factorization (NMF) that models a power spectrogram of an observation signal as a product of two nonnegative matrices; basis and activation matrices. To apply NMF to the stereo channel music signal separation, we have proposed Nonnegative Tensor Factorization (NTF) by further implementing a gain matrix to represent mixing (i.e., panning) information. However, the separation performance of this method is insufficient owing to less prior information to model acoustic characteristics of the individual music source signals. To address this issue, we propose a cepstrum regularization method for NTF to make power spectral envelopes of the separated signals close to those of individual music source signals. We conduct experimental evaluations to investigate the effectiveness of the regularization method and show remaining problems to be addressed.

11:30

1aSP5. Development of unsupervised classifier for beaked whale clicks. Natalia Sidorovskaia and Kun Li (Phys., UL Lafayette, UL BOX 44210, Lafayette, LA 70504-4210, nas@louisiana.edu)

Recent advancements in passive acoustic monitoring lead to the collection of large volumes (hundreds of Tb) of acoustic data, which are post-processed to identify various acoustic events. One of the processing goals is the identification of deep diving marine mammal calls, including different species of beaked whales (Gervais', Blainville's, Cuvier's, etc.) Traditional detectors employ a band-energy-based approach to search for acoustic events above baseline level in the click production band. However, such detectors have a large percentage of false positives and require an experienced operator to manually determine the false positive rate and subdivide the class detections (beaked whales) into species specific calls (such as Cuvier's, etc.) Modern supervised and unsupervised machine learning algorithms broadly employed for the analysis of large volumes of data may open opportunities for the development of highly accurate detectors for marine species with minimal operator involvement. Unsurpervised clustering, supervised classification, and neural networks are implemented as second stage detectors for the classification of species-specific beaked whale calls. Algorithm features, optimization, and performance are discussed. [Research supported by GOMRI.]

11:45

1aSP6. Controlling the wavefront of the de-modulated sound of a parametric loudspeaker isolated from the angle of own beam. Shigeto Takeoka (Shizuoka Inst. of Sci. and Technol., 2200-2, Toyosawa, Fukuroi, Shizuoka 437-8555, Japan, takeoka@ee.sist.ac.jp)

In this report, an array control method for de-modulated wavefront of a parametric loudspeaker is described. Parametric loudspeakers are known for a very sharp directivity due to their ultrasonic carrier wave. A desired audio signal modulated onto this carrier wave is reproduced along the beam by non-linearity in sound propagation. In general, the directivity of the output wave from array speakers is formed by the array factor, determined by the sound source arrangement and the excitation distribution. However, the output sound itself of the parametric loudspeaker is ultrasonic, and the audible sound is de-modulated along the beam secondarily. In this report, by using amplitude modulation, we present an experiment which the ultra directivity sound have individual array factors in the carrier and the modulated signal. The result shows the wavefront of the de-modulated sound is isolated from the angle of the beam.

MONDAY MORNING, 28 NOVEMBER 2016 NAUTILUS, 10:30 A.M. TO 12:00 NOON

Session 1aUW

Underwater Acoustics: Target Scattering

Kevin Williams, Chair
Applied Physics Lab., University of Washington, 1013 NE 40th St., Seattle, WA 98105

Contributed Papers

10:30

1aUW1. An investigation into bio-inspired sonar search performance. Joseph R. Edwards, Sara Beery, and Kristen E. Railey (MIT Lincoln Lab., 244 Wood St., Lexington, MA 02420, joe.edwards@ll.mit.edu)

The ability to detect, localize, and recognize objects on or under the seafloor is a critical capability for missions such as mine countermeasures (MCM), asset recovery, and feature-based navigation. Unmanned undersea vehicles (UUVs) have been increasingly tasked with such missions and are becoming more capable as technology develops. One intended application of developing UUV technology is to replace the dolphins that execute the mine-hunting task in the U.S. Marine Mammal Program. UUV development for MCM has moved toward larger vehicles with more powerful and multi-modal sensors, to the point where next-generation UUVs are significantly larger and have more powerful sensors than the dolphins they are intended to emulate. Dolphins rely on relatively modest sonar to detect and locate bottom and buried mines, but utilize it to great effect. In contrast to rigid UUV search instructions, dolphins employ a flexible approach with respect to both sonar characteristics and search trajectory. In this work, the effectiveness of biomimetic sonar search is investigated through four principal axes of adaptation: click interval, source power, source center frequency, and mobility. It is shown that the introduction of limited flexibility in the sonar search can greatly improve the overall area clearance performance.

10:45

1aUW2. Enhanced target detection and classification using two-pulse sonar methods. Nikhil Mistry, Paul White, and Timothy Leighton (Inst. of Sound & Vib. Res., Southampton Univ., Bldg. 13, Rm. 3049, University Rd., Southampton, Hampshire SO17 1BJ, United Kingdom, nm6g09@soton.ac.uk)

A number of two-pulse sonar techniques can be employed to separate linear and nonlinear scatterers. Twin Inverted Pulse Sonar (TWIPS) and Biased Pulse Summation Sonar (BiaPSS) are processes that exploit nonlinear bubble dynamics to perform such a classification, with clutter reduction. Consequently, these techniques can be used to enhance target detection in bubbly waters. TWIPS and BiaPSS rely upon bubbles being driven to large nonlinear pulsations, but this is dependent upon availability of a high-amplitude source. Previous studies could only access a narrow-band source and therefore used a low frequency so that, over the rarefaction cycle, most ocean bubbles would expand sufficiently (the size range of near-surface ocean bubbles under waves and wakes being large). However, recent access to a broadband high-power source has allowed exploitation of bubble resonances. This talk presents results of investigations using broadband linear sine sweeps, of which one is based upon the simulation of an Atlantic Bottlenose Dolphin (Tursiops truncatus) echolocation click from Capus *et al.* (2006). Insonifying a cloud of bubbles with a wide band of frequencies excites as many of those sizes at, or close to, their resonance. As such, the contrast between linear and nonlinear scatter is enhanced, improving the target detection capability.

11:00

1aUW3. Modeling the acoustic response of elastic targets in a layered medium using the coupled finite element/boundary element method. Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com), Petr Krysl (Structural Eng., Univ. of California, San Diego, La Jolla, CA), Aubrey Espana, Steve Kargl, Kevin Williams, and Dan Plotnick (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The fluid-structure interaction technique provides a paradigm for solving scattering from elastic targets embedded in a fluid by a combination of finite and boundary element methods. In this technique, the finite element method is used to compute the target's elastic response and the boundary element method with the appropriate Green's function is used to compute the field in the exterior medium. The two methods are coupled at the surface of the target by imposing the continuity of pressure and normal displacement. This results in a boundary element equation that can be used to compute the scattered field anywhere in the surrounding environment. This method reduces a finite element problem to a boundary element one with drastic reduction in the number of unknowns, which translates to a significant reduction in numerical cost. In this talk, the derivation of the technique will be outlined; the method will be applied to compute scattering from various targets, including unexploded ordnance (UXO) in complex ocean environments and the results will be compared with measurements.

11:15

1aUW4. Acoustic response of targets in a shallow water, layered, environment: Experimental observations and simple ray model results. Steven G. Kargl, Aubrey L. España, and Kevin L. Williams (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, kargl@uw.edu)

Synthetic aperture sonar (SAS) data were collected throughout May of 2014 during an experiment in St. Andrew's Bay, Panama City, FL. A brief overview is given of the SAS rail system and procedure used during the "BAYEX14" experiment. BAYEX14 was carried out in a shallow water, layered, environment, where water depth was approximately 8 m with a thin layer of mud (i.e., 0.15-0.30 m thickness) covering a sandy bottom. All targets settled into the mud and came to rest on the mud-sand interface. Scattering data from an aluminum finite cylinder, an aluminum pipe, and solid replicas of an 100-mm unexploded ordnance will be presented and compared to data taken previously in a sand-only environment and to model results. To model the scattered pressure, the ray paths are assumed to experience negligible refraction in passing through the mud layer to the mud-sand interface. However, the much higher attenuation in mud relative to water must be considered along the portion of the ray path in mud. For targets at long horizontal ranges, ray paths that interact with the water-air interface must be included to model an observed modulation of the target strength. [Research sponsored by ONR and SERDP.]

11:30

1aUW5. Stability analysis for implementing a distributed-basis transition matrix for acoustic target scattering by highly oblate elastic objects in free-field. Raymond Lim (Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave., Code X11, Panama City, FL, FL 32407-7001, raymond.lim@navy.mil)

A variant of Waterman's transition (T) matrix utilizing an ansatz for problematic outgoing basis functions in standard formulations was proposed and demonstrated in a previous meeting to improve the stability of free-field acoustic scattering calculations with highly oblate axisymmetric elastic objects. The ansatz replaced the basis causing instability with a non-local basis consisting of low-order spherical functions made into a complete set by analytically continuing and distributing them along the imaginary axis of the axial coordinate. While this ansatz was found to produce scattering predictions of comparable stability to a full spheroidal-basis implementation of the T matrix, instabilities of unknown origin eventually appeared as higher matrix truncations were used to obtain convergence at higher frequencies. Here, we present a study to uncover the origin of these instabilities and show results of alternative non-local basis ansatzes chosen to attempt remediation. [Work supported by ONR.]

11:45

1aUW6. Acoustic response of an unexploded ordnance in shallow water, layered, environments: Predictions using hybrid and coupled modeling techniques. Aubrey L. Espana (Acoust. Dept., Appl. Phys. Lab. - Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105, aespana@apl.washington.edu), Ahmad T. Abawi (HLS Res., Inc., San Diego, CA), Steven G. Kargl, and Kevin L. Williams (Acoust. Dept., Appl. Phys. Lab. - Univ. of Washington, Seattle, WA)

Previously, a hybrid, axisymmetric, finite element (FE)—propagation model has shown utility in simulating the response of objects in contact with a flat, undisturbed sand/water interface [Espana et al., J. Acoust. Soc. Am. **136**, 190-124, (2014)]. Here, we offer a three-way comparison between data, the hybrid FE/propagation model and a coupled finite element/boundary element model (Coupled FEBE) for the scattering from a replica of a 100-mm unexploded ordnance (UXO) proud, partially buried and fully buried in a sand sediment for geometries where reflection from and refraction into the sand dominate. This comparison offers a more in-depth analysis of the implications of various approximations of the hybrid model, such as the effect of improper fluid loading on the target surface in the partially buried situation. These results will be contrasted to the more complicated mud-over-sand layered environment of the BAYEX14 experiment. Preliminary model results will be compared to data, giving further insight into the dominant physical mechanisms affecting the UXO's acoustic response in the more complicated layered environment. [Research sponsored by ONR and SERDP.]

Session 1pAAa

Architectural Acoustics: Acoustics for Children and Pupils II

Keiji Kawai, Cochair
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David S. Woolworth, Cochair
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David Lubman, Cochair
DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683-4514

Chair's Introduction—1:00

Invited Papers

1:05

1pAAa1. Need of good classroom acoustics since the early childhood. Arianna Astolfi, Giuseppina E. Puglisi (Dept. of Energy, Politecnico di Torino, Corso DC degli Abruzzi, 24, Turin 10124, Italy, arianna.astolfi@polito.it), Andrea Prato (National Inst. of Metrological Res., Turin, Italy), Louena Shtrepi (Dept. of Energy, Politecnico di Torino, Turin, Italy), Benedetto Sacchetti, and Tiziana Sacco (Dept. of Neurosci., Univ. of Turin, Turin, Italy)

In everyday lives, the listening experiences often occur in adverse listening environments in which children need to tune out competing sounds to tune into speech. In the early childhood and in the first years of education, a good acoustic environment is mandatory in order to avoid difficulties in literacy development. This is true both for normal children and for children with potential learning disabilities, which can be only diagnosed after this age-period. Whenever children with poor neural processing in speech discrimination are exposed to bad acoustics, they may fall behind their peers in reading development. On the contrary, since the neuroplasticity of the auditory cortex of the human brain is up to about the age of eight years, when at-risk children are trained with learning programs based on sound and visual cues recognition in proper listening environments, the effectiveness of these programs is improved. This work focuses on the evidence of the need of good classroom acoustics by the early stage of education. Preliminary results on a study on the influence of classroom acoustics on the reading skills in the early childhood are reported. The study involved 120 second-grade pupils in three primary schools with different classroom acoustics.

1:25

1pAAa2. Relation between children's calmness and sound environment in nursery field: Influence of environmental settings and childcare practice. Saki Noguchi and Kanako Ueno (Meiji Univ., 1-1-1, Higashimita, Kawasaki, Kanagawa 214-8571, Japan, n-saki@akane.waseda.jp)

In Japanese childcare facilities, there has conventionally been a serious problem regarding too noisy environment in nursery room. This becomes even more serious in an urban area having high population density, particularly because of lack of childcare space. It is concerning that sound environment in childcare facilities, where children live for approximately 8-11 h, has a great influence on the mental and physical development of children. The relation between children's calmness, such as a state of relaxation or concentration on activity, and the sound environment is reported herein, with a view to creating a suitable sound environment for children. We explain this relation in terms of environmental factors and childcare practices using the following cases. (1) Experiment conducted by varying the environmental settings— particularly, spatial reverberation and furniture of play corner; (2) observation of the childcare practice attached great importance to create the calm of sound environment. From the result, we report the influence that sound environment has on children's calmness and discuss methods to ensure that the sound environment supports healthy development of children.

1:45

1pAAa3. Modeling of venues for music practice and instruction. Ana M. Jaramillo and Bruce C. Olson (Olson Sound Design, LLC, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu)

Music practice rooms for large groups are spaces with conflicting requirements, especially those that are part of schools. The need for an environment that is similar to the ones where the musicians will be performing (appropriate reverberation for music performance), and on the other hand, the need for music instruction (speech intelligibility for teacher/student and student/student communication) are usually opposing. As part of a renovation of band rooms in two schools, we were called to advise on the room acoustics. The rooms were having floor replacement, from carpet to tile and needed to have the room acoustics reviewed and adjusted for the new floor material with added wall or ceiling absorption to replace what was lost from the floor.

2:05

1pAAa4. Experimental measurements of word intelligibility among pre-school children under differing conditions of reverberation and signal-noise ratio. Kazunori Harada and Keiji Kawai (Graduate School of Sci. and Technol., Kumamoto Univ., 2-39-1 Kurokami, Chuo-ku, Kumamoto 860-8555, Japan, kazunori.harada.108@gmail.com)

Many children, from 0 to 5 years old, spend most of their active hours in child day-care centers. Rooms in these centers are often very noisy, due to both the voices of children or sounds of various activities and to the excessive reverberation attributable to the little use of sound-absorbing materials. Some researchers have studied the speech intelligibility of elementary school children, but we were unable to find those of pre-school children. Thus, in this study, an experiment was carried out a word intelligibility test on children from 3 to 5 years old in a classroom of a child day-care center. The procedure was like a game. Each of the presented words was mixed with pink noise at one of two S/N ratios, and convolved with the impulse responses of three reverberation times of 0.3, 0.9, and 1.8 s. Children were asked to judge whether the word was food or not, and to raise their hands holding yes or no signs. As a result, the correct answer ratio for 3-year-old children was lower than the ratio for other groups. Also, the correct answer ratio for 4- and 5-year-old children was slightly lower than control groups of elementary school pupils and college students.

2:20

1pAAa5. Effect of design on acoustic performances in classrooms: Case study from Ankara. Filiz B. Kocyigit (Faculty of FAD and Architecture, Dept. of Architecture, Atilim Univ., Kizilcasar Mah. Incek, Ankara 06380, Turkey, filizbk@gmail.com)

Education is one of the most important factors for the development of societies. Efficient education can be possible with a qualified communication. Current spaces are huge factor that effects communications. No doubt that auditory comfort in spaces with completed acoustic development provides better communication. This study is prepared to propound the effects of the acoustic comfort conditions in classrooms which are the educational structure's most important spaces and the material used. Within the scope of this study, two schools which are private and government high school in Ankara situated in Cankaya Municipality have been chosen to make measurements. Selected schools have been researched by the classes' volume in terms of compliance with acoustic and in accordance with the inner surface of equipment. Similar volume classes chosen in both schools and they are measured with acoustic measuring devices, RT, EDT, C80, D50, STI, and RASTI which are significant for the intelligibility of speech has been taken into consideration for evaluation of the measurements taken in the determined classes. The impacts of furniture and surface materials on the acoustic comfort conditions are recovered in renewed or newly designed classrooms for revealing the most ideal level of speech intelligibility in classes.

2:35

1pAAa6. Effect of music on Turkish medical schools' architectural design in the Seljuk and Ottoman period. Filiz B. Kocyigit (Faculty of Fine Art Design and Architecture, Atilim Univ., Kizilcasar Mah. Incek, Ankara 06380, Turkey, filizbk@gmail.com)

Anatolian people throughout the ages have given importance to art and science. Music has been one of the most important part of life in Ottoman, Anatolian Seljuk Turks. Music therapy has been used with not only traditional medicine but also has been used with modern therapy. It has different historical progresses according to the cultures, nations, and periods, from being a myth to a treatment method. Hospitals' design methods for music therapy have great importance for treatment. Because of the immigration of Turkish people in 11-16th century, different Anatolian works like architectural art and institution affected the other countries. Among these works, we try to state some charity foundations all rendered for the use of the public such as temples, madrasahs, schools, dervish lodges, libraries, and cottage houses. The importance of "Music Therapy" is to show its historical progress, the effects on human being, and to exist the place it takes in psychology science. In this paper, the historical progress of music therapy in Turks is mentioned and information about the hospitals that used "music therapy" in Seljukian and Ottoman era is given, as they were the pioneers for architectural design of a hospital in all over the world.

Session 1pAAb

Architectural Acoustics: Acoustical Design Metric Comparison Between the U.S. and Japan

David Manley, Cochair
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Hayato Sato, Cochair
Architecture Engineering, 1-1 Rokkodai, Nada, Kobe 6578501, Japan

Chair's Introduction—3:05

Invited Papers

3:10

1pAAb1. Metrics for acoustical measurements in North America. Robert A. Hallman (Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604, rahallman@armstrongceilings.com)

The primary source of standards for acoustical measurements in buildings and community noise for North America is ASTM E33 on Building and Environmental Acoustics. This paper focuses on the ASTM E33 standards and the metrics generated. There are six major categories of standards, Sound Absorption, Speech Privacy, Sound Transmission, Application of Materials and Systems, Mechanical and Electrical Noise, and Community Noise. The most familiar metrics include NRC, STC, and IIC, but there is an alphabet soup of metrics including AC, AI, AOITC(θ), ANISPL, ATL, CNEL, FSTC, NIC, and OINIC. Many of these are variations of the basic measurements targeting a specific measurement environment. This paper focuses on the basic measurements and how the metrics are calculated and then gives a broad overview of the range of variations and specialized test methods available through ASTM Committee E33 standards.

3:30

1pAAb2. Japanese standards of measurement method and rating for floor impact sound insulation. Atsuo Hiramitsu (National Inst. of Land Infrastructure Management, 1, Tachihara, Tsukuba-City, Ibaraki 305-0802, Japan, hiramitsu-a92ta@nilim.go.jp)

Among the objections and the troubles of buildings in Japan, the floor impact sound is one of the most serious issues. The floor impact sounds are classified in a heavy- and a light-weight floor impact sound by the characteristic of the impact source. One is the floor impact sound caused by human jumping, running, or walking and the other is the floor impact sound caused by falling lightweight object. The measurement and rating method for floor impact sound insulation is standardized in JIS (Japanese Industrial Standard). JIS A 1418-2 is method using standard heavy/soft impact sources, a car-tire source (a bang machine) and a rubber ball source. The heavy-weight floor impact sound insulation is started in Japan. JIS A 1418-1, which is corresponded to ISO140-7, is method using standard light impact source, a tapping machine. The evaluation for the floor impact sound insulation is standardized in JIS A 1419-2 which is corresponded to ISO 717-2. This paper summarizes the trends for the floor impact sound insulation in Japan. The single-number quantity "L_r" is usually used for the rating of the floor impact sound insulation from the octave band sound pressure level. Moreover, A-weighted floor impact sound level is used recently.

3:50

1pAAb3. Comparison of laboratory building acoustics standards from ASTM International, Japanese Industrial Standards Committee, and the International Organization for Standardization. Matthew V. Golden (Pliteq, 616 4th St., NE, Washington, DC 20002, mgolden@pliteq.com)

Testing standards are fundamental to the practice of building acoustics. To that end, ASTM International, the Japanese Industrial Standards (JIS) Committee, and the International Organization for Standardization (ISO) all have standards that cover the laboratory measurement of airborne and impact sound transmission. These standards include ASTM E90 and E492, JIS A 1416, and A 1418 and ISO 10140. This paper will review the theory behind each methods and the derivation of the equations used. It will also discuss each of these standards in detail, including the relative strengths and weaknesses of each as well as the interoperability the resulting data. Previous research comparing the standards will also be reviewed. Finally, a brief overview of the repeatability and reproducibility of all building acoustics standards will be discussed.

4:10

1pAAb4. Test and rating method comparison between Japan and the U.S. for airborne sound insulation in buildings. Kiyoshi Masuda (Technol. Ctr., Taisei Corp., 344-1 Nase-cho, Totsuka-ku, Yokohama, Kanagawa 245-0051, Japan, kiyoshi.masuda@sakura.taisei.co.jp)

The Japanese test methods for laboratory measurement of airborne sound transmission loss for building partitions are basically the same as the American test methods. However, the two countries' field test methods for airborne sound insulation in buildings differ. The octave band sound pressure level difference between two rooms is the most significant value for the field evaluation of sound insulation in Japan, whereas the apparent transmission loss for a partition installed in a building is the most popular value for the field evaluation of sound insulation in the US. Why is the octave band sound pressure level difference popular in Japan? It is the basic way of thinking in Japan that an acoustical engineer should take responsibility for the total spatial sound insulation performance, including sound absorption and flanking transmission from the design stage to completion and the field test and evaluation method as the completion inspection should be simple. If the insulation performance does not meet the specific criterion at the time of its completion, it is necessary for the acoustical engineer to demonstrate the cause of the insulation defect using various measurement techniques, including test methods similar to FTL in ASTM E336.

Contributed Papers

4:30

1pAAb5. Reverberation time versus volume for active acoustic systems. Roger W. Schwenke and Tobi Szuts (Meyer Sound Labs., 2832 San Pablo Ave., Berkeley, CA 94702, rogers@meyersound.com)

If two rooms have the same reverberation time but different cubic volumes, then they will differ in acoustic strength. Many authors have presented recommendations for reverberation time as a function of volume for different purposes. There are many parameters that, if held constant, yield different reverberation times for different cubic volumes: strength, clarity, critical distance, etc. For example, the ITU-R BS.1116 recommendation for reverberation time versus volume corresponds to rooms of constant average absorption per square meter. This paper will review the literature on reverberation time versus volume, show the connection between STI and reverberation versus volume, and propose a recommendation for reverberation versus volume for active acoustic systems.

4:45

1pAAb6. Acoustic design method of multifunctional small venues. Hideo Miyazaki (YAMAHA Corp., 10-1 Nakazawa-cho, Naka-ku, Hamamatsu 430-8650, Japan, hideo.miyazaki@music.yamaha.com)

Looking at the tendency of resent acoustic design projects in Japan, the number of venues specialized for a certain usage like concert halls is decreasing while the significant differences of the number of multi-purpose venues are not seen between in 1990s and 2000s. On the other hand, regarding the size (seating capacity), the number of over 1,000 seats multi-purpose venues are decreasing and that of below 1000 seats (especially, small venues below 500 seats) tends to be increasing. One of the reasons for this tendency is the finance problem at local governments. Another reason is the background that venues used by local citizens has been focused rather than large venues for touring performances under the strategy to activate the local cities. In this report, the acoustic design examples of multifunctional small venues are introduced while categorized by their design methods.

MONDAY AFTERNOON, 28 NOVEMBER 2016 — SOUTH PACIFIC 4, 1:30 P.M. TO 4:50 P.M.

Session 1pAB

Animal Bioacoustics: Bat Echolocation: New Insights on Biosonar Production, Processing, and Performance from Field and Laboratory Investigations II

Hiroshi Riquimaroux, Cochair
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Laura Kloepper, Cochair
Biology, Saint Mary's College, 262 Science Hall, Saint Mary's College, Notre Dame, IN 46556

Invited Papers

1:30

1pAB1. Significance of terminal frequency in FM sweeps for bat echolocation. Hiroshi Riquimaroux (SDU-VT Int.. Lab., Shandong Univ., 27 Shanda Nanlu, Jinan, Shandong 250100, China, hiroshi_riquimaroux@brown.edu)

For echolocation of CF-FM bats, the CF component is expected to detect wing-beat of flying insects. However, it also plays an important role to serve as a stable signal carrier from a distant target. Then, once the carrier frequency is removed, echo signal can be easily detected. For this purpose, Doppler-shift compensation and echo amplitude compensation must be conducted to stabilize the signal

carrier. What about the FM bats? FM bats also appear to utilize a relatively long pseudo-CF component to detect wing-beats of flying insects. They might also utilize pseudo-CF component for an echo signal carrier. However, Doppler-shift compensation in FM bats has not been reported. In the inferior colliculus, majority of neurons are tuned to the pseudo-CF frequency, 20 kHz for *Eptesicus fuscus* while 40 kHz for *Pipistrellus abramus*, which are corresponding to the terminal frequencies of downward FM sweep. Overrepresentation of a particular frequency may enhance frequency discrimination, detection of insect wing-beat. Therefore, we may expect majority of cochlear nerve fibers are also tuned to the pseudo-CF frequency. However, overrepresentation of hair cells for a particular frequency does not appear to be shown. [Research supported by MEXT, Japan and Shandong University, China.]

1:50

1pAB2. Midbrain neurons of the free-flying echolocating bat represent three-dimensional space. Ninad B. Kothari, Melville J. Wohlgemuth, and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., 3400 N. Charles St., Ames Hall 200B, Baltimore, MD 21218, cynthia.moss@jhu.edu)

Spatial navigation by echolocation in bats is an active and adaptive system: Its success depends upon tight coupling between motor commands for sonar signal production and neural processing that supports spatial perception and attention to objects in the 3D environment. The midbrain superior colliculus (SC) has been implicated in sensorimotor transformations to support adaptive behaviors in the echolocating bat, but the response properties of SC neurons have yet to be studied in free-flying animals engaged in natural spatial navigation behaviors. Using RF telemetry, in combination with a model that computes the direction and arrival time of echoes received at the bat's ears, we have successfully characterized the auditory response profiles of single neurons in the SC of the free-flying big brown bat. Neural data, acquired across intermediate and deep layers of the bat SC, were sorted off-line from multichannel silicon array recordings. Spikes evoked by echo returns were analyzed and used to construct 3D spatial response profiles of over 40 SC neurons in two bats. Echo response profiles showed selectivity to both the direction and delay of sonar returns over biologically relevant ranges. This work opens the door to investigate the contribution of spatial attention to 3D auditory tuning.

2:10

1pAB3. Signal characteristics and echolocation challenges of Mexican free-tailed bats during high-speed flight. Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, lkloepper@saintmarys.edu) and Meike Linnenschmidt (Div. of Neurobiology, Ludwig-Maximilians-Univ., Munich, Germany)

Female Mexican free-tailed bats (*Tadarida brasiliensis*) form large maternal colonies numbering up to several million. After their nightly emergence and feeding, they return to the roost from high elevations and speeds nearing 100 km/h. At these speeds, the bats should experience, among other things, difficulties in object distance encoding due to the large distance travelled before echo returns and significant Doppler shifts to their echoes. Further, they face navigational challenges from nearby conspecifics. In this study, we characterized the acoustics of returning Mexican free-tailed bats and investigated for possible Doppler shift compensation during various flight speeds. Synchronized acoustic and thermal imagery recordings were conducted at several caves across multiple mornings during cave re-entry. Call sequences for individual bats were extracted for different flight speeds, and changes in call parameters were compared within each call sequence. First basic analysis indicates a possible lack of Doppler shift compensation during fast and steep re-entry flights. Further, these bats demonstrate a flexible signal design and extreme behavioral maneuvers in flight.

2:30

1pAB4. Auditory-visual information transfer in echolocating Egyptian fruit bats. Corinna Schilling (Neurobiology, Univ. of Ulm, Ulm, Germany), Alan Grinnell, and Walter Metzner (IBP, UCLA, 621 Ch. E. Young Dr. S., Los Angeles, CA 90095-1606, metzner@ucla.edu)

Egyptian fruit bats have good vision but can also accurately echolocate relying on their well-developed auditory system. We asked two questions: 1. Can these bats acquire a mental image of an object using vision (or hearing) alone? 2. If such a mental image has been formed, can it be transferred from one sensory modality (e.g., vision) to another (e.g., hearing)? We trained the bats to first localize a rewarded object (an X) that was presented pseudo-randomly at either the left or right side using either vision (group 1) or hearing (group 2) alone. After only a few weeks, all experimental bats were able to perform these tasks. We then had the bats discriminate between a rewarded object (the X) and an unrewarded object (a circle) using either vision (group 1) or hearing (group 2) alone. Again, all bats mastered this task after only a few weeks of training. Finally, we tested if the same bats could also discriminate the objects using the other sensory modality (group 1: hearing; group 2: vision). After only a few sessions, the bats performed correctly suggesting that the bats can indeed transform a mental image that they obtained using one sensory modality to another modality.

2:50–3:05 Break

Contributed Papers

3:05

1pAB5. Dynamic periphery in a biomimetic sonar system introduces time-variant signatures into targets echoes. Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., ICTAS Life Sci. District (Mail Code 0917), Blacksburg, VA 24061, josephs7@vt.edu), Philip Caspers, and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

Bats have developed unique and refined systems of echolocation throughout the course of their evolutionary history, giving them the ability to navigate and hunt in extremely cluttered environments. While the mechanisms behind many of these abilities remain unknown, it has been observed that the most effective biosonar systems in bats use a variety of dynamic sensing mechanisms. One conspicuous manifestation of this dynamics can be seen in changes to the shapes of the baffle that diffract the emitted biosonar pulses (noseleaves) and the returning echoes (outer ears). Using numerical predictions as well as measurements with biomimetic hardware, our own prior work has established that the dynamics in these baffles can create time-variant emitter and receiver characteristics. However, it has yet to be demonstrated that these time-variant device characteristics have a substantial impact on the received echoes. To address this question, a biomimetic sonar head with dynamic emission and reception baffles was used to ensonify a range of different targets that included simple geometrical shapes (e.g., sphere, cylinder, and cube) as well as natural targets (foliage). The biomimetic dynamics of the sonar head resulted in time-variant signature in the received echoes that could provide a substrate for encoding additional sensory information.

3:20

1pAB6. Quantitative analysis of rigid and non-rigid motion patterns in rhinolophid and hipposiderid pinnae. Xiaoyan Yin, Peiwen Qiu, Mengna Zhang (Shandong Univ. - Virginia Tech Int.. Lab., Shandong Univ., Shanda South Rd. 27, Jinan, Shandong 250100, China, yinxiaoyan4@gmail.com), and Rolf Müller (Shandong Univ. - Virginia Tech Int.. Lab., Shandong Univ., Blacksburg, VA)

It has long been noted that horseshoe bats (Rhinolophidae) exhibit large ear motions during biosonar behaviors. Some authors have regarded these motions as rigid rotations of the pinna whereas others have pointed out that the animals are capable of non-rigid motions where the shape of the pinna changes. In the current work, we have analyzed pinna motions in greater horseshoe bats (Rhinolophus ferrumequinum) and the related great roundleaf bat (Hipposideros armiger) to investigate the issue of rigid versus non-rigid pinna motions. The bats were found to produce rigid as well as non-rigid motions. The latter were characterized as "open-close" motions where the size of the pinna aperture was changed by varying the distance between the anterior and posterior rims of the pinna. To quantify this, pinna motions were described by three-dimensional tracking of points on the pinna rims using high-speed stereo vision. The point coordinates were used to compute a matrix that contained the maximum change in distance between any pairs of points on the anterior and posterior pinna rims. Dimensionality reduction methods (e.g., linear discriminant analysis) were applied to the vectors containing all elements of these "maximum distance change matrices" to quantify the relationship between rigid and non-rigid motions.

3:35

1pAB7. Integration of emitter and receiver dynamics in a biomimetic sonar head. Philip Caspers and Rolf Müller (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061, pcaspers@vt.edu)

Horseshoe bats (family Rhinolophidae) transmit and receive sonar information about the environment using morphologically complex and dynamic noseleaf and pinnae. Both local shape features and dynamic deformations of the noseleaf and pinnae contribute to properties of the bat biosonar system. This work systematically explores the effect of local shape features on the dynamic properties of emission and reception baffles using a biomimetic robotic system inspired by the horseshoe bat. Parametric models of the greater horseshoe bat (*Rhinolophus ferrumequinum*) noseleaf and pinnae with selectable combinations of local shape features were developed. Baffles with feature realizations were cast out of flexible silicone and mounted to a platform to dynamically actuate the emission and reception baffle surfaces with motions patterns similar to the greater horseshoe bat. Motions of the baffle surfaces were synchronized to the outgoing and incoming sonar waveform, and the time-frequency properties of the emission and reception baffle realizations were systematically characterized across spatial direction—individually and jointly. Different feature combinations of the noseleaf and pinnae local shape features were ranked for overall dynamic effect. It was found shape and motion patterns impacted the overall dynamics with relative strength over time, frequency, and spatial direction.

3:50

1pAB8. Challenges of acoustic monitoring of bats—A biosonar perspective. Jens C. Koblitz (BioAcoust. Network, Eichenallee 32 a, Neuss 41469, Germany, Jens.Koblitz@web.de) and Peter Stilz (BioAcoust. Network, Hechingen, Germany)

Acoustic monitoring of bats is increasingly used in biodiversity assessments and population monitoring. Besides accurate species identification, additional factors make it challenging to derive population trends, yet—sizes based on acoustic monitoring. Inter- and intra-species as well as individual variation of acoustic parameters and acoustic activity result in varying detection probability. Changes in environmental conditions result in a large changes in the volume monitored by the device. Differences in the devices used for acoustic monitoring make it inherently difficult to compare data collected with different devices. By broadcasting bat echolocation calls from various distances to monitoring devices, the acoustic parameters influencing the successful detection of a call were examined. A microphone array was used to track bats in the vicinity of monitoring devices and the distance between device and bat was measured for each call based on the time of arrival difference. The acoustic detection function, the probability of detecting calls as a function of distance, was then derived for multiple detector types.

4:05

1pAB9. Do echo flow patterns guide flight behavior of the big brown bat? Michaela Warnecke and Cynthia F. Moss (Dept. of Psychol. and Brain Sci., Johns Hopkins Univ., 3400 N Charles St., Baltimore, MD 21218, warnecke@jhu.edu)

When animals move through the environment, they receive dynamic sensory information from surrounding objects. Past research has demonstrated that visually guided animals rely on optic flow to estimate their relative velocity and distance to objects. More recently, the flight and echolocation behavior of big brown bats was studied in a corridor constructed of poles of variable spacing. When the pole spacing on opposite walls was symmetric, animals centered themselves in the corridor, and when pole spacing was asymmetric, bats steered toward the less echoic wall. This finding raised the question whether bats adjusted their flight paths based on the flow of echoes returning from the corridor walls, or whether the animals steered away from the wall reflecting more intense echoes. To address this question, bats flew through the original corridor with the additional experimental condition of felt coverage on one corridor wall, reducing the reflectivity of pole echoes on that side, while retaining the flow pattern. If the bat's flight path is influenced by the echo intensity profile of a corridor wall, it should deviate toward the felt-covered wall. By contrast, if the bat's behavior is influenced by acoustic flow cues, flight paths should resemble those in the original (no-felt) conditions.

4:20

1pAB10. Jamming avoidance by the echolocating bat during flight with multiple conspecifics. Kazuma Hase, Yukimi Kadoya (Graduate School of Life and Medical Sci., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, emq1003@mail4.doshisha.ac.jp), Yosuke Maitani (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan), Kohta I. Kobayasi, and Shizuko Hiryu (Graduate School of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan)

It remains mystery how echolocating bats extract their own echoes from an acoustic jamming caused by vocalizations of multiple conspecifics flying together. Here, we first successfully recorded echolocation pulses of each animal during four-bat group flight by individually mounted on-board microphones, allowing us to assess vocalization in a jamming environment. We used *Miniopterus fuliginosus* which emit downward frequency-modulated (FM) ultrasounds. The bats experienced both single and group flights in an experimental chamber. As a result, each bat of the group significantly expanded differences in terminal frequency (TF) of pulses from the neighboring bat from 0.6 ± 0.6 (mean $\pm$ SD) kHz in single flight to 1.2 ± 0.6 kHz in group flight (Tukey's HSD test, $P < 0.05$). Although the difference in TF observed in group flight was much smaller than the bandwidth of pulses (approximately 40 kHz), computation of cross-correlation providing an index of similarity of two signals demonstrated that the difference is useful to distinguish own signal from others. These results suggest that innate FM signal of bats is jamming-tolerant, and FM bats flying in group highlight differences in TF to avoid or reduce jamming from conspecifics in a closed space, as implied by previous studies.

4:35

1pAB11. Obstacle avoidance navigation by echolocating bat: Compared relationships between acoustic guidance and flight control in unfamiliar space and familiar space. Yuya Yamamoto, Yasufumi Yamada, Kohta I. Kobayasi, and Shizuko Hiryu (Graduate school of Life and medical Sci., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, dmq1050@mail4.doshisha.ac.jp)

Echolocating bats can fly without colliding with obstacles by synchronizing acoustic guidance and flight control. To research how bats adapt echolocation as they became familiar with the spaces, we made CF-FM bats (Japanese horseshoe bat) exhibit S-sharp flight repeatedly in a chamber arranged acoustically transparent chains. We found (1) flight path was changed continuously to reduce its curvature as the bats repeated the flights, resulting that the maximum flight speed increased from 2.7 to 3.0 m/s in the 1st trial to 3.2 to 3.8 m/s in the 12th (last) trial for all three bats, (2) pulse emission was reduced by 45% (66/148 pulses) from the first to the last trial, (3) they shifted the pulse direction dynamically relative to their flight direction in the 1st flight whereas the pulse direction was shifted smoothly, the bats emitted more intense pulses toward the intended flight direction as they became familiar with the space. When acrylic boards were arranged, the bats could not pass through the obstacle course until they learned the space. These suggest echolocation is changed between unfamiliar and familiar space flight and the bats adapt their echolocation to feedforward-dominant control for perceiving more at distance for path planning when they have a spatial map.

MONDAY AFTERNOON, 28 NOVEMBER 2016 CORAL 2, 1:10 P.M. TO 4:40 P.M.

Session 1pAO

Acoustical Oceanography: Twenty-Five Years of Acoustical Oceanography in the ASA I

Andone C. Lavery, Cochair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, MA 02536

Michael J. Buckingham, Cochair
Scripps Institution of Oceanography, University of California, San Diego, 9500 Gilman Drive, La Jolla, CA 92093-0238

Chair's Introduction—1:10

Invited Papers

1:20

1pAO1. Twenty five happy years in ocean acoustics. Walter Munk (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, wmunk@ucsd.edu)

25 happy years in ocean acoustics

1:40

1pAO2. Twenty five years of learning what acoustics can do for oceanographers. David M. Farmer (9860 W. Saanack Rd., 9500 Gilman Dr., La Jolla, California 92093-0238, Farmer.DavidM@gmail.com)

From cracking ice and breaking surface waves to stratified flows and turbulence, the ocean reveals its secrets through underwater sound. Some personal experiences will illustrate the remarkable opportunities that flow from collaboration between oceanographers and ocean acousticians.

2:00

1pAO3. Ocean acoustic tomography: Fortieth anniversary, 1976–2016. Peter F. Worcester and Walter H. Munk (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu)

The year 1976 saw the publication of "Monitoring the ocean acoustically" by Munk and Worcester [*Science, Technology, and The Modern Navy, Thirtieth Anniversary 1946-1976*, 497-508] and arguably the first ocean acoustic tomography experiment. The field got off to a stormy start when a reviewer of an early proposal wrote that "travel times along ray paths are meaningless in a saturated environment," but a 1978 experiment showed that ray arrivals could be "resolved, identified, and tracked" at ~900 km range, just in time to save the field from an early demise. The need to measure and predict travel times with millisecond accuracy pushed the field of underwater acoustics to develop the requisite instrumentation and mathematical tools to accurately characterize the impulse response of the ocean. Important contributions to our understanding of the oceans include (1) the demonstration that internal tides are coherently generated by surface tides over the Hawaiian submarine ridge (and other similar benthic features) and can be traced for thousands of km to distant regions of dissipation, (2) measurements of oceanic deep convection, and (3) the discovery using trans-Arctic transmissions that the heat content in the Arctic was increasing. In the future, ocean acoustic tomography will likely be one component of multipurpose acoustic systems for acoustic remote sensing, navigation, communications, and passive listening, with special applicability in ice-covered regions.

2:20

1pAO4. Shallow water acoustics and oceanography at the Woods Hole Oceanographic Institution over the last quarter century. James Lynch, Timothy F. Duda, Ying-tsong Lin, and Arthur Newhall (Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543, jlynch@whoi.edu)

Shallow water (coastal) acoustics has been a topic of great interest over quarter century, both from the point of view of learning coastal oceanography and of learning coastal acoustics. Indeed, the two are difficult to separate. In this talk, the research done at the Woods Hole Oceanographic Institution (very often in collaboration with other institutions) over the past 25 years will be discussed, with an emphasis on the major experimental efforts that have been fielded, but also including theoretical and computational efforts. The Barents Sea Polar Front experiment, the Shallow Water Acoustic Random Medium experiment, the Shelfbreak PRIMER experiment, the Asian Seas International Acoustics Experiment, the Shallow Water 2006 experiment, and the Quantifying, Predicting and Exploiting Uncertainty program are some of the at-sea efforts to be discussed. Theoretical advances in shallow water acoustics, as well as computational efforts, such as the Integrated Ocean Dynamics and Acoustics project, will also be treated. Directions for future work will be discussed.

2:40

1pAO5. The enigma of the ocean internal wave spectrum: Can acoustics help solve the enigma? John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

In 1979, Carl Wunsch famously asserted "Many sources for the internal wave field have been proposed; but it has not yet been possible to make the kind of statement that can be made about surface waves: namely, when the wind blows surface waves are generated and the larger the fetch and duration, the larger the waves." In spite of decades of work using traditional oceanographic tools (e.g., acoustic doppler current profilers and conductivity, temperature, and depth devices), this statement remains essentially true today and can be equally applied to the question of internal wave sinks (dissipation), and more generally the form of the spectrum. It is therefore possible that acoustic propagation methodologies with their unique sampling properties in frequency-wavenumber space and spatial averaging could shed some light on this enigma. Several ways forward on this problem will be discussed.

3:00

1pAO6. Inspirations of past leaders in Acoustical Oceanography: Reflections of our roots in AO and bioacoustics. Timothy K. Stanton (Dept. Appl. Ocean. Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, tstanton@whoi.edu)

I had the privilege of knowing the leaders of Acoustical Oceanography (AO) during its early years, first as a topical entity, then when it was eventually formalized as a Technical Committee (TC). I first arrived at the University of Wisconsin in 1980 and did research there with Clarence Clay for eight years as a scientist. The 1977 book, "Acoustical Oceanography" by Clay and (Hank) Medwin, had recently been published and was required reading. At around the same time, Van Holliday, David Farmer, Mike Buckingham, and Medwin, established AO as a formal committee, first as a Specialty Group (Medwin as Chair), and later as a TC in 1991. Buckingham was elected first Chair of the AOTC and, given the anticipated challenges in starting a new TC, I was made Alternate Chair to work alongside Buckingham on many things. Clay and Holliday inspired much of the AO bioacoustics research that I have conducted, including developing acoustic scattering models, physics-based echo statistics, and broadband acoustic methods to use sound to characterize marine organisms such as fish and zooplankton. I will tell some stories and give examples of recent scientific advances that are traceable to Clay and Holliday.

3:20

1pAO7. Waveguides and whales: A review about using acoustic waveguide theory for localizing whales and inverting for their propagation environment. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu)

Low-frequency acoustic propagation in shallow-water waveguides is conveniently modeled using sets of normal modes that propagate with different group velocities. Analytical solutions of simple environmental models, such as the Pekeris waveguide, were almost immediately applied to estimate ranges of impulsive sources by measuring the relative arrival times of the modal components, but to the author's knowledge, it was not until the early 1990s that such techniques were first applied to low-frequency baleen whale calls, in a technical report by Gerald D'Spain. Since then waveguide-based methods have been used to track blue whales off Southern California, right whales in the Bering Sea, sei whales in the Atlantic, and bowhead whales in the Chukchi and Beaufort Seas. Some of these

applications have exploited time-separated modal arrivals on a single hydrophone, while others have used vertical arrays to measure modal interference patterns, and subsequently estimate an animal's range, depth, and local propagation environment. A recent promising development has been the introduction of nonlinear time sampling methods that permit mode isolation without the need for vertical arrays, enabling range estimation of certain whale signals from existing single-hydrophone data, even when the modal arrivals are not cleanly separated in time.

3:40

1pAO8. Passive acoustical oceanography. Martin Siderius (ECE Dept., Portland State Univ., P.O. Box 751, Portland, OR 97207, siderius@pdx.edu)

Acoustical Oceanography includes the use of both active and passive acoustic systems to measure various properties of the ocean environment. Active methods use a specifically designed and controlled sound projector while passive methods exploit naturally occurring ocean sounds such as breaking waves, ship noise, or marine mammal vocalizations. Historically, active methods have been the preferred choice when precise propagation times or calibrated sound levels are important. However, passive sensing has a strong appeal due in part to the ease in which listening-only systems can be deployed. There have been a number of advances in passive sensing even for cases where propagation times and sound levels are of interest. For example, it has been shown that surface wave noise can be used to measure vertical propagation times through the seabed and water column. More recently, there is experimental evidence that indicates noise correlations may also be used to determine horizontal propagation times in the seabed. Although replacing controlled projectors with noise sources is an attractive option there are limitations to how and when noise can be used this way. These limitations as well as experimental and modeling results from surface and ship noise will be described in this presentation.

4:00

1pAO9. Studying breaking waves with ambient noise oceanography. Grant B. Deane (Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St., La Jolla, CA 92093-0238, gdeane@ucsd.edu)

Ambient noise oceanography is a powerful tool in the study of breaking ocean waves, yielding insight into the nature of fluid turbulence and air entrainment within whitecaps. Whitecaps limit the height of waves on wind-driven seas, transfer momentum and gasses between the atmosphere and ocean, increase the ocean's albedo, are a source of marine aerosol and generate underwater ambient noise. For such reasons, the relationship between fluid turbulence and the size distribution of bubbles in whitecaps is of considerable interest, but the interior of whitecaps are quite difficult to study in the field. Large extinction rates associated with high air fractions in whitecaps limit the usefulness of optical and acoustical probes, and few measurements from in situ instrumentation have appeared. However, the sound radiated by newly formed bubbles within whitecaps provides a useful signal, one which can be interpreted in terms of bubble creation rates, provided the bubble emission signature and propagation through the breaking wave crest are understood. A physical model for the generation of wave noise and its use to provide insight into the link between fluid turbulence, its saturation in wave crests, and its relationship to bubble formation will be discussed. [Work supported by ONR Ocean Acoustics Division.]

4:20

1pAO10. The depth dependence of ambient noise in deep ocean trenches. David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca), Michael J. Buckingham, and Dieter Bevans (Marine Physical Lab, Scripps Inst. of Oceanogr., UCSD, La Jolla, CA)

A series of experiments aimed at measuring the power spectrum and vertical and horizontal noise coherence (directionality) in the deep ocean were carried out between 2009and 2015 using a family of autonomous instrument platforms named "Deep Sound." Deep Sound is a free-falling acoustic recorder designed to descend from the ocean's surface to a pre-assigned depth where it drops an iron weight and returns to the surface under its own buoyancy, while recording pressure time series on four hydrophones with vertical and horizontal spacing over the bandwidth 5 Hz—30 kHz. The complete vertical noise profile was recorded from the surface to 6 km in the Philippine Sea, 8.5 km in the Tonga Trench, and 9 km in the Serena and Challenger Deeps in the Mariana Trench. Generally, the vertical noise coherence is well described by the Cron & Sherman surface noise model with some depth-dependence explained by seawater attenuation and local sound speed variations. Below the reciprocal or critical depth (depth at which the sound speed is higher than any sound speed above it), the total noise field statistics depend heavily on the relative contributions of locally generated surface noise and distantly generated propagating noise. Thus, the depth dependence of the sound field varies with local surface conditions, with nearly depth-independent noise power and coherence found at moderate states, while a sharp drop in noise level is found below the critical depth during calm sea states.

Session 1pBAa

Biomedical Acoustics and Physical Acoustics: Photoacoustics: Light and Sound II

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

Kang Kim, Cochair
Medicine, University of Pittsburgh, 950 Scaife Hall, 3550 Terrace Street, Pittsburgh, PA 15261

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

Invited Papers

1:30

1pBAa1. Listening to light and seeing through: *In vivo* multiscale photoacoustic imaging. Chulhong Kim (Creative IT Eng., Pohang Univ. of Sci. and Technol., 328 Bonner Hall, Buffalo, New York 14260-2050, chulhong@buffalo.edu)

High-resolution volumetric optical imaging modalities, such as confocal microscopy, two-photon microscopy, and optical coherence tomography, have become increasing important in biomedical imaging fields. However, due to strong light scattering, the penetration depths of these imaging modalities are limited to the optical transport mean free path (~1 mm) in biological tissues. Photoacoustic imaging, an emerging hybrid modality that can provide strong endogenous and exogenous optical absorption contrasts with high ultrasonic spatial resolution, has overcome the fundamental depth limitation while keeping the spatial resolution. The image resolution, as well as the maximum imaging depth, is scalable with ultrasonic frequency within the reach of diffuse photons. In biological tissues the imaging depth can be up to a few centimeters deep. In this presentation, the following topics of photoacoustic imaging will be discussed; (1) multi-scale photoacoustic imaging systems (i.e., Photoacoustic Nanoscopy, Optical-Resolution Photoacoustic Microscopy, Fast 2-Axis MEMS based Optical-Resolution Photoacoustic Microscopy, Intravascular Photoacoustic/Ultrasound Catheter, Virtual Intraoperative Surgical Photoacoustic Microscopy, Acoustic-Resolution Photoacoustic Microscopy, and Clinical Photoacoustic/Ultrasound Scanner), (2) morphological, functional, and molecular photoacoustic imaging, (3) potential clinical applications, and (4) contrast agents for photoacoustic imaging.

1:50

1pBAa2. Development of three-dimensional photoacoustic imaging system using sparse array sensor. Tsuyoshi Shiina, Kengo Kondo (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), and Masakazu Toi (Dept. of Breast Surgery, Graduate School of Medicine, Kyoto Univ., Kyoto, Japan)

Photoacoustic (PA) imaging offers better spatial resolution of ultrasound for deep imaging with high optical contrast sensitive to physiological parameters such as the oxygen saturation of hemoglobin. We have conducted development of photoacoustic mammography system in Kyoto University/Canon joint research project supported by MEXT. In addition, we are promoting development of 3D-PA technologies to be applicable in wider clinical field and non-medical purposes in ImPACT program supported by cabinet office of Japan from 2014. Three-dimensional imaging is required to trace a vessel structure. However, the number of receiving channels is limited due to hardware restrictions. In such a sparse configuration, artifacts due to grating lobes become dominant. A scanning transducer suppresses grating lobes, which virtually increases the number of channels. However, it degrades temporal resolution and causes motion artifacts. We employ a compressed sensing (CS) technique to reconstruct a PA image from limited channels and bandwidth. Due to the computational cost for large 3D calculations, most CS approaches reconstruct PA images in a 2D slice. In order to reduce the cost while maintaining accuracy, we proposed 3D-PA reconstruction using CS based on k-space algorithm. The feasibility of proposed method was validated by simulation analysis and the experimental results.

2:10

1pBAa3. Volumetric multi-spectral optoacoustic tomography for high performance structural, functional, and molecular imaging. Daniel Razansky (Tech. University of Munich and Helmholtz Ctr. Munich, Inst. for Biological and Medical Imaging (IBMI), Ingolstaedter Landstrasse 1, Neuherberg 85764, Germany, dr@tum.de)

Optoacoustic (or photoacoustic) imaging is an emerging hybrid modality that can deliver excellent endogenous and exogenous optical absorption contrasts from living tissues with excellent spatio-temporal resolution. Our state-of-the-art implementations of multi-spectral optoacoustic tomography (MSOT) are based on multi-wavelength excitation of tissues to visualize specific molecules within opaque tissues. As a result, the MSOT technology can noninvasively deliver structural (i.e., vascular anatomy, solid tumors, and organs), functional (i.e., total hemoglobin concentration, hemoglobin oxygen saturation, blood flow, pH, and metabolic rate of oxygen consumption), and molecular information from living tissues. For highly sensitive molecular optoacoustic imaging, a valuable tool for personalized medicine, exogenous contrast agents (e.g., organic dyes, metallic and nonmetallic nanoparticles, reporter genes, or fluorescence proteins) with biomarkers are commonly utilized. The talk will further introduce the new realm of 5-dimensional (5D) optoacoustic imaging, which enables simultaneous acquisition of information across all the three spatial dimensions, the time and the spectral (optical wavelength) dimensions. Applications are explored in the areas of functional neuro-imaging, fast tracking of agent kinetics and biodistribution, cardiovascular research, monitoring of therapies, and drug efficacy, as well as targeted molecular imaging studies. Clinical translation roadmap will be finally discussed.

2:30

1pBAa4. Photoacoustic imaging with multiple speckle illumination. Emmanuel Bossy (LIPhy, Université Grenoble Alpes/ CNRS, LIPhy, 140 rue de la Physique, Saint-Martin d'Hères 38402, France, emmanuel.bossy@gmail.com)

In this presentation, I will illustrate how the use of multiple speckle illumination offers new degrees of freedom in photoacoustic imaging. In the acoustic-resolution regime, I will show that speckle illumination can help breaking the acoustic-diffraction limit [1]. I will show how multiple speckle illumination may also be used to perform optical-resolution photoacoustic microscopy, as an alternative to the conventional point scanning approach. [1]. Chaigne, T., Gateau, J., Allain, M., Katz, O., Gigan, S., Sentenac, A., & Bossy, E. 2016. Super-resolution photoacoustic fluctuation imaging with multiple speckle illumination. *Optica*, *3*(1), 54-57.

2:50

1pBAa5. Real-time, non-contact, non-invasive imaging of elasticity properties in soft tissues using the combination of light and air-coupled ultrasound. Ivan Pelivanov (BioEng., Univ. of Washington, 616 NE Northlake Pl, Seattle, WA 98105, ivan.pelivanov@gmail.com), Lukasz Ambrozinsky (AGH Univ. of Sci. and Technol., Krakow, Krakow, Poland), Shaozhen Song, Soon Joon Yoon (BioEng., Univ. of Washington, Seattle, WA), David Li (Chemical Eng., Univ. of Washington, Seattle, VA), Tueng Shen (Dept. of Ophthalmology, Univ. of Washington, Seattle, WA), Rikuang Wang, and Matthew O'Donnell (BioEng., Univ. of Washington, Seattle, WA)

A new non-contact method for efficient, non-invasive imaging of elasticity in soft media is presented. It uses transient mechanical waves excited in the object under study from air (i.e., non-contact) using either ultra-violet (UV) pulsed laser radiation (photoacoustic) or air-coupled focused ultrasound (US) and resultant displacements detected a high frame rate, phase-sensitive (PhS)-OCT system. UV laser radiation is absorbed in a thin subsurface region of the tissue creating transient mechanical waves due to thermal expansion, whereas the air-coupled focused US beam reflected from the air/medium interface provides acoustic radiation force (ARF) to the medium surface, launching a transient mechanical wave in the transverse (lateral) direction. Real time tracking/imaging of resultant mechanical wave propagation can be used to reconstruct tissue elastic properties. Tissue mimicking phantoms, porcine eyes, and skin were used in ex-vivoexperiments to prove the concept. Results of these experiments strongly suggest that simple, non-contact excitation holds great promise for non-invasive characterization of soft media, in general, and for elasticity measurements in delicate soft tissues and organs (e.g., in the eye), in particular.

3:10

1pBAa6. Skin vasculature imaging by acoustical resolution photoacoustic microscope with parabolic array transducer. Yoshifumi Saijo, Ryo Nagaoka, Israr Ul Haq, Syahril Siregar, Ryo Takagi, Shin Yoshizawa, and ShinIchiro Umemura (Tohoku Univ., 4-1 Seiryomachi, Aoba-ku, Sendai 980-8575, Japan, saijo@idac.tohoku.ac.jp)

Photoacoustic (PA) signal is highly dependent on the structure of the target. In the present study, PA imaging system with parabolic array transducer for multi-angle detection is developed and the skin vasculature is observed. The transducer was consisted of 256 ch 1-3-composite elements with the center frequency of 10 MHz. The diameter was 42.4 mm and the opening angle was 90 degree. A hole was made in the center of the transducer for transmission of the laser. PA signal was acquired by a programmable ultrasound data acquisition system with the sampling frequency of 60 MHz. A diode laser with the wavelength of 532 nm, pulse width of 8 ns, pulse energy of 5 mJ, and repetition frequency of 10 Hz was equipped for generation of PA signal. Three dimensional image was reconstructed by delay-and-sum beamforming of PA signals received by all channels. The frame rate for displaying x-y, y-z, and x-z images was 10 Hz, and the full width at half maximum (FWHM) of the x-z image was 70 μm. 3D structure of the skin capillary network of a human hand was successfully observed by the system. The system may contribute to the peripheral blood circulation.

3:30–3:45 Break

3:45

1pBAa7. Changes of masseter muscle in a rat unilateral occlusal model assessed by photoacoustic imaging system. Kouki Hatori (Dept. of Prosthodontics, Matsumoto Dental Univ. School of Dentistry, 1780, Gobara Hirooka, Shiojiri, Nagano 3990781, Japan, khat810@yahoo.co.jp), Yoshifumi Saijo (Dept. of Biomedical Imaging, Tohoku Univ. Graduate School of Biomedical Eng., Sendai, Japan), Masahiro Iikubo (Dept. of Oral Diagnosis, Tohoku Univ. Graduate School of Dentistry, Sendai, Miyagi, Japan), Yoshihiro Hagiwara (Dept. of Orthopaedic Surgery, Tohoku Univ. School of Medicine, Sendai, Miyagi, Japan), Kuniyuki Izumita, Yukihiro Naganuma, and Keiichi Sasaki (Div. of Adv. Prosthetic Dentistry, Tohoku Univ. Graduate School of Dentistry, Sendai, Miyagi, Japan)

Biomedical photoacoustic (PA) imaging system has the unique capability of combining high optical contrast with high ultrasound (US) resolution in a single modality. PA imaging system with 532 nm laser and 50 MHz US transducer has been developed. When the laser with the wavelength of 532 nm is used, PA signals are reflected from blood vessels. Patients with unilateral occlusion due to the extraction of unilateral molars are often seen in dental practice. Although the period of the unilateral occlusion induces the changes of masseter muscle (MM), it is not known that how blood vessels and tissue elasticity of MM are affected by the unilateral occlusion. In this study, we aimed to evaluate the sequential changes of MM due to the unilateral occlusion in rats with PA imaging system. PA signals from the MM of the non-occlusal side were significantly lower than that of the occlusal side, and US signals from the MM of the occlusal side were significantly higher than that of the non-occlusal side depending on the period of the unilateral occlusion. Unilateral occlusion affected both the blood vessels and the tissue elasticity. The PA imaging system could evaluate blood circulation and tissue elasticity simultaneously.

4:00

1pBAa8. All optical frequency domain photoacoustic microscope based on two-wave mixing in a photo-refractive crystal. Deepu George (Bio-Eng., George Mason Univ., 4400 University Dr., MSN 2A1, Krasnow Inst. of Adv. Study, Fairfax, VA 22030, dgeorg10@gmu.edu), Harriet Lloyd, Ronald H. Silverman (Edward S. Harkness Eye Inst., Columbia Univ., New York, NY), and Parag V. Chitnis (Bio-Eng., George Mason Univ., Fairfax, VA)

Optical resolution photoacoustic microscopy (OR-PAM) provides functional and molecular information at the cellular scale. OR-PAM typically employs ultrasonic transducers, which require coupling with tissue, to detect the thermo-elastic waves generated in tissues irradiated by nanosecond laser. Images are acquired by scanning the excitation beam, which is usually focused to a spot size of a few microns, i.e., lateral resolution. However, the axial resolution is constrained to an order of magnitude lower by the bandwidth of the transducer used. Additionally, imaging depth of OR-PAM is constrained to the depth that allows ballistic photons, which is usually less than 1 mm. We present an All Optical Frequency Domain Photoacoustic Microscope where the photoacoustic signal is generated by an amplitude modulated CW laser and detected using a two wave mixing (interferometer) in a photorefractive crystal. Sweeping the amplitude-modulation frequency and using an ultra-broadband optical detector can potentially provide detection bandwidths greater than conventional OR-PAM, e.g., 100-200 MHz. The loss in efficiency of photoacoustic excitation associated with violating the stress-confinement condition is overcome by using a lock-in amplifier. This method promises a high-fidelity microscopy system that does not require acoustic coupling while reducing associated costs and footprint. [Work supported by NIH-NEI 1R21EY023012.]

4:15

1pBAa9. Laser-driven resonance of light-absorbing ultrasound contrast microbubbles. Guillaume Lajoinie (Phys. of Fluids group, Univ. of Twente, Postbus 217, Enschede, Netherlands, g.P.R.Lajoinie@utwente.nl), Jeong Y. Lee (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Erik Linnartz (Phys. of Fluids group, Univ. of Twente, Enschede, Netherlands), Joshua Owen (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Pieter Kruizinga, Nico de Jong, Gijs van Soest (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Eleanor Stride (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and Michel Versluis (Phys. of Fluids group, Univ. of Twente, Enschede, Netherlands)

The sensitivity of ultrasound imaging is greatly enhanced by the use of microbubble contrast agents through resonant volumetric oscillations. While the increased acoustic contrast is of prime interest for perfusion imaging of organs, microbubbles until now have limited benefit in terms of specificity for ultrasound imaging. Original strategies are required to tackle this difficulty that rely on loading functional targeting ligands onto the microbubble encapsulation. In parallel, laser light offers great specificity in its interaction with tissue. This advantage is put to use in photoacoustic imaging where absorbed laser light is converted into a measurable acoustic signal. Here, we present a novel ultrasound contrast agent designed to make use of the specificity of laser light. The acoustic agent consists of a gas core encapsulated by an oil layer containing an absorbing dye. The resulting laser light absorption can then be used to heat up the gas and drive the system into resonance, thereby generating ultrasound. Combining finite difference simulations and ultra high-speed imaging led to a quantitative physical description of the optical and thermal interactions in the system resulting in the efficient generation of acoustic waves in the MHz range. A range of physical bubble parameters are investigated, i.e., thickness and composition of the light absorbing oil layer. This new generation of contrast agents will open up new applications in medical diagnostic and therapeutic imaging.

4:30

1pBAa10. Polypyrrole phase-change contrast agents for sono-photoacoustic imaging. David S. Li (Dept. of Chemical Eng., Univ. of Washington, 105 Benson Hall, Box 351750, Seattle, WA 98195, dsli@uw.edu), Soon Joon Yoon (BioEng., Univ. of Washington, Seattle, WA), Thomas J. Matula (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Matthew O'Donnell (BioEng., Univ. of Washington, Seattle, WA), and Lilo D. Pozzo (Chemical Eng., Univ. of Washington, Seattle, WA)

A new type of light and sound sensitive emulsion-based contrast agent is presented. It features a low boiling point liquid perfluorocarbon core and a polypyrrole (PPy) polymer shell. The PPy coated nanoemulsions can reversibly convert from liquid to gas phase by cavitating the perfluorocarbon core during the negative phase of an acoustic pulse. Alternatively, cavitation can be initiated through heat transfer from light absorption in the PPy shell from a laser pulse. By overlapping transmitted laser and acoustic pulses, activation energies required to cavitate the nanoemulsions are lower than for either source alone. The emulsions are typically produced between 150 and 350 nm in diameter and PPy has a broad optical absorption covering both the visible and near-infrared wavelengths (peak covers 700-1100 nm). The size, structure, and optical absorption properties of the contrast agents were characterized using dynamic light scattering, ultraviolet-visible spectrophotometry, transmission electron microscopy, and small angle X-ray scattering. The cavitation threshold and signal intensity was measured as a function of both acoustic pressure and laser fluence. We also demonstrate that these agents can be used as a sono-photoacoustic contrast agent, providing a non-linear photoacoustic response much more intense than particle-based contrast agents used in photoacoustic imaging.

4:45

1pBAa11. Reconstruction of vasculature in optical resolution photoacoustic microscopy using wavelet and Hessian based method. Israr Ul Haq, Ryo Nagaoka, Syahril Siregar, and Yoshifumi Saijo (Graduate School of Biomedical Engineeirng, Tohoku Univ., 6-6-05 Aramaki Aza Aoba, Aoba-ku, Sendai, Miyagi 980-8579, Japan, ryo@ecei.tohoku.ac.jp)

Photoacoustic microscopy is a high-resolution and high- contrast function modality where optical absorption is detected by ultrasonic transducer. The noise generated by the detector due to its size, sensitivity make the photoacoustic images difficult to visualize. Denoising and reconstruction of vascular structures is crucial for the early diagnosis and therapy in many medical applications. Therefore, the accurate detection of blood vessels has a great significance to assist the users including radiologists and clinicians by making them visualization of the vascular information in the images. In the proposed method, use of wavelet filtering to enhance the effect of smaller and bigger vessels in Optical Resolution Photoacoustic Microscopy (OR-PAM) is analyzed. The proposed method uses the wavelet to enhance the vasculature and then Hessian based method is applied to classify the vessel-like structures. For the evaluation, the algorithm is tested on photoacoustic images acquired non-invasively from living mouse brain, human finger cuticle, and blood filled tube, which shows appreciable results to enhance vasculature in photo-acoustic imaging.

5:00

1pBAa12. Deconvolution-based approach for super-resolution photoacoustic imaging. Jaesok Yu (BioEng., Univ. of Pittsburgh, 3550 Terrace St., Scaife Hall 958, Pittsburgh, PA 15261, JAY49@pitt.edu) and Kim Kang (Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

Recently, the microbubble-localization technique was able to increase spatial resolution by multifold in ultrasound (US) imaging, overcoming the acoustic diffraction limit. In this study, we applied the super-resolution approach to the photoacoustic (PA) imaging with a mid-frequency medical imaging array transducer. One of the biggest challenges in this application is relatively low PA response from individual contrast agent particles to the limited laser fluence under safety guidelines. The increased concentration of contrast agent enhances PA signal, but it results in convolved and blurred PA signals due to particle clumping. The conventional localization technique likely discards such blurred PA signals and thus requires a large number of imaging frames. Alternatively, a deconvolution method can be utilized to directly identify individual agent particles from the blurred PA signals. *In-vitro* experiments were performed using the PA sequence programmed US platform with a linear array transducer centered at 5 MHz. A pulsed laser of 5 ns long at 10 Hz tuned at 800 nm was synchronized to the US system. Totally 8,000 PA frames were acquired from micron-sized metal beads flowing in a polyethylene tube. Richardson-Lucy deconvolution method successfully identified individual particles from the blurred PA signals generated from clumped beads at high concentration.

5:15

1pBAa13. Visualization of ultrasound fields in a transparent medium with ultrasound attenuation using the focused shadowgraphy technique. Yukina Iijima and Nobuki Kudo (Graduate School of Information Sci. and Technol., Hokkaido Univ., N14W9 Kita-ku, Sapporo, Hokkaido 060-0814, Japan, kudo@bme.ist.hokudai.ac.jp)

Knowledge of ultrasound pressure fields of ultrasound inside the human body is essential to confirm the safety of medical ultrasound equipment. Typically, pressure waveforms in the body are estimated by correction of those measured in water considering typical attenuation in biological tissues. However, the correction of frequency-dependent attenuation of a wide band pulse is difficult because biological tissues have a non-linear transmission characteristic. In this study, acoustic fields of pulsed ultrasound propagating in a transparent medium with ultrasound attenuation were visualized using focused shadowgraphy (Kudo, Ultrasound Med. Biol. 41, 2071-2081, 2015). Pressure waveforms of the pulsed ultrasound were also measured using a membrane hydrophone. Brightness curves and their 1st-order integrations derived from the visualized field images were compared with the pressure waves. Experimental results obtained by using castor oil demonstrated that the 1st-order integrations of brightness curves are in better agreement with the pressure waveforms. Furthermore, frequency spectrum analysis showed good agreement of attenuation coefficients derived from the pressure waveforms and integrated brightness curves, indicating the usefulness of the technique not only for visualization of ultrasound fields but also for estimation of pressure waveforms and ultrasound attenuation.

5:30

1pBAa14. The acoustic noise pressure of sensors with very small apertures. Suzanne M. Leinders, Nico de Jong, and Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628 CJ, Netherlands, m.d.verweij@tudelft.nl)

For medical diagnostic modalities like intravascular ultrasound (IVUS) and intravascular photo acoustics (IVPA), it is paramount to have small, sensitive ultrasound elements for detecting the reflected pressure pulses. The development of one and two dimensional arrays for such applications will call for even smaller element sizes and advanced microfabrication techniques. In search for miniature receiving elements we developed an optical ultrasound sensor with an optical strain detector integrated on a thin acoustical membrane [Leinders *et al*., Sci. Rep. 5, 14328]. To predict the lowest detectable pressure, we wanted to determine the noise level of this sensor. Unlike a piezoelectric sensor, the noise in our sensor is not dominated by the electrical impedance and will only be caused by the thermo-acoustical noise of the sensor's internal mechanical impedance, and the noise caused by thermally agitated medium particles that hit the sensor surface. To expand the existing knowledge, we will analyze both noise mechanisms and show that in thermodynamic equilibrium these give rise to the same noise pressure at the sensor surface. Moreover,we will show that for sensors with vanishing aperture area, the noise pressure will reach a well-defined finite limit, and not go to infinity as predicted by some literature.

Session 1pBAb

Biomedical Acoustics and Physical Acoustics: Cavitation in Therapeutic Ultrasound I: General

Lawrence Crum, Cochair
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Shin-ichiro Umemura, Cochair
Graduate School of Biomedical Engineering, Tohoku University, Aoba 6-6-05, Aramaki, Aoba-ku, Sendai 980-8579, Japan

Tatiana D. Khokhlova, Cochair
University of Washington, 325 9th Ave., Harborview Medical Center, Box 359634, Seattle, WA 98104

Zhen Xu, Cochair
Biomedical Engineering, University of Michigan, 2200 Bonisteel Blvd., Rm. 1107 Gerstacker Bldg., Ann Arbor, MI 48109

Chair's Introduction—1:00

Invited Paper

1:05

1pBAb1. Cavitation-enhanced diagnostics and therapy in HIFU treatment. Yoichiro Matsumoto (RIKEN, 2-1 Hirosawa, Wako, Saitame 351-0198, Japan, yoichiro.matsumoto@riken.jp)

Medical applications of ultrasound such as High Intensity Focused Ultrasound, Extracorporeal Ultrasound Lithotripsy and Ultrasound-Mediated Gene Transfection have recently been the subject of much interest. In these applications, acoustic cavitation facilitates the medical treatment by improving the image quality and enhancing the therapeutic effects through acoustic emission, localized heating, and erosion. During the ultrasound therapy, it is important to control the focal region on the targeting area, which is moving by the respiratory motion, for the reliable treatment. Ultrasound guide is one of the effective way to track the targeting area compensating the respiratory motion. Cavitation-enhanced diagnostics is useful to obtain the clear image of targeting area utilizing the acoustic emission from the existing micro bubbles during the cavitation-enhanced HIFU where cavitation bubbles are generated at the targeting area to enhance tissue heating. In all these applications, it is essential to understand the cavitation phenomena, including bubble and bubble cloud dynamics. The bubble motion and bubble cloud behavior are strongly influenced by the thermal phenomena inside and outside bubbles and the applied ultrasound wave forms which accelerate therapeutic effects by the cavitation bubbles. In this paper, some fundamental aspects of medical application of microbubbles and acoustic cavitation are highlighted.

Contributed Papers

1:25

1pBAb2. On the formation of bubble clusters and tunnels in tissue-mimicking agarose phantoms by focused ultrasound bursts. Pooya Movahed (Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, 305 Coordinated Sci. Lab., 1308 W Main St., Urbana, IL 61801, pooyam@illinois.edu), Wayne Kreider, Adam D. Maxwell, Michael R. Bailey (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Jonathan B. Freund (Mech. Sci. and Eng. and Aerosp. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Formation of bubble clusters in soft tissues is a potential injury mechanism in therapeutic ultrasound treatments. To study this phenomenon, transparent tissue-mimicking agarose phantoms were subjected to a series of multiple-cycle ultrasound bursts, using a burst wave lithotripsy (BWL) protocol, and simultaneously imaged with a high-speed camera. The negative pressure in the initial bursts causes preexisting sub-micron bubbles to expand sufficiently to become visible in images (~200 microns). Additional bubbles appear continuously during the subsequent bursts. A Rayleigh-Plesset-type bubble dynamics model, which is generalized to include elastic resistance and damage mechanisms, is developed and used to explain key observations. It is proposed that material fatigue leads eventually to irreversible fracture-like failure. In addition to isolated, approximately spherical bubbles, long tunnel-like features are observed, which seemingly comprise lines of joined bubbles along a possible fracture or defect. Statistics regarding the geometry of these features are reported for agarose phantoms of different stiffness. A mechanism for the formation and observed growth of these long features is proposed, wherein defect/crack growth allows the movement of initially trapped bubbles within the phantom. [Work supported by NIH NIDDK grant DK043881.]

1:40

1pBAb3. Dependence of inertial cavitation activity on the high intensity focused ultrasound transducer F-number. Tatiana Khokhlova, Christopher Hunter, Wayne Kreider, Adam Maxwell, Vera Khokhlova, and Oleg Sapozhnikov (Univ. of Washington, 325 9th Ave., Harborview Medical Ctr., box 359634, Seattle, WA 98104, tdk7@uw.edu)

Cavitation induced in tumors by pulsed high intensity focused ultrasound (pHIFU) without ultrasound contrast agents was shown to significantly enhance chemotherapeutic drug uptake. Such *de novo* cavitation is commonly assumed to require very high rarefactional pressures. However, recent studies have shown that inertial cavitation threshold also correlates with formation of shocks at the focus. The shock amplitude and corresponding peak negative pressure (p-) are primarily determined by the transducer F-number with less focused transducers producing shocks at lower p-. Here, the dependence of inertial cavitation activity on F-number was investigated in gel phantoms and *ex vivo* tissue samples using passive cavitation detection (PCD). Exposures at 1.5 MHz consisted of 60 pulses delivered at 1 Hz PRF, with each pulse lasting 1 ms and p- from 1 to 15 MPa. Broadband noise emissions recorded by PCD were batch-processed to extract cavitation probability and persistence. At the same p-, both metrics indicate enhanced cavitation activity at higher F-numbers; in agarose phantoms, cavitation probability reached 100% when shocks formed at the focus, with p- values 5, 9, and 14.5 MPa for respective F-numbers 1.5, 1, and 0.75. These results confirm the impact of nonlinear waveform distortion on inertial cavitation. [Work supported by NIH K01EB015745, K01DK104854, and R01EB7643.]

Invited Papers

1:55

1pBAb4. Cavitation-enhanced ultrasonic heating and its monitoring by ultrasound plane wave imaging. Shin Yoshizawa, Ryo Takagi (Communications Eng., Tohoku Univ., 6-6-05 Aoba, Aramaki, Aoba-ku, Sendai 980-8579, Japan, syoshi@ecei.tohoku.ac.jp), Ryosuke Iwasaki, Kentaro Tomiyasu, Kai Suzuki, and Shin-ichiro Umemura (Biomedical Eng., Tohoku Univ., Sendai, Japan)

Cavitation bubbles can accelerate high-intensity focused ultrasound (HIFU) treatments thermally, chemically, and mechanically. Acoustic generation of cavitation bubbles requires extremely high-intensity or long duration time enough to form a standing-wave field. We have proposed and investigated the ultrasound sequence of a high-intensity short pulse, named as trigger pulse, followed by a relatively lower intensity and longer duration burst, named as heating burst, mainly for the cavitation-enhanced ultrasonic heating. Trigger pulses are used to generate cavitation bubbles then form cavitation clouds. The acoustic power of trigger pulses is more than hundreds of W and the duration is less than 100 μs to avoid effects of standing-wave components. Cavitation bubbles induced by trigger pulses continuously oscillate with heating bursts. The sequence was repeated 3—15 s. Cavitation clouds formation by trigger pulses was observed with a high-speed camera. The temperature measured by thermocouple and tissue coagulation region showed the effect of heating enhancement. As the cavitation region localization is an essential factor for the safety and efficacy in this method, a real-time monitoring method of bubbles is necessary to be combined. The high-speed pulse inversion images with plane wave transmissions showed its capability for the objective.

2:15

1pBAb5. Numerical study on cavitation-enhanced thermal and mechanical effects in high-intensity focused ultrasound therapy. Kohei Okita (Mech. Eng., Nihon Univ., 1-2-1 Izumi-cho, Narashino, Chiba 275-8575, Japan, okita.kohei@nihon-u.ac.jp) and Yoichiro Matsumoto (RIKEN, Wako, Saitama, Japan)

Cavitation is becoming increasingly important in therapeutic ultrasound applications such as diagnostic, tumor ablation, and lithotripsy. The behavior of cavitation bubbles is resulted by the complex interactions between bubble oscillation and ultrasound. That depends on not only the exposure parameters of ultrasound but also the physical properties of surrounding tissues and injected microbubbles which are often employed as cavitation nuclei for decreasing cavitation threshold pressure and for controlling cavitation region. An ultrasound simulator treating cavitation/microbubble has been developed for investigating the influences of various parameters in the cavitation-enhanced HIFU therapy. The simulator takes into account the heat generation due to oscillating cavitation bubbles as well as the rectified diffusion which is important role in the initial stage of cavitation bubble growth. Bubble dynamics equation considers the elasticity of the surrounding tissue and the shell of microbubbles, and is strongly coupled with mixture phase to reproduce the interaction between ultrasound and bubble oscillation. Thus, the simulator can reproduce cavitation bubble growth from nuclei around focus. In the present paper, the numerical results for the cavitation/microbubble-enhanced HIFU therapy are shown.

2:35

1pBAb6. Color-Doppler ultrasound imaging of *in situ* human kidney stones in a hyperbaric chamber. Julianna C. Simon (Appl. Phys. Lab., Dept. of Mech. Eng. and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jcsimon@uw.edu), Barbrina Dunmire, Bryan W. Cunitz (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA and Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Jeffrey Thiel, Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and James R. Holm (Div. of Hyperbaric Medicine, Virginia Mason Medical Ctr., Seattle, WA)

Overpressure has been shown to reduce the color-Doppler ultrasound twinkling artifact on *ex vivo* human kidney stones, leading to the hypothesis that surface crevice microbubbles cause twinkling. For the first time, we investigate the effect of overpressure on *in situ* human kidney stones. Thus far, five human subjects with kidney stones known to twinkle have been imaged with a Philips/ATL P4-2 transducer and Verasonics ultrasound system for 45 minutes in a hyperbaric chamber. Subjects breathed ambient air while exposed to a maximum pressure of 4 atmospheres absolute (ATA), except for a scheduled decompression stop at 1.6 ATA where subjects breathed pure oxygen. Twinkling was quantified in terms of Doppler power over 2 min intervals before pressurization (baseline), at 4 ATA, and at 1.6 ATA. Preliminary results (averaged over 3 of the 5 subjects) indicate no change in twinkling at 4 ATA compared to baseline levels (ratio of Doppler powers = 1.09 ± 0.27). Twinkling almost doubled, though, during the pure oxygen stage at 1.6 ATA compared to

baseline levels (ratio of Doppler powers = 1.85 ± 0.58). The increase in twinkling associated with breathing pure oxygen continues to support the crevice microbubble hypothesis. Higher pressures than explored in this study may be needed to reduce twinkling on *in situ* stones, as was often the case in studies using *ex vivo* human kidney stones. [Work supported by the National Space Biomedical Research Institute through NASA NCC 9-58 and NIH grants DK043881 and DK092197.]

Contributed Papers

2:55

1pBAb7. Acoustic cavitation sub-millisecond signal processing for ultrasound surgery. Bjoern Gerold, Jeremie Anquez, and Sylvain Yon (Theraclion, 102 rue Etienne Dolet, Malakoff 92240, France, bjoern.gerold@theraclion.com)

Cavitation in Focused Ultrasound Surgery (FUS) is a stochastic phenomenon that can evolve very rapidly at the high intensities typically employed. Cavitation activity can be monitored analyzing its acoustic signals in the frequency domain. Cavities act as strong scatterers of incident acoustic energy, potentially enhancing different therapeutic effects, e.g., thermal or mechanical, if properly controlled. We report on the development on an instrument capable of controlling acoustic cavitation at sub-millisecond speed. The cavitation noise signal has been recorded and processed by an FPGA coupled with a feedback loop, capable of modulating the driving field every few hundreds of microseconds to maintain an ideal level of cavitation activity and differentiate between several types of cavitation and the onset of boiling. The system has been tested in tissue phantom, ex-vivo tissue and *in-vivo* and the impact on lesion sizes has been analysed. We also used the experimental data as input for a non-linear simulation of the acoustic field specifically regarding the absorption/scatter ratio in the acoustic focus. [The research leading to these results has received funding from the People Programme (Marie Curie Actions) of the European Union's Seventh Framework Programme FP7/2007-2013/under REA grant agreement n° (623608).]

3:10–3:25 Break

3:25

1pBAb8. Elimination of therapeutic ultrasound noise in phase modulated high intensity focused ultrasound treatment. Ryo Takagi (Communications Eng., Tohoku Univ., 6-6-05 Aoba,Aramaki, Aoba-ku, Sendai 9810933, Japan, takagi@ecei.tohoku.ac.jp), Ryosuke Iwasaki (Biomedical Eng., Tohoku Univ., Sendai, Japan), Shin Yoshizawa (Communications Eng., Tohoku Univ., Sendai, Japan), and Shin-ichiro Umemura (Biomedical Eng., Tohoku Univ., Sendai, Japan)

Ultrasound image-guided High Intensity Focused Ultrasound (HIFU) has been become one of the potential surgical techniques for the treatment of the malignant tumor. The ultrasound image during the treatment is severely corrupted by the reflected HIFU wave because HIFU and imaging transducers are simultaneously activated. In our previous study, a new method to selectively eliminate only HIFU noise during the treatment was suggested, but it cannot sufficiently reduce the HIFU noise when the HIFU exposure is phase-modulated with a period less than a few milliseconds. In this study, a new method to selectively eliminate the HIFU noise even when the HIFU exposure is modulated with a certain modulation period. In this study, "multiple HIFU exposure" with a certain modulation period was employed. The focus of HIFU was moved every 25μs sequentially for six positions with spacing proper for thermal conduction. the RF signals of the same modulation period were received with (RF2) and without (RF1) transmitting an imaging pulse, respectively. The HIFU noise was eliminated by subtracting RF1 from RF2. As the result of *in vitro* study, the fundamental to higher harmonic components of the HIFU noise were reduced by 20 to 30 dB during the HIFU exposure.

3:40

1pBAb9. A portable transcranial focused ultrasound system for non-invasive applications in small animals. Pavlos Anastasiadis, Ben Nguyen (Diagnostic Radiology and Nuclear Medicine, Univ. of Maryland School of Medicine, 110 S. Paca St., Rm. 104, Baltimore, MD 21201, panast@som.umaryland.edu), David S. Hersh (Neurosurgery, Univ. of Maryland School of Medicine, Baltimore, MD), Sijia Guo, Rao Gullapalli, and Victor Frenkel (Diagnostic Radiology and Nuclear Medicine, Univ. of Maryland School of Medicine, Baltimore, MD)

Recent studies using focused ultrasound (FUS) have shown significant progress in the realm of brain applications. These transcranial exposures include localized ablation for the treatment of movement disorders and opening of the blood brain barrier to enhance the delivery of therapeutic agents. Whereas these procedures require expensive, MRI-guided systems, one of the newest and most promising applications of FUS being developed involves more portable, hand-held devices for the purpose of neuromodulation. Here, we describe the development of table-top FUS system comprised of a single element FUS transducer (400—600 kHz) specifically for investigations on the stimulation and suppression of neuronal activity in rodents. The transducer assembly includes a cone and extending flexible bolus filled with degassed water for direct coupling to the head of the animals. A custom 3D stage and two diametrically opposed lasers allow for accurate positioning and targeting. Experiments included simulations of the acoustic field using a k-space time-domain method. Localization of the focal zone and its radial and axial dimensions were obtained via thermal exposures in polyacrylamide gel phantoms containing heat sensitive proteins. Last, hydrophone measurements with various sized rodent skulls enabled determination of the expected attenuation to be encountered for future *in vivo* animal treatments.

Invited Paper

3:55

1pBAb10. High-speed observation for a better understanding of bubble dynamics in an *in vivo*-like situation of sonoporation. Nobuki Kudo (Graduate School of Information Sci. and Technol., Hokkaido Univ., N14W9 Kita-ku, Sapporo, Hokkaido 060-0814, Japan, kudo@bme.ist.hokudai.ac.jp)

Sonoporation is a technique to temporally increase the permeability of a biological cell membrane by ultrasound exposure and to transduce a foreign gene or drug that normally has no permeability. The mechanisms of sonoporation have not been fully elucidated yet; however, it is widely accepted that microbubbles oscillating adjacent to cells play an important role to induce membrane damage. Dynamics of microbubbles has strong dependence not only on exposure parameters of ultrasound but also on surrounding conditions of the bubbles. To obtain a better understanding of cell-bubble interactions affected by the surrounding conditions, high-speed observation was carried out. A newly designed chamber was used for observation from a lateral direction, enabling visualization of translational

movements of bubbles beside a scaffold and of cell-bubble interaction without hiding the interacting point by the oscillating bubble itself. It was observed that a bubble on a soft scaffold surface moves translationally in a direction separating from the surface, whereas a bubble on a hard scaffold surface remains on the surface. Adhesion of a bubble to the top of a cell caused a considerable decrease in bubble oscillation amplitude, but membrane damage was confirmed in all of the observed cells with attached bubbles, indicating the importance of taking into account the conditions surrounding a bubble in order to understand the mechanisms of sonoporation.

Contributed Papers

4:15

1pBAb11. Toward treatment of abscesses using non-thermal HIFU. Andrew A. Brayman, Anna McClenny, Yak-Nam Wang, Brian MacConaghy, Keith Chan, Wayne Monsky, and Tom Matula (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@uw.edu)

Abscesses are infected walled-off collections of pus and bacteria. They can affect any part of the body. Current treatment is typically limited to antibiotics, catheter drainage and hospitalization, or surgical wash-out when inaccessible, loculated or unresponsive to initial care efforts. Although bacteria can develop drug resistance, they remain susceptible to thermal and mechanical damage. High Intensity Focused Ultrasound (HIFU) generates localized heating and cavitation, and represents a potential new noninvasive treatment modality. This talk describes initial experiments in which non-thermal HIFU treatment was used to inactivate small volumes (100 μL—10 mL) of *Escherichia coli*suspensions (~1 × 10^9 cells/mL) with 1 or 2-MHz transducers. Free-field focal acoustic pressures were as high as 16 (9.9) MPa peak positive (negative). Survival was assessed by coliform counting, and by alamarBlue® vital staining. At duty factors of 0.01 or 0.2, and the highest acoustic pressures, there was no biologically significant heating of the exposed samples. Inactivation was treatment time-dependent, and was well described by a half-life model. There was a well-defined (free field-equivalent) acoustic pressure threshold, with significant cell inactivation above ~7 MPa peak negative pressure.

4:30

1pBAb12. Comparative studies on enhancement of cell killing induced by ultrasound in the presence of platinum and gold nanoparticles. Takashi Kondo (Dept. of Radiological Sci., Univ. of Toyama, Sugitani 2630, Toyama 9300194, Japan, kondot@med.u-toyama.ac.jp)

In this study, we report on the potential use of platinum nanoparticles (Pt-NPs), a superoxide dismutase (SOD)/catalase mimetic antioxidant, in combination with 1 MHz ultrasound (US) at an intensity of 0.4 W/cm(2), 10% duty factor, 100 Hz PRF, for 2 min. Apoptosis induction was assessed by DNA fragmentation assay, cell cycle analysis and Annexin V-FITC/PI staining. Cell killing was confirmed by cell counting and microscopic examination. The mitochondrial and Ca(2+)-dependent pathways were investigated. Caspase-8 expression and autophagy-related proteins were detected by spectrophotometry and western blot analysis, respectively. Intracellular reactive oxygen species (ROS) elevation was detected by flow cytometry. The results showed that Pt-NPs exerted differential effects depending on their internalization. Pt-NPs functioned as potent free radical scavengers when added immediately before sonication while pre-treatment with Pt-NPs suppressed the induction of apoptosis as well as autophagy AP, and resulted in enhanced cell killing. In addition, we used gold nanoparticles without anti-oxidant activities and will discuss the difference between the two.

Invited Paper

4:45

1pBAb13. Cavitation-mediated extravasation and transtumoral drug delivery by microstreaming: What role do the gas nuclei and the physical properties of the therapeutic play? Constantin Coussios, Rachel Myers, James Kwan, Harriet Lea-Banks, Steven Mo, Megan Grundy, Christophoros Mannaris, Margaret Duffy, Catherine Paverd, Dario Carugo, Christian Coviello, Eleanor Stride, and Robert Carlisle (Dept. of Eng. Sci., Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, constantin.coussios@eng.ox.ac.uk)

For effective cancer therapy, therapeutic agents must extravasate from the blood stream into the tumour mass, then overcome the elevated intratumoural pressure and dense extracellular matrix to reach each and every cancer cell. Several recent studies have suggested that inertial cavitation in the absence or presence of artificial cavitation nuclei can significantly enhance the delivery, penetration, and distribution of small-molecule or nanoparticulate therapeutic agents into tumors. We first present a comparison of the reported enhancements in delivery achieved for a range of frequencies and therapeutic sizes without or with pre-seeding of cavitation, with particular emphasis on the potential role of cavitation persistence and spatial distribution. With microstreaming hypothesized to be the dominant transport mechanism for drug delivery, the likely benefit of using sub-micron cavitation nucleation agents capable of extravasating alongside the therapeutic, rather than microbubbles confined to the blood pool, is then investigated. Lastly, potential physical and biological modification strategies capable of enhancing the transport of therapeutics by cavitation-microstreaming are discussed, and compared in terms of their relative delivery efficacies *in vivo*.

Contributed Paper

5:05

1pBAb14. 1-MHz ultrasound stimulates protective mechanisms in cardiac endothelial cells during oxygen and glucose deprivation.
Azzdine Y. Ammi (Knight Cardiovascular Inst., Oregon Health & Sci. Univ., 3181 SW Sam Jackson Park Rd., UHN-62, Portland, OR 97239, ammia@ohsu.edu), Catherine M. Davis (Anesthesiology & Perioperative Medicine, Oregon Health & Sci. Univ., Portland, OR), Igor V. Dykan (Knight Cardiovascular Inst., Oregon Health & Sci. Univ., Portland, OR), Mohanika Gowda, Nabil J. Alkayed (Anesthesiology & Perioperative Medicine, Oregon Health & Sci. Univ., Portland, OR), and Sanjiv Kaul (Knight Cardiovascular Inst., Oregon Health & Sci. Univ., Portland, OR)

Ultrasound improves myocardial perfusion during coronary occlusion. Our aim was to study the production of vasodilatory compounds by primary mouse cardiac endothelial cells (ECs) exposed to ultrasound after a 2-hour oxygen and glucose deprivation (OGD). A 1.05-MHz transducer was used to insonify ECs with a 50-cycle tone burst at a peak rarefactional pressure of 0.5 MPa and a PRF of 50 Hz. US exposure of ECs after OGD increased the adenosine release to 168±16% of control (n = 11, $p<0.05$). It also resulted in an increase in 8,9-, 11,12- and 14,15-EETs reaching 125±19%, 123±17%, and 118±15% of control (n = 7, $p<0.05$), respectively. US caused an increase of 5,6-, 8,9-, 11,12-, and 14,15-DHETs to 135±16%, 138±17%, 133±15%, and 133±14% of control, respectively. It also caused levels of 18-, 19-, and 20-HETEs to be statistically different when compared to control and OGD alone (n = 7, $p<0.05$). eNOS phosphorylation level was not statistically significant to either control or OGD alone. Higher level of cell viability compared to the ECs not exposed to US was observed (n = 7, $p<0.05$). Pulsed ultrasound at 1.05 MHz has the ability to increase adenosine and eicosanoids production by cardiac ECs after OGD and increase their viability.

MONDAY AFTERNOON, 28 NOVEMBER 2016 — CORAL 3, 1:00 P.M. TO 2:45 P.M.

Session 1pED

Education in Acoustics: Attractive Educational Methods and Tools in Acoustics II

Kazuhiko Kawahara, Cochair
Department of Design, Kyushu University, 4-9-1 shiobaru, Minami-ku, Fukuoka 815-8540, Japan

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

Invited Papers

1:00

1pED1. Advanced technical listening training program at Kyushu University. Kazuhiko Kawahara, Masayuki Takada, and Shin-ichiro Iwamiya (Faculty of Design, Kyushu Univ., 4-9-1 shiobaru, Minami-ku, Fukuoka 815-8540, Japan, kawahara@design.kyushu-u.ac.jp)

This presentation introduces the Advanced Technical Listening Training (TLT) program at Kyushu University. TLT is a systematic training program designed to improve auditory sensitivity; it consists of discrimination and identification tasks for a wide variety of acoustic features (e.g., frequency, sound pressure level, and spectrum pattern). The TLT II class (i.e., the advanced TLT program) provides training in identification of reverberation time, quantization bit depth, mixing level balance, and spectral slope of harmonic tones. TLT facilitates improved understanding of acoustic theory and phenomena, and psychoacoustics.

1:20

1pED2. The use of all-in-one sensor devices in the general education acoustics laboratory. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

For a general education acoustics course meeting the physical science laboratory requirement, we have implemented the use of IOLab devices. The IOLab is an all-in-one collection of sensors in a box approximately the size of a graphing calculator. The devices include a microphone, accelerometer, force sensor, light sensor, magnetic field sensor, analog inputs and outputs among other built-in features. Each of the devices communicates wirelessly with laboratory computers. Our acoustics course serves not only students seeking general education credit but also students pursuing careers in music technology and as sonography technicians. The common bond between students in these groups is their shared avoidance of mathematical analysis. In our course, we attempt to build student skills in analytical thinking by scaffolding laboratory activities in a series of mini-labs which make use of the IOLab devices. We discuss the successes and challenges faced using this approach.

1:40

1pED3. Use of smartphones for introductory acoustics education. Kimihiro Sakagami (Environmental Acoust. Lab., Dept. of Architecture, Grad. School of Eng., Kobe Univ., Rokkodai 1-1, Nada, Kobe, Hyogo 657-8501, Japan, saka@kobe-u.ac.jp), Fumiaki Satoh (Dept. of Architecture, Chiba Inst. of Technol., Narashino, Japan), and Akira Omoto (Faculty of Design, Kyushu Univ., Fukuoka, Japan)

In order to propose an effective method using smartphones (including tablets) of acoustics education for introductory courses of architectural and environmental acoustics in architectural studies, the authors have examined some applications which work on smartphones. As the first step, applications measuring sound level and spectrum at reasonable prices are chosen and their precision have been verified. Results showed that the most iOS devices has somewhat reasonable precision, e.g., for SPL measurement, it is similar to Class 2 sound level meter, though some Android devices give lower precision. Then, the authors introduced these tools to student and encouraged them to use these tools to produce a noise map (with dBA only and with both dBA and sound spectra). Even though these simple tools allow students to understand the relationship between their sensation and physical values. As further studies, the authors also tried to use some more applications which enables students to measure more advanced physical values, such as Band Levels, Leq, Impulse Responses, Reverberation Times, etc. Some examples of the measurement results of them are also presented.

Contributed Papers

2:00

1pED4. Visualizing sound directivity via smartphone sensors. Scott H. Hawley (Chemistry & Physics, Belmont Univ., 1900 Belmont Blvd., Nashville, TN 37212, scott.hawley@belmont.edu) and Robert E. McClain (OmegaLab Studio, Nashville, TN)

We present a fast, simple method for automated data acquisition and visualization of sound directivity suitable for use with introductory acoustics education, made convenient and accessible via a new free smartphone app, "Polar Pattern Plotter." The app synchronizes measurements of sound volume with the phone's angular orientation, obtained from either compass, gyroscope, or accelerometer sensors, and produces a graph and exportable data file. It is generalizable to various sound sources and receivers via the use of an input-jack-adaptor to supplant the smartphone's (omnidirectional) microphone. We show results obtained in a free-field environment for loudspeaker directivity, microphone polar patterns, and two-speaker interference effects. Results provide both a visual and quantitative representation of sound fields and device responses, useful for introductory acoustics experiments and demonstrations.

2:15

1pED5. Efficacy of a new spatial ear training program for "Ensemble width" and "Individual source width". Hidetaka Imamura (Res. Area of Creativity of Music and Sound, Tokyo Univ. of the Arts, 1-25-1, Senju, Adachi-Ku, Tokyo, Adachi 120-0034, Japan, xiulong1990@gmail.com) and Sungyoung Kim (Elec., TeleCommun., and Comput. Eng. Technol., Rochester Inst. of Technol., Rochester, NY)

The authors have devised a matching-based training paradigm that assists listeners in evaluating spatial width of a reproduced sound field with increased sensitivity and memory. The program uses a learning paradigm of repeated comparison and match to references. All sound sources were horizontally spread using a pairwise constant power panning law over front five loudspeakers located at +-60, +-30, and 0 degree. Trainees were asked to adjust a parameter controlling the panning intervals of five-channel sound sources until it matched to the perceived width of a given reference. The program provided visual feedback for isomorphic mapping. To make the training equipment more accessible, the headphone version of the program was also developed using a simulation of listener and sound sources positions in a room. Thirty participants were divided into three groups: a group with loudspeaker-based training, a group with headphone-based training, and a group without training. Participants in the trained groups took 6 days training program with 3 to 5 sessions in a day. The difference between reported and presented intervals before and after the training were measured. The analyzed results showed that regardless of training method (loudspeaker- or headphone-based), trained groups showed improvement in their matching precision.

2:30

1pED6. Underwater acoustics education at University of Washington Bothell. Shima Abadi (Univ. of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011, abadi@uw.edu)

University of Washington Bothell was founded in 1990 to bring the University of Washington traditions of academic excellence to residents of the Puget Sound region. Mechanical engineering launched as a new undergraduate program in engineering & mathematics division in fall 2014. Underwater acoustics is one of the main focus areas in ME program which is integrated in several elective courses to provide students with formal training prior to the graduation and prepare them for research, internships, and ocean acoustics job positions. University of Washington Bothell in collaboration with local research institutes and industries supports underwater acoustics capstone projects. In addition to the capstone projects, many students are involved in theoretical and experimental research projects such as marine mammal bioacoustics, acoustical array signal processing, and underwater acoustic sensor design. In this talk, the underwater acoustics emphasis at the University of Washington Bothell will be presented along with the courses offered, research activities, capstone projects, and collaborations with local communities.

Session 1pMU

Musical Acoustics: Music and Sound in Multimedia

Shin-ichiro Iwamiya, Cochair
Kyushu University, Shiobaru 4-9-1, Fukuoka 814-8540, Japan

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

Invited Papers

1:30

1pMU1. The advanced technique of the counterpoint between music and video in Evangerion: 2.0 You can (not) advance. Masashi Yamada (Dept. of Media Informatics, Kanazawa Inst. of Technol., 3-1 Yatsukaho, Hakusan, Ishikawa 924-0838, Japan, m-yamada@neptune.kanazawa-it.ac.jp) and Riu Yanagida (Graduate School of Eng., Kanazawa Inst. of Technol., Nonoichi, Ishikawa, Japan)

Film director, Akira Kurosawa, used a technique called the counterpoint between music and video, where a part of a film consisted of incongruent features between the music and motion picture, in his films. Hideaki Anno, a director of animated movies, invented an advanced technique of the counterpoint in his film EVENGERION:2.0 YOU CAN (NOT) ADVANCE. In the present study, the emotions of the music and video, and the congruency between them were measured continuously, to reveal the effects of this advanced technique. The results showed that, in the long last scene, the emotional features of the music were incongruent with the video at first. Then, the emotions of the music gradually coincided with the video. Finally, they became congruent. In the present study, various types of music were presented with the original video, and these versions were also used for the continuous measurements. Moreover, overall impressions of these versions, as well as the original version, were measured. The results showed that the original version was perceived as significantly more preferable and enjoyable than the versions where music was consistently congruent with the video.

1:50

1pMU2. Effects of the tempo of background music on prospective and retrospective duration of a video program. Akira Nishimura (Dept. of Informatics, Tokyo Univ. of Information Sci., 4-1, Onaridai, Wakaba-ku, Chiba 2658501, Japan, akira@rsch.tuis.ac.jp)

To improve understanding of the effects of music on perception of video productions, subjective duration of a video program was measured in the paired-comparison paradigm using a 10-s video program selected from a 70-s daily news program mixed with a 90 beats-per-minute (bpm) music as a reference. Seven video programs were used for comparison, with actual durations ranging from 8.5 s to 11.5 s in steps of 0.5 s. These programs were selected randomly from the same video program and mixed with music played at 50, 90, and 140 bpm. Seven participants listened to a pair of video programs and then answered which program of the two was subjectively longer. The psychometric functions of retrospective perception of duration showed that video programs mixed with the fast-tempo music were perceived as short. In addition, five other participants were asked to prospectively produce a 10-s interval on a stopwatch while watching the same video programs. The actual times recorded generally exceeded 10 s, and were significantly long for the program mixed with slow-tempo music. The conflict between the results obtained from the two experiments on subjective duration and its relation to the fixed physical duration is discussed in terms of internal clocks.

2:10

1pMU3. Influence on story prediction of movie by difference in music. Yuki Mito, Ayame Suzuki (Music, Nihon Univ. College of Art, 2-42-1 Asahigaoka Nerima-ku, Tokyo 176-8525, Japan, mito.yuuki@nihon-u.ac.jp), and Hiroshi Kawakami (Music, Nihon Univ. College of Art, Tokyp, Japan)

The music often may assist to predict of movie work that has story line. The interpretation of the movie may sometimes change by music. In fact, there are many movies changing a story by music intentionally. In this way, interpretation of the story changes by using multi-modal. In the preceding research, we have known that the story of the movie can change by music but there are kind of movies used in the research, there is no assumption research of actual television or commercial. In this study, we studied how the difference of the music will influence. We have created four different types of music and silence for the movie, only played by piano as to lose prejudice by using types of other instruments. The short movie which created lasts 30 seconds will ends unfinished in order to match all kinds of stories. As a result, we have found that when the story is vague, prediction will change by the difference of the music after the movie. We hope this study helps producers of television and movies to convey their intentions more appropriately to ordinance.

2:30

1pMU4. Optimum insertion timing of symbolic music to induce laughter in video contents. Ki-Hong Kim (Media & Information Resources, Surugadai Univ., 698 Azu, Hanno, Saitama 357-8555, Japan, kim.kihong@surugadai.ac.jp), Mikiko Kubo (Hitachi Solutions, Ltd., Tokyo, Tokyo, Japan), and Shin-ichiro Iwamiya (Faculty of Design, Kyushu Univ., Fukuoka, Fukuoka, Japan)

In television variety shows or comedy programs, in addition to real sounds, various sound effects and music are combined with humorous scenes to induce more pronounced laughter from viewer-listeners. Symbolic music whose imagery is associated with special meaning based on past usage is dubbed just after humorous scenes as a sort of "punch line" to emphasize their humorous nature. Rating experiments using a method of paired comparisons were conducted to clarify the best timing for insertion of such music into humorous video contents. In the case of purely comical scene, the optimal insertion time was 0.0 to 0.5 seconds after the target scene. In the case of tragicomic scene (humorous accident), the optimal insertion time was 0.5 to 1.0 seconds after the scene, i.e., a short pause before the music was effective in this case. In both cases, the effects of symbolic music decreased 1.5 to 2.0 seconds after the scenes. The rating experiments showed that optimal timing was associated with highest impressiveness of the videos, highest evaluations, highest congruence between moving pictures and sounds, and inducement of maximum laughter.

2:50

1pMU5. Synchronization and periodicity of auditory and visual accents affect perceived congruence between sound and moving pictures. Saki Liu, Fumiyuki Takishita, Jingjing Gao, Kae Ishida, and Shin-ichiro Iwamiya (Kyushu Univ., 4-9-1, Shiobaru, Minami-ku, Fukuoka, Japan, sf0530@gmail.com)

Continuous rating experiments focused on the congruence between moving pictures and sounds have shown that synchronization between auditory and visual accents is fundamental to audiovisual congruence. We conducted experiments with snare sounds as auditory accents and changes in the color of a full screen as visual accents. We found that the periodicity of the auditory and visual accents contributed to the audiovisual congruence, in addition to the synchronization between auditory and visual accents. We found the highest level of congruence when the accents were periodic and frequently synchronized. Conversely, the congruence level was lower when the auditory and visual accents were less frequently synchronized. The contribution of auditory periodicity to the perceived congruence was greater than that of visual periodicity. Additionally, when we used stimuli consisting of synchronized and asynchronized audiovisual elements, we observed an effect of context on congruence with respect to continuous ratings. When we presented a stream of asynchronized audiovisual accents after a synchronized stream, the level of high congruence was maintained for a short period. In contrast, just after presentation of a steam of asynchronized accents, we found that the congruence level of a stream of synchronized accents was lower than the original level.

3:10–3:25 Break

3:25

1pMU6. Making sense of multiple senses. Barbara Shinn-Cunningham (Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu)

Our brains are expert at processing multi-sensory inputs. We automatically "fuse" sounds and sights that come from one source– to the point that we cannot turn off cross-sensory perceptual interactions. Examples like the McGurk effect (in speech perception), the "Flash-Beep illusion" (in temporal perception), and the "Ventriloquism Effect" (in spatial perception) demonstrate how robust and obligatory cross-sensory integration can be. Indeed, these sensory "illusions" are not truly "illusions," but rather examples of how perceptually integrated sensory inputs are in the real world, compared to how we view them in the laboratory. Such cross-modal effects also underscore differences in the kind of information that vision and audition "specialize" in, and what our brain trusts. Specifically, visual inputs convey spatial information precisely, but temporal information poorly. Conversely, auditory inputs convey temporal information precisely, but spatial information poorly. Recent cognitive neuroscience work from our lab gives insight into how the brain handles sensory specialization and cross-sensory coding, revealing distinct cortical networks that support visuo-spatial and auditory-temporal processing. These results will be reviewed, along with their implications for multi-sensory coding in films, video games, and other forms of multimedia.

3:45

1pMU7. Telematic systems seen from a music instrument building perspective. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

New digital technologies enabled the Virtual Reality movement in the 1990s, which led to the euphoric belief that we could virtually teleport ourselves to a distant location with such great realism that the user would lose the awareness for the enabling technology. Especially, in the particular case of music collaborations, however, it has meanwhile become clear that this goal is unreachable because of the unavoidable transmission latency over long distances and other secondary factors. This presentation explores how we can create new and better audiovisual design goals for telematic music collaborations, by treating telematic systems as a new form of musical instrument. From this viewpoint, the creation of new forms of art for the new medium become a central challenge, and not only the development of better telematic technology. In particular, the new works need to focus on the new affordances that the telematic system brings with it, and not only circumnavigate current system limitations. [Work supported by NSF grant #12293911.]

4:05

1pMU8. Virtual acoustics in multimedia production—Beyond enhancing the acoustics of concert halls. Wieslaw Woszczyk (Virtual Acoust. Technol. Lab, McGill Univ., Schulich School of Music, 527 Sherbrooke St. West, Montreal, QC H3A1E3, Canada, wieslaw@music.mcgilll.ca)

The paper describes a range of applications of virtual acoustics, the rendering of artificial acoustic spaces, which allow musicians to interact with ambient spaces created in real time. With a number of loudspeakers suitably distributed within a physical enclosure, such a projection system can be used to introduce a range of sound fields, which may effectively transform the acoustic environment to become a creative partner in multimedia production. A necessary component in this system is a low-latency, high-resolution multichannel convolution engine that converts a live audio signal into a structured ambient response, creating a scene in real time. Scenes can be changed to suit various goals of production and sonic narration. A number of techniques have been used to capture and modify impulse responses including temporal segmentation, shaping of magnitude envelope, noise reduction, spectral enrichment, time shifting and alignment, and parallel and sequential convolution. With these methods, artists may interact with novel acoustic responses as if they were musical instruments. Artists claim that many of the familiar as well as novel acoustic responses stimulate their creativity. Flexible variable virtual acoustics opens additional creative possibilities when employed as a component of a recording studio, either directly or when rendered over headphones.

MONDAY AFTERNOON, 28 NOVEMBER 2016 SOUTH PACIFIC 3, 1:00 P.M. TO 2:00 P.M.

Session 1pNSa

Noise: Sound Design II

André Fiebig, Cochair
HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

Takeshi Toi, Cochair

Patricia Davies, Cochair
Ray W. Herrick Labs., School of Mechanical Engineering, Purdue University, 177 South Russell Street, West Lafayette, IN 47907-2099

Contributed Papers

1:00

1pNSa1. Influence of rotor eccentricity on electromagnetic vibration and noise in permanent magnet synchronous motor with different slot-pole combinations. Fu Lin and Shuguang Zuo (Tongji Univ., Rm. 403B, Clean Energe Automotive Eng. Ctr., No. 4800, Cao'an Rd., Shanghai 201804, China, linfu_911@163.com)

Rotor eccentricity (RE) is a common fault in permanent magnet synchronous motor (PMSM). In this paper, a multiphysics model is proposed to investigate the variation of electromagnetic vibration and noise due to two types of RE including static eccentricity (SE) and dynamic eccentricity (DE). Maxwell stress tensor is first employed to derive the spatial and temporal changes of electromagnetic force from eccentricity, which are validated by 2-D decomposing of radial force calculated by a finite element (FE) model. Then, based on the proposed multiphysics model, the electromagnetic vibration and noise considering RE with different eccentricity levels and types is calculated. The vibration and noise changes with eccentricity are all explained by the force harmonics induced by eccentricity. It is concluded that the influence of RE on vibration and noise is closely related to the slot-pole combination. The vibration and noise in PMSM with higher space order of force is more sensitive to RE. It is also found that, due to the low space order force induced by RE, vibration and noise peaks would transfer to the frequency band close to lower modal frequencies.

1:15

1pNSa2. Investigation of vibration and noise characteristics in axial flux permanent magnet synchronous motor with different magnet shapes. Wenzhe Deng, Shuguang Zuo, Fu Lin, and Shuanglong Wu (Clean Energy Automotive Eng. Ctr., Tongji Univ., No.4800 Cao'an Rd., Jiading District, Shanghai 201804, China, deng_wenzhe@foxmail.com)

This paper investigates the vibration and noise characteristics of axial flux permanent magnet (AFPM) motor with different magnet shapes. A multiphysics model is developed to predict electromagnetic vibration and noise of AFPM motor which contains three steps. In the first step, nodal force exerted on the surface of stator tooth is calculated using 3D finite element method (FEM). Then, structural analysis is performed to investigate the structure natural characteristics and the nodal force calculated is mapped into structural model as the excitation in the second step. In the third step, the vibration and noise of AFPM motor are calculated through modal superposition method (MSM) and boundary element method (BEM) respectively. Vibration and noise characteristics of AFPM motor with different magnet shapes are investigated based on the multiphysics model. The results reveal that AFPM motor with sine-shaped magnet demonstrates lower noise level compared to AFPM motor with sector- or cylinder-shaped magnet.

1:30

1pNSa3. Damping contributions of sound absorbing materials in three-dimensional spaces to obtain efficient layouts. Takao Yamaguchi (Mech. Sci. and Technol., Gunma Univ., 1-5-1 Tenjincho, Kiryu, Gunma 376-8515, Japan, yamagme3@gunma-u.ac.jp)

In this research, we have proposed a method for determining an efficient layout of sound absorbing materials in three-dimensional spaces using damping contributions with eigenmodes as weight coefficients. We use the complex effective density and the complex bulk modulus of elasticity for finite elements of sound absorbing material. We formulate discrete equations using the elements. Solutions to the complex eigenvalue problem of the discrete equations are expanded with a small parameter that is related to the damping. By adopting the first order approximation with respect to the small parameter, we derive the dominant equations of motion. By arranging these derived equations, the contribution of each element to modal loss factors can be obtained in consideration of multiple modes. According to these contributions, the sound absorbing materials are filled in the spaces. In this proposed method, the components of eigenvectors at the evaluation points are used as the weight coefficients for the damping contributions. The weight coefficients play an important *role in intensively damping the acoustic modes that exert dominant influences on the sound pressure at the evaluation points. We explain numerical examples to verify the effectiveness of this layout method.*

1:45

1pNSa4. Evaluation of low cost microphone for active noise control in duct. Pedro Henrique R. Lima (Universidade de Brasília - UNB, Rua 138, Aparecida de Goiânia 74946-450, Brazil, cefas_hrl@hotmail.com)

Economic feasibility and cost minimizing of experimental setups is one of the most important factors in many research and this is no different for active noise control (ANC) in ducts. Cheaper acquisition and assembly of equipment is desirable, provided you keep the same efficiency of acquisition, filtering, control, and attenuation. This study aims to compare the performance of low cost microphones with precision microphones, when applied to the acoustic control system in feedforward configuration. This system makes use of a control board with Digital Signal Processor (DSP) in order to identify, analyze, and compare systems of acoustic control in real time. To validate the performance of the sensors, an experimental bench was setup which included a PVC duct and speakers as noise source and actuator. Microphones are the error and reference sensors. The system performance is obtained by a sound pressure meter acting on the outlet duct.

MONDAY AFTERNOON, 28 NOVEMBER 2016 SOUTH PACIFIC 3, 2:15 P.M. TO 5:15 P.M.

Session 1pNSb

Noise: Acoustics of Microperforated Materials

J. S. Bolton, Cochair
Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, 177 S. Russell St., West Lafayette, IN 47907-2099

Masahiro Toyoda, Cochair
Kansai University, 3-3-35, Yamate-cho, Suita-shi, Osaka 564-8680, Japan

Chair's Introduction—2:15

Invited Papers

2:20

1pNSb1. Modeling and formulation of microperforated panels in the finite-difference time-domain method. Daiki Eto, Masahiro Toyoda, and Yasuhito Kawai (Kansai Univ., 3-3-35, Yamate-cho, Suita-shi, Osaka 564-8680, Japan, k832661@kansai-u.ac.jp)

A microperforated panel (MPP) was first developed by Maa and is recently regarded as one of the most promising absorption materials for next generation. Considering the approximation formula suggested by Maa, impedance of the perforations cannot be directly implemented in time-domain calculations because the impedance depends on frequency. Some new ideas are therefore necessary to take into account the effects of an MPP in time-domain analyses. Herein, a method to consider effects of an MPP in the finite-difference time-domain (FDTD) analyses is proposed. An MPP is considered as a boundary condition and the frequency dependence in the calculation is removed by replacing the frequency-dependent terms with constant values which are derived based on the multiple regression analysis. Additionally, the stability condition of the MPP boundary for one-dimensional sound field is developed. Absorption characteristics of MPP absorbers with back air layers of various depths are calculated numerically and compared with analytical ones. Consequently, stable calculations are achieved by satisfying both the CFL condition and the stability condition of the MPP boundary and the high prediction accuracy and wide applicability are confirmed.

2:40

1pNSb2. Computational investigation of microperforated materials: End corrections, thermal effects, and fluid-structure interaction. J. S. Bolton (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 S. Russell St., West Lafayette, IN 47907-2099, bolton@purdue.edu), Thomas Herdtle (3M Corporate R & D, SEMS, Predictive Eng. and Computational Sci., 3M Co., St. Paul, MN), and Nicholas Kim (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN)

The concept of microperforated noise control treatments was introduced by Maa 1975; in that theory, the transfer impedance of the microperforated layer was calculated based on oscillatory viscous flow within a small cylinder combined with resistive and reactive end corrections. Recently, new manufacturing procedures have dramatically lowered the cost of these materials, and as a result, there has been renewed interest in studying their properties. Since 1975, Maa's original theory has been widely used. However, in principle, that theory can only be used to describe cylindrical perforation, while in practice, perforations are rarely cylindrical. In addition, there have been questions about the dependence of end corrections on frequency, and on the effect of coupling between the motion of the fluid in the perforations and the solid sheet in which they are formed. Additionally, in his original paper, Maa drew a distinction between the dissipative properties of thermally conducting and adiabatic materials. The latter topic, in particular, has not been considered by any investigators since the idea was introduced. The purpose of this presentation is to introduce numerical tools that can be used to address the open questions mentioned above and to highlight important results obtained by using those tools.

3:00

1pNSb3. Limp microperforated panel finite elements and its application. Takeshi Okuzono and Kimihiro Sakagami (Architecture, Kobe Univ., 1-1 Rokkodai-cho, Kobe, Hyogo 657-8501, Japan, okuzono@port.kobe-u.ac.jp)

Microperforated panel (MPP) absorbers provide relatively broadband sound absorption with the superior material properties in durability, recyclability, and flexibility of design, and they can be considered as an attractive alternative of conventional sound absorbers. Recently, with the rapid progress of computer technology, numerical methods such as finite element method (FEM) and boundary element method have become powerful design tools for developing MPP absorbers with the consideration of advantage in handling arbitrary geometries, and there exist some computational models of MPP. This paper presents a simple and computationally efficient limp MPP elements to treat MPP absorbers in finite element analysis and the performance is demonstrated in various problems. First, the theory of MPP elements is presented briefly. Then, the validity is shown using impedance tube problems, in which the absorption characteristics of some MPP absorbers calculated using FEM are compared with the theoretical values. Finally, the applicability to room acoustics simulations is demonstrated by calculating transfer functions and impulse responses of rooms with practical dimensions.

3:20

1pNSb4. Sound absorption of micro-perforated panels in complex vibroacoustic environment: Modeling, mechanism, and applications. Li Cheng (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon 00000, Hong Kong, li.cheng@polyu.edu.hk)

As an environmental-friendly and broadband sound absorption materials, micro-perforated panels start to find their wide applications in compact mechanical systems featuring complex acoustic behaviors. Evidences show that, under such circumstances, MPPs work quite differently from the way they behave in traditional architectural or building acoustic applications. This paper summarizes some of our recent work on the modelling, analyses and optimizations of mechanical systems with embedded MPP absorbers. It is demonstrated that the complex acoustic environment in which the MPP is put in creates formidable technical challenges in terms of modeling, analyses, and optimization on one hand, but also opportunities for achieving better sound absorption by making use of the vibroacoustic coupling principles for various applications. [This project was supported by the National Natural Science Foundation of China, Grant No. 11272272.]

3:40–3:55 Break

3:55

1pNSb5. Optimum duct liners and modal filters. Mats Åbom, Stefan Sack, and Raimo Kabral (The Marcus Wallenberg Lab., KTH-The Royal Inst of Technol., Teknikringen 8, Stockholm 10044, Sweden, matsabom@kth.se)

Two strategies for applying micro-perforated plates (MPP:s) to create duct sections with a high or optimum damping of sound are discussed. The first is based on realizing the so called Cremer impedance by creating a locally reacting wall boundary ("liner") that consists of a MPP backed by an air cavity. This principal can be used to realize the Cremer impedance at a single frequency where typically a damping of several hundred dB/m can be achieved. The proposed solution is robust and gives high damping also in a wide band around the target (optimum) frequency. The second strategy is based on MPP plates arranged along pressure nodes (minima) for a given acoustic cross-mode. This means that the MPP will be located at modal velocity maxima, thereby creating a large modal damping. In principle both strategies are optimum for a particular mode and can be seen as a modal filter, i.e., a device that reduces or "kills" a particular incident mode. The proposed strategies have been investigated theoretically as well as numerically and a number of prototypes have been built and tested. In particular, two applications will be discussed: IC-engine compressor noise and aircraft ventilation noise.

4:15

1pNSb6. Three-dimensional space sound absorbers using microperforated panels. Kimihiro Sakagami (Dept. of Architecture, Grad. School of Eng., Kobe Univ., Rokkodai 1-1, Nada, Kobe, Hyogo 657-8501, Japan, saka@kobe-u.ac.jp), Masahiro Toyoda (Dept. of Architecture, Kansai Univ., Suita, Osaka, Japan), Motoki Yairi (Kajima Tech. Res. Inst., Chofu, Tokyo, Japan), and Takeshi Okuzono (Dept. of Architecture, Grad. School of Eng.., Kobe Univ., Kobe, Hyogo, Japan)

A microperforated panel (MPP) is one of the most promising alternatives of the next-generation sound absorbing materials. It has a large potential for various use. However, its conventional use is to place it with a rigid-back wall and an air-cavity in-between. Therefore, its arrangements in a room are limited on to room boundaries, and in some cases, this may cause a problem in room design and its strength: for example, in Japan, it is not allowed by regulation to use it for a wall. Therefore, the authors have been looking at the possibility for using MPPs for space sound absorbers without any backing structure. This can allow MPPs for wider use: a three-dimensional shape MPP space absorbers can be easily produced if a suitable material is used. They can be placed easily in various places: they can be put on the floor or hang from the ceiling, etc. Besides, MPPs can be made of designable, lightweight materials, which can add designability to the absorbers. In this paper, the main results of our studies on three-dimensional MPP space absorbers (mainly cylindrical and rectangular three-dimensional MPP space absorbers) are introduced and summarized.

4:35

1pNSb7. Equivalent gradient properties of porous material with macro void and panel structures. Fengxian Xin (MOE Key Lab for Multi-functional Mater. and Structures & State Key Lab for Strength and Vib. of Mech. Structures, Xi'an Jiaotong Univ., Xi'an, China), Changyong Jiang (HKU-ZIRI and Dept. of Mech. Eng., The Univ. of Hong Kong, Haking Wong Bldg., Rm. 704, Hong Kong 00000, Hong Kong), Tianjian Lu (MOE Key Lab for Multi-functional Mater. and Structures & State Key Lab for Strength and Vib. of Mech. Structures, Xi'an Jiaotong Univ., Xi'an, China), and Lixi Huang (HKU-ZIRI and Dept. of Mech. Eng., The Univ. of Hong Kong, Hong Kong, Hong Kong, lixi@hku.hk)

In order to reduce the interface sound reflection, best sound absorption may be achieved by having a low acoustic impedance at the air-absorber interface and a gradient property is used inside the absorber. This study explores how such gradient properties may be realized by a distribution of voids and/or perforated panel structures. Finite-element simulation for the basic module is conducted to reveal the relationship between the design and equivalent fluid properties. The gradient of various properties are then identified. Genetic algorithm is used to optimize the geometrical parameters for best absorption performance under the constraint of a given thickness and a set of boundary conditions. The optimized design is then considered as parallel-series arrangement of homogeneous layers each with different acoustic impedance. For a chosen practical application, a feasible design is determined and its performance is compared with the traditional absorber of optimized, uniform properties. Gains in broadband sound absorption performance is analyzed with a single-digit descriptor relevant to the application.

4:55

1pNSb8. Thin low frequency sound absorbers based on shunted loudspeakers. Jiancheng Tao, Chaonan Cong (Key Lab. of Modern Acoust., Inst. of Acoust., Nanjing Univ., Nanjing, China), and Xiaojun Qiu (School of Elec., Mech., and Mechatronic System, Univ. of Technol. Sydney, Sydney, NSW 2007, Australia, xiaojun.qiu@uts.edu.au)

There are two main reasons for using shunted loudspeakers as sound absorbers. One is that the absorbers can be made very thin for the low frequency sound with electronic energy dissipation mechanism and the other is that their absorption performance can be adjusted with the shunt circuit. A shunted loudspeaker typically has high absorption coefficients around its resonance frequencies which are determined by the loudspeaker unit and the shunt circuit together. To extend the effective frequency range of the shunted loudspeaker, one method is constructing more resonant sub-systems in the absorber to have absorption at more resonance frequencies. In this research, two designs based on shunted loudspeakers will be introduced. One is the composite absorber with a micro-perforated panel in front of a shunted loudspeaker, which combines two resonance subsystems to provide broadband absorption. The other is a shunted loudspeaker with a specifically designed shunt circuit for the absorption of three tonal components. Experimental results in an impedance tube will be reported to demonstrate the feasibility of the designs based on the combination concept of multiple resonances.

Session 1pPA

Physical Acoustics: Advances in Hazards Monitoring with Distributed Infrasound Sensor Networks

Milton A. Garces, Cochair
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Chair's Introduction—1:00

Invited Papers

1:05

1pPA1. Infrasonic and seismic eruption tremors and their relation to magma discharge rate: A case study for the 2011 eruption of Shinmoe-dake, Japan. Mie Ichihara (Earthquake Res. Inst., Univ. of Tokyo, 1-1-1 Yayoi, Bunkyo-ku, Tokyo 113-0032, Japan, ichihara@eri.u-tokyo.ac.jp)

Infrasound monitoring is a useful tool to capture surface activity of volcanoes. Previous works have attempted to connect observations of infrasonic eruption tremors with magma discharge rate that is an important parameter to characterize an eruption. Power law scaling relations with different power indices have been proposed based on various datasets and models. Independently, different power law relations have been proposed between seismic eruption tremors and magma discharge rate. The 2011 eruption of Shinmoe-dake, Japan, provides an excellent dataset with which to investigate these relationships for a sustained explosive eruption sequence. Magma discharge rates are well constrained by geodetic, geologic, and remote sensing methods. Seismic and infrasonic data were recorded close to the vent. Linear power law relationships are found to fit all pairs of variables (seismic eruption tremor power, infrasonic eruption tremor power, and magma discharge rate) in the quasi-stable or slowly growing stages of the eruption. Existing models do not fully explain the observed relationships. It is proposed that the eruption generated eruption tremors in the ground and atmosphere by successive explosions in the conduit and the linear relation is observed only under the stable condition of eruptive behaviors.

1:25

1pPA2. Explosion energy estimates based on propagation modeling of observed infrasound airwaves. Keehoon Kim and Arthur Rodgers (Geophysical Monitoring Program, Lawrence Livermore National Lab., 7000 East Ave., L-103, Livermore, CA 94550, kim84@llnl.gov)

Energetic explosions in the atmosphere produce blast waves, which propagate as infrasound (low-frequency sound waves) and can be detectable at long ranges. Numerous empirical models have been developed to relate explosion energies of events (in terms of trinitrotoluene-equivalent yield) to the metrics of observed infrasonic signals (e.g., peak amplitude, period, and impulse per unit area). However, the existing empirical models often do not take into account variability of atmosphere (e.g., temperature, pressure, and wind) and an empirical model determined in a certain meteorological specification may not be valid in another weather scenario. In this case, unmodeled propagation effects can compromise the accuracy of source energy estimates. We propose another approach to determine explosion energy based on numerical simulations. Infrasound propagation in a given meteorological circumstance is simulated by a full 3-D finite difference method providing accurate Green's functions between the source and receivers. With the numerical Green's functions full-waveform inversion of infrasonic signals is performed to obtain the source time history of the event, and the metrics of the acoustic source time function are related to the energy of explosion source. The technique is applied to controlled chemical explosions and volcanic explosions for verification.

1:45

1pPA3. Detection of snow avalanche locations using infrasound array data. Nobuo Arai (Nagoya Univ., Furo-cho, Chikusa-ku, Nagoya 464-8601, Japan, arai.nobuo@e.mbox.nagoya-u.ac.jp), Toshiaki Imai, Masaya Otsuki, Yoshihiko Saito (Yuki-ken Snow Eaters Inc., Sapporo, Japan), Takahiko Murayama, and Makiko Iwakuni (Japan Weather Assoc., Tokyo, Japan)

Infrasound observation measures the energy radiated by a snow avalanche into the atmosphere and can be used to detect snow avalanches over large areas. Our research team has conducted infrasound observation in Tokamachi, Niigata Prefecture, Japan, during the last three winter seasons. First, between January and April 2013, we deployed an infrasound sensor in front of a selected slope, obtained visual observations using a web camera, and identified infrasound signals generated by snow avalanches. During the second season

(2013-2014 winter), we deployed two infrasound sensors approximately 1 km apart and obtained the attenuation characteristics of infrasound signals. In the 2014-2015 winter season, we deployed three sensors spaced at 1-2 km in a triangular array and attempted automatic extraction of signals associated with snow avalanches from observed raw data using time-domain processing. For the extracted signals, the locations of snow avalanches were estimated with the cross-correlation method. Twelve events were detected and located. The estimated locations were in an area with many steep slopes. An infrasound array monitoring system with real-time processing is capable of providing significant information on snow avalanche activity.

2:05

1pPA4. ARISE project capabilities for hazards monitoring. Elisabeth Blanc, Nicolas Brachet, Alexis Le Pichon (CEA/DAM/DIF, CEA/DAM/DIF, Arpajon 91297, France, elisabeth.blanc@cea.fr), Emanuele Marchetti (Dept. of Earth Sci., Firenze Univ., Firenze, Italy), Thomas Farges (CEA/DAM/DIF, Arpajon, France), Maurizio Ripepe (Dept. of Earth Sci., Firenze Univ., Firenze, Italy), Pierrick Mialle (Comprehensive Nuclear-Test-Ban Treaty Organization, Vienna, Austria), and Philippe Husson (Meteo France, Toulouse, France)

The ARISE (Atmospheric dynamics InfraStructure in Europe) project funded by the European Union combines the International infrasound monitoring system developed for the CTBT (Comprehensive nuclear-Test-Ban Treaty) verification, the NDACC (Network for the Detection of Atmospheric Composition Changes) lidar network, European observatories at mid latitudes (OHP observatory), tropics (OPAR observatory), high latitudes (ALOMAR), the European infrasound stations and satellites for the study of the dynamics of the atmosphere from ground to thermosphere. The ARISE network is unique by its coverage from equatorial to polar regions and the involved scales both in time (to tens of years) and space (one to thousands of kilometers). One of the major objective of ARISE is the monitoring of extreme events including lightning, earthquakes, tornadoes, and volcanoes. A review of the ARISE results will be presented. Volcano monitoring by the infrasound technology presents a strong interest for aviation safety in case of eruption of distant non instrumented volcanoes in broad coverage in complement to satellite observations, which can be limited by the cloud cover. One of the ARISE objectives is to provide eruption notifications including confidence index representative on propagation conditions and additional analysis results. Such studies interest the VACCs (including VAAC Toulouse) and the IAVWOPSG.

Contributed Papers

2:25

1pPA5. Past, present, and future infrasound volcanic's explosion detection capability in Southeast Asia. Benoit Taisne, Anna Perttu (Earth Observatory of Singapore, 50 Nanyang Ave., N2-01b-25, Singapore 639798, Singapore, btaisne@ntu.edu.sg), and Milton Garces (Infrasound Lab., Univ. of Hawaii, Manoa, HI)

On August, 26th 1883, meteorologist across Europe, observed an anomaly in the atmospheric pressure measurements, turns out that it was generated by an explosive eruption of Krakatau that happened more than 11,000 km away. Nevertheless, detection is not easily granted and will be affected by regional to global atmospheric conditions, mainly winds and temperature profile. Other atmospheric phenomena, such as clouds, would not be affecting infrasound, but might reduce the chance of observing the expanding ash cloud from space. Those limitations are of paramount importance when talking about aviation safety. The ever-growing demand for air-traffic is accommodated by increasing the number of travellers in the air at any one time, bringing with it the need to improve early detection of volcanic ash cloud so that planes could be rerouted to safer paths. The issue is especially pertinent in Southeast Asia with the recent boom in air-traffic, and the 749 potentially active volcanoes in the region. In terms of volcanic explosivity index (VEI), the probability of VEI-4 (ash up to 25 km and spreading for more than 1000 km) in any ten years span in SE-Asia is one, while this probability is of 0.630 and 0.135 for VEI-5 and 6, respectively. In this presentation, we will focus on the remote detection capability that we have with today's instruments, and how we can improve it in the future.

2:40

1pPA6. Acoustic surveillance of hazardous volcanic eruptions (ASHE) in Asia. Maria Ngemaes (Palau National Data Ctr. and National Weather Service, Palau National Weather Service and National Data Ctr., PO Box 520, Koror, Koror 96940, Palau, maria.ngemaes@noaa.gov), Benoit Taisne (Earth Observatory of Singapore, NTU, Singapore, Singapore), Takahiku Murayama (Japan National Data Ctr. and Weather Assoc., Tokyo, Japan), Elisabeth Blanc (Commissariat à l'Energie Atomique et aux Energies Alternatives, Arpajon, France), Andrew Tupper (Bureau of Meteorol., Melbourne, VIC, Australia), Pierrick Mialle (Comprehensive Nuclear-Test-Ban-Treaty Organization, Vienna, Austria), Hee-Il Lee (Korea National Data Ctr. and Inst. of GeoSci. and Mineral Resources, Daejeon, South Korea), and Milton A. Garces (HIGP Infrasound Lab., Univ. of Hawaii at Manoa and Palau National Data Ctr., Kailua-Kona, HI)

The ASHE Ecuador (2004-2012) collaboration between Ecuador, Canada, and the United States demonstrated the capability to use real-time infrasound to provide low-latency volcanic eruption notifications to the Volcano Ash Advisory Center (VAAC) in Washington DC. The Atmospheric dynamics Research Infrastructure in Europe (ARISE, 2012-2018) supported by the European Commission fosters integrating innovative methods for remote detection and characterization of distant eruptive sources through collaborations with the VAAC Toulouse and the Comprehensive Nuclear-Test-Ban-Treaty Organization (CTBTO). The ASHE Asia project proposes an international collaboration between the Earth Observatory of Singapore, the VAAC Darwin, National Data Centers in Japan, Korea, and Palau, and will receive the support of ARISE, to provide improved early notification of potentially hazardous eruptions in Asia and the Western Pacific using a combination of established technologies and next-generation mobile sensing systems. The increased availability of open seismo-acoustic data in the ASEAN region as well as recent advances in mobile distributed sensors networks will facilitate unprecedented rapid progress in monitoring remote regions for early detection of hazardous volcanic eruptions and other natural disasters.

2:55

1pPA7. Seven years of infrasound monitoring of volcanic eruptions of Sakurajima. Hee-il Lee, Il-Young Che (Earthquake Res. Ctr., Korea Inst. of GeoSci. and Mineral Resources, 124 Gwahang-no, Yuseong-gu, Daejeon 305-350, South Korea, leel@kigam.re.kr), and Milton Garces (Infrasound Lab., Univ. of Hawaii at Manoa, Kailua-Kona, HI)

Korea Institute of Geoscience and Mineral Resources (KIGAM) is operating an dense infrasound array network which is comprised of eight seismo-acoustic array stations (BRDAR, YPDAR, KMPAR, CHNAR, YAGAR, KSGAR, ULDAR, and TJIAR) and we are continuously monitoring all the natural and artificial infrasonic events occurred in and around Korean peninsula with this network. Sakurajima volcano is one of the most active volcanoes in the world to the point where the Volcanic Ash Advisory Center in Tokyo issued 6,012 warnings over the past 7 years from 2009 to 2015. We have checked and analyzed all the infrasound signals possibly associated with the eruptions chronology of Sakurajima volcano based on the Tokyo VAA (Volcanic Ash Advisories) information, in order to know how daily, seasonal and yearly atmospheric changes affect the detection capability of each infrasound station. The result of preliminary analysis will be presented and how the geographical location of the infrasound network which is fan-shaped to the Sakurajima volcano is affected by stratospheric wind dynamics will be discussed.

3:10–3:25 Break

3:25

1pPA8. Characterizing infrasonic ocean ambient noise in Northwest Pacific. Alexis Le Pichon, Julien Vergoz (CEA, DAM, DIF, CEA, DAM, DIF, Arpajon F-91297, France, julien.vergoz@cea.fr), Christoph Pilger, Lars Ceranna (B4.3, BGR, Hannover, Germany), and Fabrice Ardhuin (LPO/CNRS, IFREMER, Brest, France)

The ability of the International Monitoring System (IMS) infrasound network to detect atmospheric explosions and events of interest strongly depends on station specific ambient noise which includes both incoherent wind noise and real coherent infrasonic waves. To characterize the coherent ambient noise, a broadband array processing was performed with IMS continuous waveform archive from 2007 to 2016 using the Progressive Multi-Channel Correlation algorithm (PMCC). The processing parameters include a new implementation of adaptive frequency dependent window length and a logarithmic band spacing. Such configuration allows to better discriminate between interfering signals with improved accuracy in the wave parameters estimations. Multi-year comparisons between the observed and modeled directional microbarom amplitude variations at several IMS stations using two-dimensional wave energy spectrum ocean wave products are performed to build of a reference database of infrasound oceanic sources in Northwest Pacific. The expected benefits of such studies concern the use of multi-year complementary data to finely characterize coupling mechanisms at the ocean-atmosphere interface. In return, a better knowledge of the source of the ambient ocean noise opens new perspectives by providing additional integrated constraints on the dynamics of the middle atmosphere and its disturbances where data coverage is sparse.

3:40

1pPA9. Analysis of explosive sources using a distributed network of infrasound arrays. Philip Blom (Los Alamos National Lab., Los Alamos National Lab., PO Box 1663, Los Alamos, NM 87545, pblom@lanl.gov), Omar Marcillo, Garrett Euler (Los Alamos National Lab., Los Alamos, NM, NM), Fransiska Dannemann, and Junghyun Park (Southern Methodist Univ., Dallas, TX)

Analysis of signatures observed on a distributed network requires a model describing the propagation of the energy between network nodes as well as a statistical framework to quantify confidence in analysis conclusions. For the case of infrasound analysis, the propagation model must account for the inhomogeneous, dynamic nature of the atmosphere and the resulting temporal and spatial variations of propagation effects as well as the anisotropic nature of infrasonic propagation due to the dependence on the directionality of the winds. Association and localization methods have been developed utilizing a Bayesian framework to identify explosive sources from distant, infrasonic observations with possible applications to other infrasonic sources. The confidence of the analysis conclusions is dependent on the accuracy and realism of the likelihood definitions utilized in the method, which are therefore an area of ongoing research. An overview of the association and localization methods will be presented along with several possible definitions of the infrasound likelihoods for comparison using explosive infrasonic signatures.

3:55

1pPA10. Cosmos supersonic reentry analysis: A case study using traditional and mobile infrasound arrays. Julie Schnurr (HIGP, Univ. of Hawaii Manoa, 1680 East-West Rd., Honolulu, HI 96822, jschnurr@hawaii.edu), Milton Garces (HIGP, Univ. of Hawaii Manoa, Kailua-Kona, HI), and Anthony Christie (ICS, Univ. of Hawaii Manoa, Honolulu, HI)

On August 31, 2015, the fiery reentry of the Soviet Cosmos 1315 was sighted over the Hawaiian Islands. The supersonic reentry of the satellite was detected with both traditional infrasound arrays located on the Big Island, Maui and Kauai, and with iPods using the RedVox infrasound recorder on the Big Island and Oahu. We apply array-processing techniques to reconstruct the trajectory of the satellite. In addition, we have developed a standardized multi-resolution framework that uses the Gabor limit to define a scaled set of frequencies and windows for transient feature extraction. Metrics are evaluated over a set of standardized log-scaled time windows in order to characterize waveforms. I describe a method to detect transients in infrasonic arrays with low signal to noise where beam forming is not possible, and apply it to the analysis of signals produced by the supersonic Cosmos 1315 reentry.

4:10

1pPA11. Interactions between intermittent gravity waves and infrasounds. Christophe Millet (CEA, DAM, DIF, CEA, DAM, DIF, Arpajon 91297, France, christophe.millet@cea.fr), Bruno Ribstein (CMLA, ENS, Paris, France), and Francois Lott (LMD, ENS, Paris, France)

Infrasound propagation modeling requires accurate wind speed and temperature gradients. Even though the accuracy of atmospheric specifications is constantly improving, it is well-known that the main part of gravity waves is not resolved in the available data. In most infrasound modeling studies, the unresolved gravity wave field is often represented as a deterministic field that is superimposed on a given average background state. Direct observations in the lower stratosphere show, however, that the gravity wave field is very intermittent, and is often dominated by rather well-defined wave packets. These observations suggest that the vertical spectra are likely to result from ensemble averages of quite narrow-banded periodograms. In this study we sample the spectrum by launching few monochromatic waves and choose the gravity-wave properties stochastically to mimic the intermittency. The statistics of acoustic signals are computed by decomposing the original signal into a sum of modal pulses. Owing to the disparity of the gravity and acoustic length scales, the interaction can be described using a multiple-scale analysis and the appropriate amplitude evolution equation involves certain random terms that are related to the gravity wave sources. More specifically, it is shown how the unpredictable low-level small-scale dynamics triggers multiple random stratospheric waveguides in which high frequency infrasound components can propagate efficiently.

4:25

1pPA12. Bayesian model averaging of infrasound propagation models. Christophe Millet (CEA, DAM, DIF, CEA, DAM, DIF, Arpajon 91297, France, christophe.millet@cea.fr) and Michaël Bertin (CMLA, ENS, Cachan, France)

In infrasound propagation modeling, we often face a large number of potential models with only a limited number of recorded signals to conduct inference from. Prediction of acoustic signals is a complex issue due to constantly changing atmospheric conditions and to the random nature of small-scale flows (turbulence, gravity waves). Thus, the uncertainty over which model to use (both for atmospheric specifications and acoustic propagation) is an important aspect of forecasting and indeed any inference from data. In standard practice data analysts typically select a model from some class of models (linear propagation models, ray tracing techniques,...) and then

proceed as if the selected model had generated the recorded signal. Such an approach ignores the uncertainty in propagation model selection, leading to over-estimated confidence intervals. In the present work, we use Bayesian Model Averaging (BMA) as a basis for inference about parameters of interest. This approach offers a systematic method for checking the robustness of one's results to alternative model specifications. The method's performance is demonstrated using several data sets from the International Monitoring System (IMS). Our results demonstrate that a Bayesian probabilistic approach together with methods of model reduction to infrasound propagation can improve significantly the posterior distributions from the continuous stream of automatic detections of IMS infrasound stations.

4:40

1pPA13. A big data pipeline for temporospatial infrasound analysis. Anthony Christe (Infrasound Lab., Univ. of Hawaii at Manoa, 1314 Victoria St., Apt. 1402, Honolulu, HI 96814, achriste@hawaii.edu), Milton Garces (Infrasound Lab., Univ. of Hawaii at Manoa, Kailua-Kona, HI), Julie Schnurr (Infrasound Lab., Univ. of Hawaii at Manoa, Honlulu, HI), and Steven Magana-Zook (Lawrence Livermore National Lab., Livermore, CA)

Smartphone and IoT sensors allow us to build dense distributed sensor networks that supplement traditional networks. We examine next-generation technologies powering the acquisition, analysis, and reporting of infrasound data. With the advent of distributed computing, managing data flow from sensor networks has become increasingly complex. Due to the increased volume, velocity, and variety of data being produced, data acquisition, storage, analysis, and reporting techniques are evolving from single server to distributed computation architectures. In collaboration with Lawrence Livermore National Laboratory, we survey and implement several Big Data technologies to tackle these issues. We implement a system that allows distributed acquisition using Akka actors, a custom time synchronization protocol, intermediate persistent queues with Apache Kafka, long term persistence using a NoSQL database, and real-time analysis and reporting with Apache Spark and Python. We describe how these software components work together to provide acquisition and analysis for recent infrasound signatures of surf, volcanic eruption, and supersonic sources.

MONDAY AFTERNOON, 28 NOVEMBER 2016 — CORAL 5, 1:15 P.M. TO 4:55 P.M.

Session 1pPP

Psychological and Physiological Acoustics: Spatial Hearing and Its Applications II

Kazuhiro Iida, Cochair
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Yukio Iwaya, Cochair
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Invited Papers

1:15

1pPP1. Localization-based audiovisual speech-recognition benefit with degraded spatial cues. Sterling W. Sheffield (Walter Reed National Military Medical Ctr., 4954 North Palmer Rd., Bethesda, MD 20889, sterling.sheffield.ctr@mail.mil), Harley J. Wheeler (Walter Reed National Military Medical Ctr., Harrisonburg, VA), and Douglas S. BrungartJoshua G. Bernstein (Walter Reed National Military Medical Ctr., Bethesda, MD)

Degraded sound-localization accuracy is usually associated with reduced spatial awareness, but can also lead to reduced speech intelligibility in cases where audiovisual speech cues are available from a talker in an unknown location. Previous work showed that sound-localization cues improved speech perception for bilateral cochlear implantees by helping them face the talker of interest in a multitalker environment. This study evaluated the degree of localization accuracy required to obtain an audiovisual benefit. Normal-hearing adults

used auditory localization cues to orient their gaze toward a video of a target talker. The target-talker location varied randomly among four azimuths spanning 180°. Auditory-localization cues were parametrically degraded with a real-time hearing-loss simulator without affecting auditory speech-recognition performance. Results revealed a maximum audiovisual benefit for speech recognition in noise of 5 dB, diminishing to zero with increasing localization degradation. Head tracking verified the reduction in localization accuracy with increasing degradation. Results were compared to data on traditional broadband-noise localization accuracy. This analysis suggested that listeners with hearing loss, cochlear implants, or wearing hearing protection would need to be able to localize sounds within 35 (~50% of the separation between talkers) to obtain at least some visual benefit from a talker in an unknown location.

1:35

1pPP2. Head movement in human sound localization. Hayato Sato, Masayuki Morimoto (Environ. Acoust. Lab., Graduate School of Eng., Kobe Univ., 1-1 Rokkodai, Nada, Kobe 6578501, Japan, hayato@kobe-u.ac.jp), and Hiroshi Sato (Dept. of Information Technol. and Human Factors, National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Japan)

The aim of the present research is to optimize acoustic guide signals for visually impaired persons. The signal must be localized in the correct direction by the impaired person, since they aid their safe movement. Generally speaking, head movements increase localization accuracy of continuous sound. First, we clarified whether or not the effect of head movement appears in sound localization of intermittent sound in place of continuous one. Localization tests showed that effectiveness of head movements persist even if the sound is intermittent rather than continuous. However, investigations have not yet been carried out to make it clear whether or not spontaneous head movements occur at all during natural sound localization, although the previous studies were premised on the movements. If the movements always occur spontaneously, we could expect greater flexibility on designing acoustic guide signals. Second, we discussed directly whether or not spontaneous head movements occur during natural sound localization. Localization tests showed that spontaneous head movements occur during sound localization; however, they do not always occur.

1:55

1pPP3. Factors influencing auditory localization by moving listeners. Douglas S. Brungart, Tricia Kwiatkowski (Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil), Sarah E. Kruger (National Intrepid Ctr. of Excellence, Bethesda, MD), and Thomas Heil (Walter Reed NMMC, Bethesda, MD)

Previous results have shown that observers who are walking at a constant pace on a treadmill are able to localize continuous sound sources and acquire and identify audiovisual targets more quickly than observers who are standing on a stationary platform, with no apparent loss in accuracy. In order to better understand the conditions where self-motion might enhance localization performance, a follow-on experiment was conducted that measured localization accuracy with sounds of three different durations (250 ms, 1000 ms, and continuous) under four different movement conditions (standing on a stationary platform, standing on a moving platform, walking on a treadmill at a self-paced rate, and walking on a treadmill at a constant rate). Localization accuracy was also measured both standing and walking when spatial cues were degraded by the use of a hearing protection device. As expected, the results suggest that short sounds are localized less accurately than long-duration sounds. However, preliminary results also suggest that walking at a constant pace may provide a greater benefit for localization than self-paced walking. The results are discussed in terms of the possible dynamic localization cues listeners might have access to when they are moving relative to the location of a distant sound source. [Work supported by MRMC Grant W81XWH-12-2-0068; 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 Department of Defense.]

2:15

1pPP4. Auditory space perception during self-motion. Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Wataru Teramoto (Faculty of Letters, Kumamoto Univ., Kumamoto, Japan), Akio Honda (Dept. of Human Sci., Yamanashi-Eiwa College, Kofu, Japan), Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Jiro Gyoba (Graduate School of Arts and Letters, Tohoku Univ., Sendai, Japan)

Spatial information inputted to the auditory periphery dramatically changes with a listener's body movements relative to the sound source. Nevertheless, listeners can perceive a stable auditory environment and react appropriately to the sound source. This suggests that the spatial information is reinterpreted in the brain by being integrated with information regarding movement, while it is well known that the motion itself sometimes negatively affects sound localization acuity. We have focused on how people perceive auditory space during movement. A linear-motor-driven chair and a spinning chair were installed in an anechoic room to provide linear and rotatory motion, respectively. The results revealed that the perceived sound position during linear self-motion was displaced compared with the physical sound position. This displacement was observed regardless of the self-motion direction and the inputted sensory information which induces self-motion. Such degradation of sound localization acuity was also observed when listeners were rotated; the detection thresholds for the listener's subjective front were elevated during the motion. Here, we introduce the details of our study and discuss the characteristics of auditory space perception during self-motion. [Work supported by JSPS KAKENHI Grant (26280067).]

2:35

1pPP5. Relative head angle and eye position drive systematic distortions in the perception of auditory space. W. Owen Brimijoin and Graham Naylor (Inst. of Hearing Res., Medical Res. Council/Chief Scientist Office, MRC/CSO Inst. of Hearing Res., Fl. 3 New Lister Bldg., 16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, owen.brimijoin@nottingham.ac.uk)

The minimum audible movement angle (MAMA) increases as a function of azimuth. This study examined a potential consequence of this: if this change in resolution across azimuth is not perceptually compensated for it would suggest that a sound rotating at a constant angular velocity around the head would not appear to do so but would appear to move faster at the front than at the side. We therefore examined whether the azimuth of two signals affects their point of subjective similarity for motion. We found that equal movement from two different azimuths is not judged as being equivalent: signals centered at 90° must move at least twice as much as ones centered at 0° to be judged as moving the same amount. Signals centered at 45° were also perceived in a proportionally distorted manner.

Furthermore, in some listeners, this expansion/contraction of auditory space appears to shift systematically with eye position. These interactions suggest that the MAMA may not be simply a measure of acuity, but also reflects an underlying head- and eye-driven nonlinear perceptual organization of space, one we term the "equivalent arc," in which an absolute number of degrees does not correspond to a consistent perceptual unit across azimuth.

2:55–3:15 Break

3:15

1pPP6. Effects of source- and head-motion on auditory perceptual organization. Hirohito M. Kondo (Human Information Sci. Lab., NTT Commun. Sci. Labs., 3-1 Morinosato Wakamiya, Atsugi, Kanagawa 243-0198, Japan, kondo.hirohito@lab.ntt.co.jp), Daniel Pressnitzer (CNRS UMR 8248 & ENS, Paris, France), Iwaki Toshima, and Makio Kashino (Human Information Sci. Lab., NTT Commun. Sci. Labs., Atsugi, Japan)

The perceptual organization of auditory scenes needs to parse the incoming flow of acoustic inputs into perceptual streams. It is likely that cues from several modalities are pooled for auditory scene analysis, including sensory-motor cues related to the active exploration of the scene. We show effects of source- and head-motion on auditory streaming. In a streaming paradigm, listeners hear a sequence of repeating tone triplets and indicate their perception of one or two subjective sources called streams. We used a robotic telepresence system, Telehead, to disentangle the effects of head motion: changes in acoustic cues at the ear, subjective location cues, and motor cues. We found that head motion induced perceptual reorganization even when the auditory scene had not changed. We further analyzed the data to probe the time course of sensory-motor integration. Motor cues impacted perceptual organization earlier and for a shorter time than acoustic or subjective location cues, with successive positive and negative contributions to streaming. An additional experiment showed that arm or leg movements did not have an impact on scene analysis. Our results suggest a loose temporal coupling between the different mechanisms involved.

3:35

1pPP7. Multi-channel loudspeaker reproduction and virtual acoustic environments in the context of hearing aid evaluation. Giso Grimm, Stephan Ewert, and Volker Hohmann (Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Germany, Marie-Curie-Str. 2, Oldenburg 26129, Germany, g.grimm@uni-oldenburg.de)

Virtual acoustic environments reproduced via multiple loudspeakers are increasingly used for evaluating hearing aids in complex acoustic conditions. To further establish the feasibility of this approach, this study investigated the interaction between spatial resolution of different reproduction methods and technical and perceptual hearing aid performance measures using computer simulations. Three spatial audio reproduction methods—discrete speakers, vector base amplitude panning and higher order ambisonics—were compared in regular circular loudspeaker arrays with 4 to 72 channels. The influence of reproduction method and array size on performance measures of multi-microphone hearing aid algorithm classes with spatially distributed microphones and a representative single channel noise-reduction algorithm was analyzed. In addition to the analysis of reproduction methods, algorithm performance was tested in a number of different virtual acoustic environments in order to assess the underlying factors of decreased hearing aid performance in complex environments. The results confirm previous findings that spatial complexity has a major impact on hearing aid benefit and demonstrate the potential of virtual acoustic environments for hearing aid evaluation.

3:55

1pPP8. Discrimination of virtual sound fields different in spatial aliasing. Yukio Iwaya (Faculty of Eng., Tohoku Gakuin Univ., 1-13-1 Chuo, Tagajo, Miyagi 985-8537, Japan, yukio@iwaya-lab.org), Makoto Otani (Kyoto Univ., Kyoto, Kyoto, Japan), and Takao Tsuchiya (Doshisha Univ., KyoTanabe, Kyoto, Japan)

There are theoretical techniques for sound field reproduction with a loudspeaker array. However, physical precision of synthesized sound field is limited in frequency domain. When the distance among loudspeakers is below a half of wavelength, spatial aliasing will occur. The ideal distance among loudspeakers was less than 8.5 mm@20 kHz, and it is impossible to arrange loudspeakers. Therefore, we carried out experiments to know an upper-limit frequency condition, which can give subjective experience identical to an ideal sound field. We assumed a virtual spherical boundary around a listener. Numerous virtual loudspeakers were set on the boundary to simulate a WFS system. Room impulse responses and head-related impulse responses were calculated by computer simulation. Then, binaural impulse responses were synthesized as combinations of RIRs and HRIRs. We picked up the boundary points so that upper-limit frequency without the spatial aliasing of BRIRs was systematically controlled. Listening test was conducted to investigate discrimination of sound fields different in spatial aliasing. We found that the sound field of upper-limit frequency 4 kHz cannot be discriminated from that of 14 kHz. Therefore, sound field reproduced without spatial aliasing up to 4 kHz could give same sound experience identical to the ideal sound field.

4:15

1pPP9. Efficient physics based simulation of spatial audio for virtual and augmented reality. Ramani Duraiswami and Dmitry N. Zotkin (Comput. Sci. & UMIACS, Univ. of Maryland, A.V. WIlliams Bldg., #115, College Park, MD 20742, ramani@umiacs.umd.edu)

Many applications in spatial hearing are being enabled by the virtual and augmented reality revolution. Because of these there has been a tremendous improvement in the quality of head mounted displays, and the availability of tracking (3 DOF and 6 DOF) in an integrated environment with headphones suitable for binaural rendering. While these devices were initially focused on creating visual representations of the 3D world (in the case of VR) or a visual representation of artificial 3D objects overlaid on the real world (in the case of augmented reality), it was soon realized that unless the audio provided a degree of spatial realism that complemented the visual and was consistent with it, the sense of immersion crucial to maintaining the illusion was lost. In this talk, I will describe the engine for creating virtual simulations in real time that incorporates head motion, environmental reverberation, room materials, as well as generic or individual HRTFs. Trade-offs that allow the engine to work on architectures ranging from single core CPUs, to mobile devices to high end

multicore devices will be discussed. In contrast to these "object-based" representations of spatial auditory objects, there has also been a revival in approaches based on expanding the sound-field in low order ambisonics representations, and their subsequent binaural rendering. After providing a description of the optimizations made in our engine, we discuss future work.

4:35

1pPP10. Possibility of binaural production by means of bone conduction actuators. Xiuyuan Qin and Tsuyoshi Usagawa (CSEE, Kumamoto Univ., Kumamoto Prefecture, Japan, Kumamoto City 860-0862, Japan, qnini@126.com)

A pair of bone conduction actuators is used for music reproduction as replacement of headphone. Because of strong inter-aural coupling, it is though the stereophonic reproduction is not easy to realize by bone conduction actuators. Haim Sohmera, *et al*. 2000, reported that "there is no need to vibrate bone in order to obtain 'bone conduction' response." They have shown a fluid pathway from bone to ear. The coupling between left and right side stimuli by bone conduction actuators set at head-of-mandible or mastoid is occurred not only due to bone but also fluid. On the other hand, when we are listening the music by bone conduction actuators, we can enjoy the some extent of special image, at least some extend of lateralization. The authors are examined the possibility of binaural production by means of inter-aural time and intensity trading using a pair of bone conduction actuators. Results of psycho-acoustical experiment indicate that the trading phenomenon between time and intensity is observed around 1000 Hz even if the results are not stable.

MONDAY AFTERNOON, 28 NOVEMBER 2016 SEA PEARL, 1:00 P.M. TO 5:30 P.M.

Session 1pSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration II

Benjamin Shafer, Chair
Technical Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

Contributed Papers

1:00

1pSA1. Strong fluid-structure coupling in a two-dimensional waveguide bounded by elastic plates. Jerry H. Ginsberg (School of Mech. Eng., Georgia Inst. of Technol., 5661 Woodsong Dr., Dunwoody, GA 30338-2854, j.h.ginsberg@comcast.net)

The typical model for compliant walls in a waveguide take them to be locally reacting. The corresponding boundary conditions for the transverse mode functions are Robin-type, which leads to an analysis that is a straightforward modification of the conventional analyses for rigid and pressure-release walls. The present investigation takes up the case where the walls are elastic plates, and the fluid loading is heavy. The consequence is strong coupling of the fluid and structural responses. The fact that the walls are not rigid obviates the possibility of a planar mode, so it is necessary to consider the transverse dependence of modes at any frequency. The analytically derived characteristic equation for the coupled response vividly displays the interaction of elastodynamics and acoustics. The eigenvalues are the transverse wavenumbers of the various modes at a specified frequency. They can be obtained through numerical methods, but implementing such a procedure to obtain dispersion curves of transverse wavenumber vs. frequency for each mode is cumbersome. An alternative procedure based on contours of the characteristic equation is exceptionally easy to implement, and extensible to other problems. One of the results that follow from the analysis is the observation that there is a minimum frequency below which all modes evanesce.

1:15

1pSA2. Identification the vibroacoustic properties of sandwich-composite panels from wavenumber measurements. Noureddine Atalla, Raef Cherif (GAUS Mech. Eng., Univ. of Sherbrooke, UdeS, Mech. Eng., 2500 Blvd de l'Universite, Sherbrooke, QC J1K0A2, Canada, noureddine.atalla@usherbrooke.ca), and Gary Burns (Cabin Interiors & Structures, Zodiac Aerosp., Huntington Beach, CA)

This paper investigates the identification of the vibroacoustic properties of sandwich composite panels from structural wavenumber measurements over a large frequency band. Several indicators are investigated including the modal density, damping loss factor, radiation efficiency, and sound transmission loss. The accuracy of the identified indicators is studied by comparison with several direct and indirect measurements and analytical predictions. Results show that all identified indicators are in good agreement with measurements and theory for the studied constructions.

1:30

1pSA3. Sound radiation from unbaffled vibrating plates, using an edge-diffraction based approach. Sara R. Martín Román, U. Peter Svensson (Acoust. Res. Ctr., Dept. of Electronics and TeleCommun., Norwegian Univ. of Sci. and Technol., Trondheim, Norway, O.S. Bragstads plass 2a, Trondheim 7034, Norway, sara.martin@ntnu.no), Lauri Savioja (Dept. of Comput. Sci., Aalto Univ., Espoo, Finland), and Julius O. Smith (Ctr. of Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., Stanford, CA)

The sound radiation from unbaffled vibrating plates is a classical problem that is often solved numerically by coupling a vibration model to the surrounding air with the boundary element method, or the finite differences in the time domain (FDTD). Here, the surrounding air is instead represented with Green's functions computed with the edge source integral equation (ESIE) [Asheim and Svensson, JASA **133**, pp. 3681-3691, 2013]. Expressions are derived for the analytical edge directivity functions in the ESIE for the special case of a thin flat plate with sources on both sides of the plate, of opposite polarity. Frequency-domain as well as time-domain results are presented, for specified vibration patterns, including an unbaffled piston. The coupling of the Green's functions to an FDTD model of the plate vibration is outlined. The singularity issues that appear with the ESIE for certain radiation directions are handled using a recent hybrid approach [Martín *et al*., J. Acoust. Soc. Am. **139**, 2202 (2016)] where the ESIE is used to calculate the sound pressure on the surface of the radiator, followed by the same propagation stage as in the boundary element method. This approach offers a computationally efficient method, and the time-domain version makes auralization straightforward.

1:45

1pSA4. Wave dispersion in randomly layered materials. Mieczyslaw Cieszko and Michal Pakula (Inst. of Mech. and Comput. Sci., Kazimierz Wielki Univ. in Bydgoszcz, ul. Chodkiewicza 30, ul. Kopernika 1, Bydgoszcz 85-064, Poland, michalp@ukw.edu.pl)

Wave scattering in materials composed of two kinds of alternating layers with different elastic properties and randomly distributed thicknesses has been modeled. The general form of the dispersion equation is derived for the unbounded layered medium. It defines two basic macroscopic characteristics of the scattered wave: phase velocity and attenuation, which are explicit functions of wave frequency and microscopic parameters of the system: acoustic properties of the layers and stochastic characteristics of their thickness distributions. The analytical expressions are derived for three special cases: for long waves; for a periodic medium composed of layers with constant thicknesses and for random medium with uniform distribution of layer thicknesses. Special attention is paid to the analysis of the frequency dependence of the wave parameters. It was shown that the predictions of the model for long waves and for periodic medium are compatible with the results obtained in the literature. Moreover, comparison of theoretical results for frequency dependent wave parameters with numerical simulations of pulse transmission through the slab of the randomly layered medium shows good qualitative and quantitative agreement in wide frequency range.

2:00

1pSA5. Vibration analysis of high-speed rotating conical shell with arbitrary boundary conditions. Jinzhao Wang (College of Power and Energy Eng., Harbin Eng. Univ., No.145 Nantong St. Nangang District, Harbin 150001, China, wjinzhao08@163.com), Yipeng Cao (College of Power and Energy Eng., Harbin Eng. Univ., Harbin City, China), and Guang Lin (School of Mech. Eng., Dept. of Mathematics, Purdue Univ., West Lafayette, IN)

Based on the energy method, an analytical model for the vibration characteristics of high speed rotating conical shell with arbitrary boundary is constructed. Effects of centrifugal and Coriolis accelerations as well as initial hoop tension due to rotating are all taken into account. A set of translational and rotational artificial springs to simulate the elastic constraints imposed on the conical shell edges. Under the framework, each of the displacement components of the conical shell is invariantly described by a modified Fourier series, which is composed of a standard Fourier series and closed-form auxiliary functions. All the expansion coefficients are determined by using the familiar Rayleigh-Ritz procedure as the generalized coordinates. To validate the present analysis, a series of comparisons between the present results and those available in the open literature are carried out and very good agreements are obtained. On this basis, some numerical results are given to illustrate the effects of rotating speed and geometric parameters on the frequencies of rotating conical shell. Further, the influence of the boundary springs'stiffness are also discussed.

2:15

1pSA6. Vibro-acoustics of cylindrical shells with distributed damping liners. Rajendra Singh (Mech. & Aerosp. Eng., The Ohio State Univ., 201 West 19th Ave., Columbus, OH 43210, singh.3@osu.edu)

Distributed passive damping treatments (such as cardboard liners, and foam or visco-elastic materials) are routinely placed inside the automotive drive shafts to reduce the airborne and structureborne noise over the applicable mid-frequency range. To better understand the underlying mechanism(s), experimental and analytical studies are undertaken. First, a laboratory experiment is designed to measure vibro-acoustic response functions of a thin cylindrical shell with different cardboard liner thicknesses. Second, the properties of a cardboard liner material are determined by conducting an experiment on an elastic plate treated with a cardboard liner. Third, an analytical framework for shell vibration is developed and the relative sliding motion between substructure and liner is explicitly described. The proposed model agrees well with the experiment in terms of natural frequency shifts, modal loss factors, and modal insertion losses. Several damping mechanisms such as material damping, dry friction, and interfacial viscous friction are examined. Finally, various aspects of the problem including unresolved issues will be briefly discussed.

2:30

1pSA7. Quantification of disorder sensitivity of a subordinate oscillator array. John A. Sterling (US Navy, 9500 Macarthur Blvd., Bethesda, MD 20817, jsterling@gmail.com), Joseph Vignola (Mech. Eng., Catholic Univ. of America, College Park, MD), Aldo Glean (Mech. Eng., Catholic Univ. of America, Boston, MA), and Teresa Ryan (Mech. Eng., East Carolina Univ., Greenville, NC)

Recent research has shown that arrays of small dynamic elements attached to a larger, primary structure can be tuned to significantly alter the time and frequency response of the system. Often referred to as "fuzzy structures," such subordinate oscillator arrays have applications including damping, radio frequency filtering, energy harvesting, micro electromechanical systems, and chemical vapor sensing. This work shows that an attached array adds damping and demonstrates a distinct rejection band in the spectral response.The subordinate system here is an array of cantilever beam attachments with a range of isolated natural frequencies that surround a target band. Small errors in the distributions of masses and stiffnesses of the attachments can have a significant degrading effect on desired response. Error on the order of 0.1% can be significant. This paper discusses experimental verification of this phenomena. In the experiment, small amounts of mass were added to the tip of both individual and combinations of the cantilevers to deliberately deviate from the design. The resultant responses were recorded. A direct relationship between perturbed mass and frequency distributions and response deviation will be shown.

2:45

1pSA8. Dynamic response of a beam supported with variable gear to base excitations. Hyukju Ham, Deokman Kim, and Junhong Park (Mech. Eng., Hanyang Univ., 222, Wangsimni-ro, Seongdong-gu, Eng. ceneter, 306, Seoul ASI|KR|KS013|SEOUL, South Korea, shadowpl@nate.com)

The beam supported with gear having nonlinearly varying stiffness characteristics is analyzed to study vibration transfer characteristics. The gear attached at the end of the beam was considered as rotational spring and damper having angular dependent variation. The dynamic characteristics of the gear were determined by the contact point between the gear teeth. The displacement-dependent properties of the gear had an effect on vibration transmissibility of the beam system. The computational simulation based on the spectral element method (SEM) was used with several variables (the dynamic characteristics of the gear, the gear design, and excitation level). The validity was verified by comparison with the measured results. The vibration of the beam was strongly influenced by the gear-pair. These results can be utilized to design the optimal gear-pair system for minimizing undesirable vibration.

3:00–3:15 Break

3:15

1pSA9. Axial force measurement for U-bolt using wave propagation speed monitoring. Gyungmin Toh, Dongki Min, Jaehong Lee, and Junhong Park (Mech. Eng., Hanyang Univ., 222. Wangsimni-ro, Seongdong-gu, Seoul 04763, South Korea, avlrudals@gmail.com)

It is important to inspect clamped parts of a produced vehicle in the manufacturing process non-destructively. In this study, a measuring device of axial force is suggested by utilizing relations between wave propagation characteristics of a beam and a clamping force. Experiments were carried out to determine characteristics of vibration signals for the structure designed for inspection. Neodymium magnet was utilized as the components of the device to attach to the joint without any damage to the products. A rectangular parallelepiped magnet was used to detect the change in the wave propagation speeds in the structure. The change in the wave propagation through the bolt was measured and used in estimation of the axial force. The correlation coefficient was calculated to quantify the change in the wave propagation characteristics. The coefficient was utilized to determine the axial force of the bolted joint by the experiment. The proposed method was applied to actual U-bolts to validate the methodology.

3:30

1pSA10. Thermoacoustic power sensors: Principles and prediction. Christopher M. Dumm, Jeffrey S. Vipperman (Mech. Eng., Univ. of Pittsburgh, 504 Benedum Hall, 3700 O'Hara St., Pittsburgh, PA 15261, jsv@pitt.edu), Jorge V. Carvajal, Melissa M. Walter, Luke Czerniak, Amy S. Ruane, Paolo Ferroni, and Michael D. Heibel (Westinghouse Electric Co., Cranberry Township, PA)

Thermoacoustic Power Sensor (TAPS) technology can be used to wirelessly measure the temperature and radiation flux conditions in a nuclear reactor core. A TAPS is a self-powered, standing-wave thermoacoustic engine, enclosed in a cylinder, and placed in the reactor core (for example, inside instrumentation tubes). TAPS utilize heat from a radiation-powered heater and cooling from the reactor core coolant to generate acoustic waves; these waves have amplitude proportional to the local radiation flux and frequency proportional to the local coolant temperature. The acoustic waves propagate physically into the reactor coolant and structure, and are detected with receivers (e.g. accelerometers) placed on the outside of the reactor vessel. TAPS signals are interpreted (and the reactor state conditions measured) by comparison of the received signals to a reference generated by predictive modeling. Since TAPS are wireless and self-powered, they offer advantages in safety (e.g., by reducing the required number of vessel penetrations) and in maintenance reduction. Since TAPS contain no internal electronics, they are particularly well-suited to high-temperature, fast radiation environments for which no in-core instrumentation currently exists, such as in sodium-cooled fast reactors (SFRs). A methodology for predictive modeling of TAPS signals is discussed and evaluated experimentally.

3:45

1pSA11. Estimating mechanical properties of ice on Europa: Impact simulation and sensitivity analysis. Joshua Capozella, Edward Pfaeffle, Martin Ferreira Fernandes, Masataka Okutsu, Joseph Vignola, and Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., N.E., Washington, DC 20064, 78pfaeffle@cua.edu)

The primary objective of NASA's Europa Multiple Flyby Mission to Jupiter's moon Europa is to determine the thickness of the ice covering the global subsurface ocean. If the ice proves too thick for the ice penetrating radar, however, an indirect method must be used to estimate the depth of the ice-water boundary. Unfortunately, the properties of Europan ice are not known precisely, with Young's modulus having two orders of magnitude of uncertainty. It has been suggested that seismic response produced by an artificial impact could help estimating the mechanical properties of this ice. By optically measuring the induced surface motion with enough resolution, the dispersion curves can be determined. The associated elastic wave speeds are then used to estimate the moduli and density of the ice. The seismic response of the ice is modeled using a Finite Element Method solver. This presentation describes the sensitivity of the estimated mechanical properties to both temporal and spatial resolutions of the optical measurement. The sensitivities of the elastic constants to the temporal and spatial resolutions are analyzed by iterating the simulations at increasing time steps and finer meshes.

4:00

1pSA12. Study on the exactness of alternative indirect force estimation methods, power dissipation calculation by power or energy flow, and transfer path analysis for structure-borne sound. Akira Inoue (R&D APL, Hitachi America, Ltd., 34500 Grand River Ave, R&D-APL, Farmington Hills, MI 48335, akira.inoue@hitachi-automotive.us) and Yosuke Tanabe (Hitachi, Ltd., Hitachinaka, Ibaraki, Japan)

In our presentation, it is first clarified that the matrix inverse method which gives interfacial forces using X/F-type frequency response functions among parallel interfacial points is an exact method derived from Newton's equation of motion. The alternative method using F/X-type frequency response functions with blocked boundaries is next explained as another exact method, though implementation of the blocked boundary may be difficult for real-life systems. Next, as an application of interfacial force, it is shown that the real part of power flow and the imaginary part of energy flow correspond to mechanical power dissipation. When all the power or energy flows into a system are counted, it gives the exact power dissipation of the system. As another application, the exactness of transfer path analysis for structure-borne sound is examined with brief discussion of velocity potential. Computational demonstrations and validations are given for the derived formulas, using a simple finite element model. All the systems are supposed to be linear time-invariant, discretized and in a steady state.

4:15

1pSA13. Passive defect detection in plate from nonlinear conversion of low-frequency vibrational noise. Emmanuel Moulin, Lynda Chehami, Jamal Assaad (IEMN, UMR CNRS 8520, Univ. of Valenciennes, Le Mont Houy, Valenciennes F-59313, France, emmanuel.moulin@univ-valenciennes.fr), Julien de Rosny, Claire Prada (Institut Langevin UMR CNRS 7587, ESPCI ParisTech, Paris, France), Eric Chatelet, Giovanna Lacerra (LaMCoS UMR CNRS 5259, INSA, Univ. of Lyon, Lyon, France), Konstantinos Gryllias (Faculty of Eng., Dept. of Mech. Eng., KU Leuven, Leuven, Belgium), and Francesco Massi (Univ. of Rome La Sapienza, Dept. of Mech. and Aerosp. Eng., Rome, Italy)

It is known that, under the assumption of diffuse noise, the cross-correlation of acoustic signals recorded at two points of a medium allows to passively estimate the impulse response between these points. This principle, associated with coherent array processing, has been successfully applied to defect detection and localization in reverberant plates subject either to distributed noise sources or friction noise. In order to extend the applicability of this principle even in the absence of an adequate ambient noise, we have introduced the concept of secondary noise sources based on the conversion of low-frequency modal vibrations into high-frequency noise by exploiting frictional contact nonlinearities. The device consists of a mass-spring resonator coupled to a flexible beam by a rough frictional interface. The extremity of the beam, attached to the surface of a plate, excites efficiently flexural waves in the plate in the ultrasound range when the primary resonator vibrates around its natural frequency. We have shown that a set of such devices distributed on a plate excited by a shaker at low frequencies is able to produce high-frequency noise suitable for passive defect detection and localization based on noise correlation.

4:30

1pSA14. The control of resonance curve using the shape modulation of the scattering region in elastic waveguide. Yoshimasa Naito, Hatuhiro Kato, and Takaaki Ishii (Human Oriented Eng., Univ. of Ymanashi, Takeda, Kofu, Ymanashi 400-8511, Japan, g15mh012@yamanashi.ac.jp)

This study focuses on the control of Fano resonance in the elastic plate, which is initially developed in atomic scattering and has been seldom to discuss. Fano resonance is a phenomena involved by the localized/quasi-localized wave, in which the resonance curve becomes asymmetric due to the phase difference between the incident wave and the localized wave. The typical resonance curve available in the system is the frequency dependence of the reflectance under the scattering problem. We will demonstrate that the controllability of the resonance curve using various shapes of the

scattering region under the simulation and experiment. The experiment system is an elastic waveguide made of aluminium plate and the scattering region is formed by modulating the plate thickness. The localized wave is induced by incident wave in the thinned scattering region and causes the wave tail that penetrates into the lead region of waveguide. The controllability can be realized by the overlap of the wave tail and the incident wave. For analyzing the experiment data, we also develop the method to extract S-parameters. The controllability was already confirmed by the preliminary experiment. Now, we are preparing verify various shapes of the scattering region for detail experiments.

4:45

1pSA15. Radiation of rail noise regarding surface impedance of ground by using wavenumber domain finite and boundary element. Jungsoo Ryue (School of Naval Architecture and Ocean Eng., Univ. of Ulsan, Rm 309, Bldg. 41, Ulsan 44610, South Korea, jsryue@ulsan.ac.kr) and Seungho Jang (Korea RailRd. Res. Inst., Uiwang, South Korea)

An important source of noise from railways is rolling noise caused by wheel and rail vibrations induced by acoustic roughness at the wheel-rail contact. In conventional approaches to predicting rail noise, the rail is regarded as placed in a free space so that the reflection from the ground is not included. However, in order to predict rail noise close to the rail, the effect of the ground should be contained in the analysis. In this study the rail noise reflected from the ground is investigated using the wavenumber domain finite element and boundary element methods. First, two rail models, one using rail attached to the rigid ground and one using rail located above rigid ground, are considered and examined to determine the rigid ground effect in terms of the radiation efficiency. From this analysis, it was found that the two models give considerably different results, so that the distance between the rail and the ground is an important factor. Second, an impedance condition was set for the ground and the effect of the ground impedance on the rail noise was evaluated for the two rail models.

5:00

1pSA16. Modeling of multi-jointed structures by complex stiffness based on vibration transfer characteristics. Sangmok Park, Yunsang Kwak, and Junhong Park (Hanyang Univ., 222, Wangsimni-ro, Seongdong-gu, Eng. Ctr, 306, Seoul 04763, South Korea, tkdahr619@hanyang.ac.kr)

Complex stiffness representation is proposed to analyze the dynamic characteristics of jointed structures. Measured results for single-jointed specimens were compared with predictions to show the influence of connecting stiffness. The proposed dynamic stiffness representation showed accurate vibration transfer characteristics of the single-jointed structures. For multi-jointed structures, the developed prediction model was extended based on the results of single-jointed structures. For experimentations, multi-jointed specimens having different binding strength and various defected parts were produced. Vibration transfer characteristics were measured. In order to fully represent dynamic characteristics of multi-jointed specimens, the spectral element method was used for theoretical modeling. Complex stiffness allowed modeling of coupling effect by multiple joints. In aspect of wave propagation characteristics, the proposed model was verified by comparison with measured vibrations of the multi-jointed specimens. Locations of defects in specimens were theoretically predicted by the verified model. Binding strength of the multi-jointed structures was quantitatively evaluated using the complex stiffness.

5:15

1pSA17. You really can hear the corn grow! Acoustic emissions in the growth and breakage of maize. Douglas Cook (Eng., New York Univ. Abu Dhabi, PO Box 903, New York, NY 10276, prof.laji@gmail.com)

Corn (or maize) is the leading grain crop in the United States. More than 350 million metric tons are harvested annually, generating twice as much revenue as any other crop. But a lack of understanding of corn stalk mechanics now hinders further improvement of corn production. The most promising new varieties of corn produce high yields, but are often susceptible to wind-induced failure of the stalk. Failure of this kind is prevalent at two distinct periods: in mid-summer during rapid growth, and after physical maturity, but before harvest. This study utilized acoustic measurements to collect new information about corn growth and failure. Measurements during the growth period were conducted in July 2016 at fields in Nebraska. Late-season stalk experiments involved stalks harvested from South Africa. Flexible, shielded piezo film sensors were used in all measurements (SDT1-028K, Measurement Specialties, Hampton, VA). Acoustic emissions were found to occur continuously during corn stalk growth, though typically at levels far below those audible to the human ear. The authors hypothesize that special conditions would be required to render these emissions audible to humans. Acoustic emissions provided valuable information which used to determine modes of stalk failure in late season lodging.

Session 1pSC

Speech Communication and Signal Processing in Acoustics: Speech Robotics

Masakaki Honda, Cochair

Philip Rubin, Cochair
Haskins Laboratories, 300 George Street, Suite 900, New Haven, CT 06511

Chair's Introduction—1:30

Invited Papers

1:35

1pSC1. Talking heads: communication, embodiment, and interaction. Philip Rubin (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, rubin@haskins.yale.edu)

The desire to create robotic speech, including talking human heads, stems back at least several hundred years. This quest continues and presently combines approaches that are computational, cognitive, biological, mechanical, and acoustic, cutting across a wide variety of domains and interests. This presentation provides some highlights of the broad, international effort that includes speech robotics and social robotics, the historical antecedents of this effort, and considers aspects of embodiment and interaction between speakers. By talking heads, we mean models that produce audio-visual speech, either physically or on a computer screen, through the control of parameters derived from physiological, behavioral, or other sources. In addition to their use as scientific tools for exploring speech perception and production, and for simulating physiological systems under normal and adverse conditions, these models can be key components in the synthesis and manipulation of virtual personalities. Avatars in cyberspace environments, facially-realistic animated characters, and virtual personalities on our computer, videogame, video and movie screens, and mobile devices are becoming increasingly common. The realism of these simulations will depend, in part, on the incorporation of the relevant knowledge (including physiological, behavioral, and social) in the models of the wide varieties of cyber-entities that will inhabit our future.

1:55

1pSC2. A physical figure model of lips for speech production. Makoto J. Hirayama (Faculty of Information Sci. and Technol., Osaka Inst. of Technol., 1-79-1 Kitayama, Hirakata 573-0196, Japan, makoto.hirayama@oit.ac.jp)

A physical figure model of lips were made. In previous studies, tongue models consisting of intrinsic and extrinsic tongue muscles were made using viscoelastic material of urethane rubber gel. At this time, as other important speech articulators, upper and lower lips are modeled using urethane rubber gel. Also, a movable jaw model is made from plastic cray. The lips model is placed on the jaw model. As the lips model is made from viscoelastic material similar to human skin, reshaping and moving lips are possible by pulling or moving lips and jaw by hand to emulate speech articulation of lips and jaw. Orofacial muscles such as orbicularis oris, levator labii superioris, levator anguli oris, depressor anguli oris, depressor labii inferioris, and mentalis are considered for lip motions with the figure model. The proposed model is useful for speech science education or a future speaking robot development based on realistic speech production mechanisms.

2:15

1pSC3. Learning from visual speech. Sarah Taylor (Univ. of East Anglia, School of Computing Sci., Norwich, United Kingdom, sarah.hilder@gmail.com), Taehwan Kim, Yisong Yue (Caltech, Pasadena, CA), Ben Milner (Univ. of East Anglia, Norwich, United Kingdom), and Iain Matthews (Disney Res., Pittsburgh, PA)

Realistic animation of faces has proven to be challenging due to our sensitivity in perceiving visual articulation errors. We study the problem of learning to automatically generate natural speech-related facial motion from audio speech which can be used to drive both CG and robotic talking heads with low bandwidth requirements and low latency. A many-to-one mapping from acoustic phones to lip shapes (i.e., static "visemes") is a poor approximation to the complex, context-dependent relationship visual speech truly has with acoustic speech production. We introduced "dynamic visemes" as data-derived visual-only speech units associated with distributions of phone strings and demonstrated they capture context and co-articulation. Further improvement in predicting visual speech can be achieved using an end-to-end deep learning approach. We train a sliding window deep neural network that learns a mapping from a window of phone labels or acoustic features to a window of visual features. This approach removes the requirement for visual units and can very accurately predict visual speech from phone labels or the audio signal directly. We demonstrate the effectiveness of these approaches on CG face models and discuss their application to animatronic heads.

2:35

1pSC4. A robot-based enjoyable conversation system. Tetsunori Kobayashi (Dept. of Comput. Sci., Waseda Univ., 27 Waseda-machi, Shinjuku-ku, Tokyo 162-0042, Japan, koba@waseda.jp)

A robot-based enjoyable conversation system, SCHEMA, is introduced. The first main function of this robot is acting as a facilitator who keeps the balance of engagement density in a group conversation. In a group conversation, sometimes only a particular pair of participants talk each other, and the other particular side participants lose the opportunities to join in the conversation. In terms of total balance of enjoyment of the conversation, this kind of engagement-density-imbalance should be corrected. The proposed robot can find the participants who feel difficulty in joining in a conversation, and give him a chance to join in without destroying existing conversation. The second characteristic function of this robot is generating subjective opinion. In terms of functionality of conversations, it is enough to reply directly to the question that was explicitly asked. However, in our daily conversations, more informative responses are usually employed in order to hold enjoyable conversations. These responses are usually produced as forms of one's additional opinions, which usually contain their original viewpoints as well as novel means of expression, rather than simple and common responses. The proposed robot can select some messages from WEB review site, arrange them, and generate a message as if it is the opinion of one's own. In this talk, we show these functions successfully utilized to achieve enjoyable conversation.

2:55

1pSC5. Spoken and non-verbal interaction experiments with a social robot. Jonas Beskow (Speech Music and Hearing, KTH Royal Inst. of Technol., Lindstedtsvägen 24, 5tr, Stockholm 11428, Sweden, beskow@kth.se)

During recent years, we have witnessed the start of a revolution in personal robotics. Once associated with highly specialized manufacturing tasks, robots are rapidly starting to become part of our everyday lives. The potential of these systems is far-reaching; from co-worker robots that operate and collaborate with humans side-by-side to robotic tutors in schools that interact with humans in a shared environment. All of these scenarios require systems that are able to act and react in a *social way*. Evidence suggests that robots should leverage channels of communication that humans understand—despite differences in physical form and capabilities. We have developed Furhat—a social robot that is able to convey several important aspects of human face-to-face interaction such as visual speech, facial expression, and eye gaze by means of facial animation that is retro-projected on a physical mask. In this presentation, we cover a series of experiments attempting to quantize the effect of our social robot and how it compares to other interaction modalities. It is shown that a number of functions ranging from low-level audio-visual speech perception to vocabulary learning improve when compared to unimodal (e.g., audio-only) settings or 2D virtual avatars.

3:15

1pSC6. Infant–caregiver interaction is essential for early vocal development. Minoru Asada (Adaptive Machine Systems, Osaka Univ., 2-1 Yamadaoka, Suite, Osaka 565-0871, Japan, asada@ams.eng.osaka-u.ac.jp)

Vocal interactions between an infant and a caregiver facilitates early vocal development, and one of the issues is how an infant learns to vocalize the caregiver's native language. Explanation of this ability of infant is attempted by many theories which advocate imitation based on acoustic matching. However, the infant's and caregiver's speech are quite different in their acoustic qualities. Therefore, it cannot be fully explained by imitation. In this talk, I assume that the interaction itself may have an important role to play, but the mechanism is still unclear. I review studies addressing this topic based on explicit interaction mechanisms using computer simulations and/or real vocal robots. Modeling approaches are classified into four categories: (a) motor control ability develops through self-monitoring of vocalizations, (b) statistical estimation of caregiver's vowel categories from caregiver's vocalizations, (c) self-organization of shared vowels through imitative interaction, and (d) whole dynamics of interactions. The relationships between them are analyzed after a brief review of the early development of an infant's speech perception and articulation based on observational studies in developmental psychology and a few neuroscientific imaging studies. Finally, future issues related to real infant-caregiver vocal interaction are outlined.

3:35

1pSC7. Coordination as a simplifying factor in human-machine communicative interaction. Eric Vatikiotis-Bateson (Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6N2W4, Canada, evb@mail.ubc.ca)

Spoken communication has traditionally been treated as a problem of sending and receiving signals containing elements whose sequential organization specifies meaningful content that is the same for both sender and receiver. Unfortunately, successful communication depends on a host of contextualizing factors that are either not present or impossible to identify in short signal streams (by either humans or machines). This presentation focuses on the crucial role of one such factor, the necessary coordination between perceiver and producer, to suggest that human communicative interaction can be conceived more simply than it has been and thereby improve the likelihood of successful human-machine interaction. For example, humans depend on physiological and cognitive coordination for successful interaction, particularly in signal parsing, alignment, and error correction. However, coordination is loosely constrained under most conditions and can be achieved in human-machine interaction without imbuing machines with human attributes; for example, shared attention. In many other cases, machines outperform humans; for example, in processing multimodal speech signals under adverse acoustic conditions to disambiguate the labial viseme, /p,b,m/. In sum, these and other factors provide texture and coherence, rather than daunting complexity, to our efforts to understand spoken communication.

Contributed Paper

3:55

1pSC8. The clockwork music of speech: Gestural synthesis in 18th and 19th century speaking machines. Gordon Ramsay (Dept. of Pediatrics, Emory Univ. School of Medicine, Marcus Autism Ctr., 1920 Briarcliff Rd. NE, Atlanta, GA 30329, gordon.ramsay@emory.edu)

Gestural theories of phonology developed in the latter half of the 20th century have proposed that intrinsically timed sequences of linguistically significant actions of the vocal tract are the natural embodiment of speech, in stark contrast to earlier theories that emphasized the role of pure acoustic, auditory or articulatory representations in speech production and perception. Although these recent theories apparently represent a significant change in viewpoint, and have been highly influential, it has been largely forgotten that most of these ideas can actually be traced back to the 18th and 19th centuries, when they were the dominant perspective. Remarkably, early attempts to create speaking robots considered from the outset, and endeavoured to replicate, not only the physics of sound production in realistic vocal tract geometries, but also the sequencing of those geometries over time. This presentation traces the prehistory of gestural synthesis by using unpublished historical documents to analyze the mechanisms used to create the Abbé Mical's Talking Heads, the first programmable speech synthesizer producing the first conversation between machines, and Joseph Faber's Euphonia, the first mechanical synthesizer to transduce timed sequences of symbols into speech. Both of these mechanisms have direct implications for creating 21st century speaking robots.

4:10–4:45 Panel Discussion

MONDAY AFTERNOON, 28 NOVEMBER 2016 — SOUTH PACIFIC 1, 1:30 P.M. TO 5:25 P.M.

Session 1pSPa

Signal Processing in Acoustics: Time Delay Estimation of Acoustic Signals with Applications

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

Kenta Niwa, Cochair
Media Intelligence Laboratories, NTT, 3-9-11, Midori-Cho, Musashino-Shi 1808585, Japan

Cliff Carter, Cochair

Chair's Introduction—1:30

Invited Papers

1:35

1pSPa1. Time delay estimation of broadband acoustic signals and its defense applications. Brian G. Ferguson (Maritime Div., DSTG, PO Box 44, Pyrmont, NSW 2009, Australia, Brian.Ferguson@dsto.defence.gov.au), Michael V. Scanlon (US Army Res. Lab., Adelphi, MD), and Jay W. Chang (US Army RDECOM-ARDEC, Picatinny, NJ)

The measurement of the difference in the times of arrival (or time delay) of an acoustic signal at two spatially separated sensors is fundamental to localizing its source. For continuous broadband signals, the time delay is best estimated by the generalized cross-correlation of a pair of sensor outputs. The lag at which the generalized cross-correlation function reaches its maximum value provides an estimate of the time delay. Generalized cross-correlograms are presented for sensor output data recorded at sea on a submarine. The correct peak is readily identified by cross-correlating the conventional (and also the adaptive) beamformer outputs of a pair of widely-spaced arrays, rather than the outputs of two single hydrophones with the same separation distance (66.3 m). Similarly, the use of various prefiltering methods is shown to suppress secondary (ambiguous) peaks in the cross-correlogram. This approach is then extended to include mechanical and biological underwater acoustic transient signals with the time delay estimates being used to locate the source using the principle of passive ranging by wavefront curvature. Finally, the role of time delay estimation in the localization of aircraft, ground vehicles, and weapon fire using unattended land-based acoustic surveillance systems is demonstrated using real data.

1:55

1pSPa2. Real-time time difference of arrival estimation for moving wideband sources. Alexander Sedunov (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asedunov@stevens.edu)

The modified generalized cross-correlation method is used for finding the time difference of arrival (TDOA) and resolving the angle of arrival toward moving targets in real-time. It addresses the decorrelation due to differential Doppler effect leading to relative time companding (RTC) between sensors. The proposed method efficiently approximates the RTC prior to the computation of cross-correlation, thus working faster than methods using maximization of the passive ambiguity function and other approximation methods. The algorithm is based on computing the spectrum of deskewed short-time cross-correlator (DSTC) in the frequency domain by averaging a short-term spectrogram computed on overlapped segments of analysis in a time window. The same spectrogram is used to compute magnitude-squared coherence (MSC), then by maximizing the average MSC in the band of interest, the RTC is found. It is used to compute a single inverse Fourier transform. This method is applied to data collected by the Acoustic Aircraft Detection system that was developed at Stevens and that comprises of a volumetric 5-microphone cluster connected to a data acquisition and an embedded computer that processes the data in real-time. Examples demonstrate the improved TDOA finding for aircraft within an airport at a larger distance by using longer analysis time window and maintenance of tracking at close range, thanks to the RTC compensation. [This work was supported by DHS's S&T Directorate.]

2:15

1pSPa3. Phase shifts in passive fathometer processing. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu) and Peter Gerstoft (Scripps Inst. of Oceanogr., La Jolla, CA)

In passive fathometer processing, the presence of wavelets/sincs in the estimate of the medium's Green's function corresponds to the location of reflectors in the seabed; the amplitudes of the wavelets point to properties of the reflecting layers. Particle filtering has been successful in tracking reflectors that define layer interfaces from data from the Boundary 2003 Experiment after both conventional and MVDR processing. Further working on the problem, however, revealed that phase shifts may be present in the wavelets of the Green's function, which hinder accurate layer identification if unresolved. In this work, using a process similar to time delay estimation, we identify and model these shifts and demonstrate the significance of their modeling in accurate reflector identification as well as accurate estimation of reflector strengths. [Work supported by ONR.]

2:35

1pSPa4. Time delay estimation in the presence of source or receiver motion. Brian G. Ferguson and Kam W. Lo (Maritime Div., DSTG, PO Box 44, Pyrmont, NSW 2009, Australia, Brian.Ferguson@dsto.defence.gov.au)

Generalized cross-correlation of the outputs of two spatially-separated sensors provides an estimate of the time delay (or differential time of arrival) of a broadband acoustic signal at the pair of sensors. The time lag at which the cross-correlation function attains its maximum value corresponds to the time delay. However, if relative motion between the source and the sensors leads to the time scales of the received signals being mismatched, then the time delay will be in error. This mismatch in the time scales is also referred to as the differential Doppler effect. Two practical examples are presented that demonstrate this effect, which is observed when a jet aircraft transits over and along the axis of two widely-spaced microphones or a scuba diver travels past two spatially separated hydrophones. It is shown that the correct time delay and relative time scale can be obtained by implementing wideband cross-correlation with differential Doppler compensation using the continuous wavelet transform. The continuous wavelet transform has the same functional form as the more familiar wideband cross-ambiguity function.

2:55

1pSPa5. Huge microphone array to maximize mutual information between many microphones and sound sources. Kenta Niwa (Media Intelligence Labs., NTT, 3-9-11, Midori-Cho, Musashino-Shi, Tokyo 1808585, Japan, niwa.kenta@lab.ntt.co.jp)

Hardware technological evolution makes it possible to capture over 100 signals and apply complex processing for, e.g., source enhancement in real time. Assuming that many microphones are available and we want to emphasize a desired sound source located at a distance in noisy environments, how do we place them? In this talk, the principle of spatial sampling to maximize the mutual information between multiple inputs (microphones) and multiple outputs (sound sources) is discussed. The mutual information is commonly used for measuring how much information related to the sound sources is included in many observed signals. In our previous work, we found that it is possible to increase the mutual information by surrounding the microphones with reflectors. Since the effective spatial cues for identifying a desired sound source are included in observed signals, accurate source enhancement would be achieved by applying appropriate mapping functions. Beamforming with/without post-Wiener filtering is used as an implantation of a mapping function. Experiments for emphasizing a desired sound source located 16.5 meters from the constructed microphone array were conducted. Pinpoint pick up of the desired sound source was achieved.

3:15

1pSPa6. Acoustic distance measurement based on phase interference between transmitted and reflected waves. Masato Nakayama, Takanobu Nishiura (College of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Noji Higashi, Kusatsu, Shiga 525-8577, Japan, mnaka@fc.ritsumei.ac.jp), and Noboru Nakasako (Faculty of Biology-Oriented Sci. and Technol., Kindai Univ., Kinokawa, Japan)

The distance to talkers and objects is very important information for both hands-free speech interfaces and nursing-care robots. However, conventional distance measurement methods cannot measure short distances because the transmitted wave, which has not attenuated sufficiently by the time the reflected waves are received, suppresses the reflected waves for short distances. Therefore, we have proposed an acoustic distance measurement (ADM) method based on interference between the transmitted and reflected waves, which can measure distance even in a short range. In this presentation, we explain principle of the ADM method, and various studies. In the proposed method, the arbitrary audible sound with wideband is emitted as the transmitted wave with a loudspeaker, and the transmitted wave is recorded with a microphone. Thus, we can estimate the distance to a target with single loudspeaker and single microphone. Also, the localization of a target can be achieved using single loudspeaker and multiple microphones on basis of the acoustic imaging with microphone-array. In addition, we have improved the ADM method in order to estimate the target distance in noisy environments. Finally, we confirmed the effectiveness of the proposed method through experiments in real environments.

3:35–3:55 Break

3:55

1pSPa7. Phase DEMON algorithm for time delay estimation used in small boat tracking. Alexander S. Pollara, Alexander Sutin (Maritime Security Ctr., Stevens Inst. of Technol., 1 Castle Point on Hudson, The Babbio Ctr., Hoboken, NJ 07030, apollara@stevens.edu), Kil W. Chung (Maritime Security Ctr., Stevens Inst. of Technol., Gyeonggi-do, Hwaseong-si, South Korea), and Hady Salloum (Maritime Security Ctr., Stevens Inst. of Technol., Hoboken, NJ)

The Detection of Envelope Modulation on Noise (DEMON) algorithm is a widely used tool in underwater passive acoustics for the detection and classification of vessel sound. The DEMON algorithm extracts the frequencies that modulate the high frequency cavitation noise created by a vessel's propeller. We propose an extension of the DEMON method for time delay estimations of acoustic signals received by two or more hydrophones. This method, based on the phase difference between components in the two DEMON spectra received by different hydrophones, allows the extraction of the Time Difference Of Arrival (TDOA) and direction of arrival of the modulated signals. This method was applied to the acoustic signatures of six small boats collected by Stevens in a large glacial lake in NJ and showed agreement with the traditional cross-correlation method of TDOA estimation and boat GPS tracks. This method allows the separation of several boats TDOA. The DEMON algorithm also provides information of potential use for vessel classification. [This work was supported by DHS's S&T Directorate.]

4:10

1pSPa8. Measuring low-frequency ocean acoustic coherence with an estimator-correlator. Matthew Dzieciuch (SIO/UCSD, 9500 Gilman Dr., IGPP-0225, La Jolla, CA 92093-0225, mad@ucsd.edu)

Acoustic signals in the ocean are scattered by a variety of processes, but the end result is a partially coherent signal. The signal processing solution for estimating the travel-time of a partially coherent signal is to use an estimator-correlator (EC). An assumption of the EC is that the coherence time and the coherent bandwidth are known. When used properly, the EC leads to in increase in the SNR of the detected signal as well as reducing the width of the detected peak. An interesting question is "Can the EC be used to estimate the coherence parameters by maximizing the signal SNR and minimizing its width?" This is in contrast to the standard method of estimating the signal coherence from the sample covariance of the matched filter output. A simple model is constructed to test this proposition and is compared to experimental results obtained in the Philippine Sea.

4:25

1pSPa9. Perceptual thresholds of spatial audio update latency in virtual auditory and audiovisual environments. Narayan Sankaran, James Hillis, Marina Zannoli, and Ravish Mehra (Res., Oculus, 8747 148th Ave. NE, Redmond, WA 98052, ravish.mehra@oculus.com)

When observers in a virtual sound environment are in motion relative to a source, the virtual-auditory display must rapidly track the users head position and update the location-cueing acoustic filters—known as head-related transfer functions (HRTFs)—in order to accurately reflect the source's location relative to the current head orientation and position. The end-to-end spatial audio system latency (SASL) is the time elapsed between the listener assuming a position and sound being delivered to the ears with an HRTF corresponding to that same position. To maintain the stability and plausibility of a virtual sound "object," the SASL must lie below a perceptual temporal threshold. The current study sought to rigorously probe the threshold of SASL detectability in human observers at three different rotational head velocities and at various source locations. These conditions were repeated in an audiovisual experiment in which the sound stimulus was now accompanied by a zero-latency visual stimulus. Thus, we characterized SASL thresholds in environments where cross-modal interactions occur due to visual information accompanying sound as is often the case in virtual reality (VR) applications. Initial results reveal a rich interaction between the contextual factors of the virtual environment and listeners' sensitivity to SASL.

4:40

1pSPa10. Self-shape estimation algorithm for a flexible ultrasonic array probe. Yoshiaki Nakajima, Kazuhiro Matsui, Takashi Azuma, Etsuko Kobayashi, and Ichiro Sakuma (The Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo 113-8654, Japan, nakajima@bmpe.t.u-tokyo.ac.jp)

A self-shape estimation algorithm for a flexible ultrasonic array probe was described in the paper. Position information of each element in the array is essentially required to achieve a focal control in ultrasonic imaging process. The purpose of this study is to develop an algorithm to estimate array shape without other sensors. In our proposed algorithm, beam image (BI) was used as an evaluation function in the estimation of the shape. BI is an image representing a transmitted beam profile in the imaging object. BI was obtained by scanning of receive focal points around the transmit focal point. The quality factor of BI was used as an evaluation function and parameters describing assumed shapes were searched. We conducted simulation and gel experiments with commercially available flexible probe unit. The algorithm was evaluated by the quality of an estimated self-shape and reconstructed images using an estimated self-shape. The theoretical lateral and depth resolutions were 0.5~1mm and 0.3mm in light of transmitted beam profile. The algorithm could have estimated free curve shapes in error by less than 1mm and the sizes of imaged wires were 0.74~1.5 mm and 0.625~1.25 mm in lateral and depth direction.

4:55

1pSPa11. Analysis of a baffled circular array to avoid ambiguity in detection of arrival estimation. Fabricio A. Bozzi, William S. Filho (Signal Processing, Brazilian Navy Res. Inst., Rio de Janeiro, Brazil), Fernando P. Monteiro, Fabio O. Silva (Acoust. Instrumentation, Brazilian Navy Res. Inst., Rio de Janeiro, Brazil), and Leonardo M. Barreira (Signal Processing, Brazilian Navy Res. Inst., Rua Ipiru n04, Jd Guanabara, Rio de Janeiro, Brazil, barreira@ipqm.mar.mil.br)

The number of sensors present in an array imply in cost, length, weight, and computational complexity. So, it is desired that an array satisfies its project purpose using the minimum of elements. In this study, we analyze the characteristics of a uniform circular array (UCA). Previous studies show the ambiguity problem when working with only few sensors. The grating lobes and potentials ambiguities in uniform linear array are generally avoided limiting the space between sensors in half of the wavelength. In UCA, these problems are solved also by limiting the space between sensors using an adequate number of sensors for a given radius or changing the radius for a given number of sensors. This study shows that it is possible to avoid the grating lobes controlling the sensor's directivity. The directional gain model is used to represent a realistic baffled vertical stave, three hydrophone each, and the Multiple Signal Classification (MUSIC) is applied to detect sources. Simulation results show the accuracy when using directives sensors. This array is analyzed here, working with experimental data, acquired in an acoustic tank.

5:10

1pSPa12. Acoustic distance measurement based on the interference between transmitted and reflected waves using cross-spectral method by introducing analytic signal of linear chirp sound. Noboru Nakasako, Shinya Honda (Faculty of Biology-Oriented Sci. & Tech., Kindai Univ., Nishi-Mitani 930, Kinokawa, Wakayama 649-6493, Japan, nakasako@waka.kindai.ac.jp), Toshihiro Shinohara (Faculty of Biology-Oriented Sci. & Tech., Kindai Univ., Kinikawa, Nishi-mitani 930, Wakayama, Japan), Masato Nakayama (College of Info. Sci. & Eng., Ritsumeikan Univ., Kusatsu, Nojihigashi, 1-1-1, Shiga, Japan), and Tetsuji Uebo (Faculty of Biology-Oriented Sci. & Tech., Kindai Univ., Kinokawa, Nishi-mitani 930, Wakayama, Japan)

The distance to a target is fundamental information in many engineering applications. Recently, an acoustic distance measurement (ADM) method has been proposed based on the interference between transmitted and reflected waves, but it requires two applications of the Fourier transform. The ADM method in which a linear chirp whose frequency changes linearly

with lapse of time is adopted as a transmitted sound has been also proposed. However, due to the influence of the measuring system from the loudspeaker to the microphone, the ADM would often estimate the spurious short distance different from true distance. This paper describes a fundamental study on the ADM method by applying the cross-spectral method to observed signals of the adjacent two-channel (2ch) microphones, adopting a linear chirp as transmitted wave and removing the influence of the measuring system. More concretely, though the linear chirp in a time domain corresponds to the frequency spectrum in a frequency domain, it is not a complex signal but a real signal. To apply the cross-spectral method, the analytic signal of linear chirp is introduced. We confirmed the validity of the chirp-based ADM method by performing a computer simulation and by applying it to an actual sound field.

MONDAY AFTERNOON, 28 NOVEMBER 2016 CORAL 3, 3:15 P.M. TO 5:35 P.M.

Session 1pSPb

Signal Processing in Acoustics and Speech Communication: Spoken Document Processing

John H. L. Hansen, Cochair
Jonsson School of Engineering & Computer Science, CRSS: Center for Robust Speech Systems; UTDallas, 800 W Campbell Road, The University of Texas at Dallas, Richardson, TX 75080-3021

Norihide Kitaoka, Cochair
Tokushima University, 2-1 Minami-Johsanjima-cho, Tokushima 770-8506, Japan

Invited Papers

3:15

1pSPb1. Signal processing challenges for reconstructing NASA's Apollo missions to the moon. Douglas W. Oard (College of Information Studies, Univ. of Maryland, College Park, MD 20740, oard@umd.edu) and John H. Hansen (Univ. of Texas at Dallas, Richardson, TX)

During the Apollo Program, NASA recorded dozens of audio channels to document the coordination activity in the Apollo Mission Control Center. We have digitized these recordings for the entire Apollo 11 mission, but several challenges arise when seeking to use these unique recordings for mission reconstruction. One is that 50 of the 56 recorded channels contain headset audio that documents the mix of radio and intercommunication "loops" that an individual flight controller actually heard. Reconstructing the audio that was available for selection therefore requires multi-channel source separation. The same loop might be recorded on multiple time-synchronized channels, and the some known speakers routinely speak only on certain loops, so a considerable amount of side information is available for this source separation task. Second, several additional audio channels are available from different recorders, notably including recordings made aboard the two Apollo spacecraft). Integrating these additional channels requires content alignment techniques, which can leverage both speech content and incidentally recorded non-speech signals. Third, we have only limited metadata for each channel, indicating (for example) which headset audio was intended to be recorded on that channel and who (by name) might have used that deadest. Signal processing techniques can help to enrich that metadata, helping (for example) to recognize speaker changes associated with shift changes or specific mission events.

3:35

1pSPb2. Video semantic indexing and localization. Koichi Shinoda (Dept. Comput. Sci., Tokyo Inst. of Technol., 2-12-1 W8-81 Okayama, Meguro-ku Tokyo 152-8552, Japan, shinoda@cs.titech.ac.jp)

Nowadays Internet traffic has been largely occupied by consumer video but most of them are not accompanied with text tags for search. Hence, video semantic indexing, which extracts visual concepts such as objects, scenes, and actions directly from video contents, has been intensively studied. Fundamentally, this task consists of two problems: localization and recognition. While until recently these two problems have been studied independently, emerging end-to-end deep learning techniques using convolutional neural networks (CNNs) and recurrent neural networks (RNNs) offer effective ways to solve them simultaneously. These techniques are deeply related to spoken word detection techniques in the speech field. In this talk, we overview the recent progress in this area and discuss potential directions for future research.

3:55

1pSPb3. Challenges and opportunities in spoken document processing: Examples from keyword search and the use of prosody.
Andrew Rosenberg (IBM Watson, IBM TJ Watson Res. Ctr., Yorktown Heights, NY 10598, amrosenb@us.ibm.com)

Spoken document processing presents challenges and opportunities when compared to text processing. Speech transcription contains errors, but speech conveys information beyond the words that are said. To deal with errors, a spoken document should be viewed as a structure of viable hypotheses not an absolute transcription. Also, the manner in which words are spoken, their prosody, can be mined for information about the speaker and his or her intent. This talk will use keyword search as an case study that requires operating under errorful and adverse conditions. It will focus on the efforts of the IBM-led LORELEI Consortium during the IARPA BABEL program over the last four and a half years. This period has been a time of rapid change in automatic speech recognition, and the BABEL program has served as a proving ground for a number of innovations including DNNs for acoustic modeling, multi-lingual acoustic features, graphemic (vs. phonemic) recognition, active learning (and reduced resource conditions), morphological analysis, and so-called "end-to-end" speech recognition. The talk will then present areas of opportunity to leverage information communicated via prosody to improve spoken document processing including emotion recognition, dialog act classification, and information extraction on spoken documents.

4:15

1pSPb4. Taking advantage of spontaneous speech for document retrieval: Lessons from the organization of evaluation tasks.
Tomoyosi Akiba (Dept. of Comput. Sci. and Eng., Toyohashi Univ. of Technol., 1-1 Hibarigaoka, Tenpaku, Toyohashi, Aichi 441-8580, Japan, akiba@cs.tut.ac.jp)

This work reports our experiments on organizing evaluation tasks aiming at spoken document retrieval (SDR). The NTCIR Project is a series of evaluation workshops designed to enhance research in information access technologies by providing reusable test collections and a forum for researchers. The author's group has proposed the evaluation tasks for SDR at the NTCIR from 2010. They resulted in holding the four successive evaluation tasks called SpokenDoc-1 and 2, and SpokenQuery&Doc-1 and 2. In the SpokenDoc tasks conducted from 2010 to 2013, we designed and evaluated the SDR tasks targeting spontaneously spoken speech. Based on this data collection, we conducted two subtasks: the spoken term detection subtask and the spoken content retrieval subtask. After two rounds of them, we enhanced our research focus with the query part of information retrieval. In the SpokenQuery&Doc tasks conducted from 2013 to 2016, we investigated the information retrieval systems that take advantage of a spontaneously spoken query as a medium of expressing one's information need. We have collected spontaneously spoken queries through subjective experiment to construct search topics for information retrieval. All of those experiences has been compiled into the reusable test collections provided at the NTCIR project, which enables researchers to test their ideas anytime on the common testbeds. In this presentation, we will report our task design, methodologies and challenges on constructing our data, and outcomes found through our organization.

Contributed Papers

4:35

1pSPb5. Prof-Life-Log: Monitoring and assessment of human speech and acoustics using daily naturalistic audio streams. John H. L. Hansen, Abhijeet Sangwan, Ali Ziaei, Harishchandra Dubey, Lakshmish Kaushik, and Chengzhu Yu (Jonsson School of Eng. & Comput. Sci., CRSS: Ctr. for Robust Speech Systems; UTDallas, 800 W Campbell Rd., The Univ. of Texas at Dallas, Richardson, TX 75080-3021, john.hansen@utdallas.edu)

Speech technology advancements have progressed significantly in the last decade, yet major research challenges continue to impact effective advancements for diarization in naturalistic environments. Traditional diarization efforts have focused on single audio streams based on telephone communications, broadcast news, and/or scripted speeches or lectures. Limited effort has focused on extended naturalistic data. Here, algorithm advancements are established for an extensive daily audio corpus called Prof-Life-Log, consisting of + 80days of 8-16 hr recordings from an individual's daily life. Advancements include the formulation of (i) an improved threshold-optimized multiple feature speech activity detector (TO-Combo-SAD), (ii) advanced primary vs. secondary speaker detection, (iii) advanced word-count system using part-of-speech tagging and bag-of-words construction, (iv) environmental "sniffing" advancements to identify location based on properties of the acoustic space, and (v) diarization interaction analysis which highlights the amount of speech each individual produces along with recipient direction. The diarization advancements are evaluated on CRSS-UTDallas Prof-Life-Log. Results show improvements in speech activity detection, word count estimation, and environmental sniffing for naturalistic audio streams. The advancements suggest improved speech/language processing algorithms can address the increased diversity in daily audio for speaker knowledge detection/estimation and summarization.

4:50

1pSPb6. The role of consonants in the perception of vocal attractiveness. Emily Blamire (Linguist, Dept. of Linguist, Univ. of Toronto, Sidney Smith Hall, 4th Fl., 100 St. George St., Toronto, ON M5S 3G3, Canada, emily.blamire@mail.utoronto.ca)

Research on vocal attractiveness has identified several acoustic cues intrinsic to vowels (e.g., pitch) that affect judgements of attractiveness and have shown listeners generally agree in their ratings (e.g., Babel, McGuire, & King 2014). Research in this area has, however, been largely limited to an examination of vowels. The current study builds on past work by examining an array of acoustic cues intrinsic to consonants to ascertain how they affect judgements of vocal attractiveness. Two native speakers of Canadian English were recorded saying one-syllable words of the form sCVC or p/t/kVC. The acoustic cues of center of gravity and duration of /s/ and voice onset time of /p/, /t/, and /k/ were manipulated to both increase and decrease the value of each cue. Vowel pitch was also included in the analysis as a baseline for comparison with past research. Participants listened to the original and manipulated tokens and were asked to judge vocal attractiveness using a 7-point Likert scale. The data of 32 people (female, N = 16) aged 19-32 were analyzed, using mixed effects models. The most interesting finding of the current study was that while not all acoustic cues of consonants played a role in perceptions of vocal attractiveness, /s/ duration did, and for this cue both the speaker and listener sex were important to how the cue was perceived. This suggests that not only are male and female voices judged by different standards as to what is attractive, but that men and women are using different criteria when making these judgements.

5:05

1pSPb7. A speech pre-processing method to reduce overlap masking in reverberant environments. Julian Grosse and Steven van de Par (Acoust. Group, Cluster of Excellence "Hearing4all", Univ. of Oldenburg, Carl-von-Ossietzky-Straße 9-11, Oldenburg 26129, Germany, julian.grosse@uni-oldenburg.de)

In daily life, we are often exposed to speech that is rendered over loudspeakers in a reverberant acoustical environment (examples are public-address systems used in a train station or conference halls). Whereas the early reflections can support speech intelligibility, the late reflections smear the speech in time which will result in an overlap of consecutive speech portions and an effective low-pass filtering of the speech-specific modulation spectrum. This study proposes a perceptually motivated pre-processing approach, based on previous work of Hodoshima *et al*. [J. Acoust. Soc. **119**, 4055-4064 (2006)], which reduces the detrimental effects of reverberation by suppressing steady-state portions and emphasizing potentially inaudible/masked segments of speech before it is emitted in the reverberant environment. For pre-processing, the impulse response is separated into a direct and a reverberant path to decide whether speech segments are inaudible and can be neglected or should be emphasized. A speech intelligibility prediction model is used to select the optimal pre-processing parameters for each specific acoustical environment. Listening tests showed, that this pre-processing approach is able to partially compensate the detrimental effects of reverberation leading to a reduction in speech reception thresholds of about 2 to 5 dB measured in speech shaped noise.

5:20

1pSPb8. An effective speech compression based on syllable division. Yan Zhang, Dong Xiao, Qunyan Ren, Shengming Guo, and Fuyuan Mo (Key Lab. of Underwater Acoust. Environment, Inst. of Acoust., Chinese Acad. of Sci., No. 21,BeiSiHuan XiLu, Beijing 100190, China, yudeyinji512233@126.com)

For medium or long distance acoustic speech communication, transmission rate is relatively low due to various disadvantageous factors such as multipath propagation, under such condition, speech compression methods are necessary for effective transmission. A good method needs to retain the characteristics of the speaker and achieve a good compression rate. Considering that the speech signal is usually redundant, a speech pre-compression algorithm based on syllable division is proposed. It can extract the voiced speech and the silence part of the speech signal, and subsequently delete the redundant part that does not affect the semantic comprehension. Through data processing, this pre-compression algorithm is demonstrated can compress the speech signal up to 80% and not affect listener differentiating or semantic comprehension.

MONDAY AFTERNOON, 28 NOVEMBER 2016 NAUTILUS, 1:00 P.M. TO 5:30 P.M.

Session 1pUW

Underwater Acoustics and Acoustical Oceanography: Underwater Acoustics Studies in Asian Seas I

Chi-Fang Chen, Cochair
Engineering Science and Ocean Engineering, National Taiwan University, No. 1 Roosevelt Road Sec.#4, Taipei 106, Taiwan

Ching-Sang Chiu, Cochair
Department of Oceanography, Naval Postgraduate School, 833 Dyder Road, Room 328, Monterey, CA 93943-5193

Chair's Introduction—1:00

Invited Papers

1:05

1pUW1. The North Pacific Acoustic Laboratory deep-water ocean acoustics experiments in the Philippine Sea: A review. Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu)

A series of experiments with participants from many institutions investigated deep-water ocean acoustics in the oceanographically and geologically complex Philippine Sea during 2009-2011. These experiments employed various configurations of a newly developed distributed vertical line array (DVLA) receiver. The time fronts recorded on the DVLAs for transmissions from a variety of acoustic sources, both moored and lowered form shipboard, provide a rich data set on acoustic scattering and variability due to internal waves, internal tides, density-compensated temperature and salinity variations (spice), oceanic fronts, eddies, and bottom interaction. During 2010-2011, a water-column-spanning DVLA was embedded within an ocean acoustic tomography array of six transceivers, enabling the observation of acoustic time fronts and measurement of ambient noise over a 5000-m aperture. The observed travel-time time series were similar to time series computed from ocean state estimates made using an eddy-permitting regional ocean model constrained by non-acoustic data, and the travel-time differences have in turn been successfully used to further constrain the model. Acoustic Seagliders also recorded the tomographic transmissions, and the travel times have been used to test the feasibility and accuracy of long-range acoustic navigation (Underwater GPS). Analyses of the data sets collected during the NPAL Philippine Sea experiments are continuing.

1:25

1pUW2. Single-mode field excitation experiment in shallow water. Juan Zeng, Da Yong Peng, Wen Yao Zhao, Hai Feng Li, and Hai Jun Liu (Inst. of Acoust., CAS, No. 21 Bei Si Huan Xi Lu, Beijing 100190, China, zengjuan_ioa@sina.com)

Single-mode field has advantages in active detection, underwater communication, bottom backscattering measurement, etc. Some of papers have reported on the theory and simulation about the single-mode excitation by open- and closed-loop methods, respectively. The experiment in the tank has also been reported many years ago, but the experiment in the water has not been reported yet. In this paper, an effective closed-loop method for exciting the single-mode field is discussed, which can obtain the optimum estimation of the matrix of the Green's function in the shortest feedback times, with high efficiency of emission of the single-mode signal. The experiments at the different sites in different seasons are reported. The vertical source array with the center frequency at 700 Hz was used as the source in the experiment. The experimental results show that when the source array spans almost the effective water depth the single-mode field with high quality can be excited. The first three single modes were excited at the sites with the water depth of 14 m and 36 m respectively. The experimental results show that more than 97% of the energy of the received field is concentrated in the single-mode.

1:45

1pUW3. Shallow-water acoustic variability experiment 2015 (SAVEX15) in the northern East China Sea. Hee-Chun Song (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu), Sea-Moon Kim (Korea Res. Inst. of Ships and Ocean Eng., Dajeon, South Korea), Byoung-Nam Kim (Korea Inst. of Ocean Sci. and Technol., Ansan, South Korea), and SungHyun Nam (Seoul National Univ., Seoul, South Korea)

The SAVEX15 experiment was conducted in shallow water (~100 m deep) in the Northern East China Sea (ECS), ~100 km southwest of Jeju Island, South Korea, over the period 14-28 May 2015. The goal of SAVEX15 was to obtain acoustic and environmental data appropriate for studying the coupling of oceanography, acoustics, and underwater communications in the ECS. Surprisingly, an underwater sound channel (USC) typical in deep water was discovered in this shallow water waveguide with the channel axis at ~40 m, apparently due to the cross-frontal interleaving of water masses carried by the Kuroshio Current and Changjiang river discharge. This talk presents an overview of SAVEX15 including preliminary research results.

2:05

1pUW4. Acoustic experiments over the South China Sea upper-slope sand dunes. Ching Sang Chiu (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943, chiu@nps.edu), Linus Y. Chiu (Inst. of Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan), Chi-Fang Chen (Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), Yiing Jang Yang (Inst. of Oceanogr., National Taiwan Univ., Taipei, Taiwan), Ruey Chang Wei (Inst. of Undersea Technol., National Sun Yat-sen Univ., Kao-Hsiung, Taiwan), Steven R. Ramp (SOLITON Ocean Services, Inc., Monterey, CA), Chris Miller (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), D. Benjamin Reeder (Dept. of Oceanogr., Naval Postgrad. School, Honolulu, Hawaii), and Andrea Y. Chang (Asia-Pacific Ocean Res. Ctr., National Sun Yat-sen Univ., Kaohsiung, Taiwan)

A series of experiments were methodically carried out between 2012 and 2014 to study the impact of large underwater sand dunes and the combined impact of these sand dunes and large-amplitude nonlinear internal waves on sound propagation over the upper continental slope including anisotropic propagation characteristics and focussing/defocussing scattering phenomena. The spatial distribution and scales of the sand dunes were first mapped by two multibeam echo sounding (MBES) surveys in 2012 and 2013. The 2013 experiment also provided some coring and initial acoustic transmission data to give useful knowledge of the geoacoustic properties of the sand dunes based on forward propagation modeling and least-squares fitting to the measured levels. The 2014 experiment was more comprehensive, entailing the deployment of autonomous mobile sources, a towed source and a moored source transmitting signals in different frequency bands and different geometries to a vertical line hydrophone array. Our data analysis results, aided with modeling interpretations, have provided physical insights into the observed spatial and temporal propagation characteristics. These results will be highlighted in this presentation. [The research is jointly sponsored by the Taiwan MOST and the US ONR.]

2:25

1pUW5. Nonlinear internal wave characteristics during the Sand Dunes 2014 acoustic propagation experiment. Steven R. Ramp (Soliton Ocean Services, Inc., 691 Country Club Dr., Carmel Valley, CA 93924, sramp@solitonocean.com), Yiing Jang Yang (Inst. of Oceanogr., National Taiwan Univ., Taipei, Taiwan), Ching-Sang Chiu (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), and Frederick L. Bahr (Monterey Bay Aquarium Res. Inst., Moss Landing, CA)

The Sand Dunes 2014 field experiment in the South China Sea was located in a region frequented by large amplitude (40-80 m) nonlinear internal waves. Four moorings spanning 386-266 m across the slope near 21° 52'N, 117° 36.5'E observed wave arrival patterns during June 3-19, 2014, that were similar to those previously observed nearby consisting of large (a-waves) arriving diurnally with smaller (b-waves) in between, with all amplitudes modulated by the fortnightly tidal beat. Small waves continued to gain amplitude as they shoaled, whereas large waves maxed out near 342 m then became smaller between 342 and 266 m. The number of waves/packet increased upslope due to wave dispersion. With just one exception, wave breaking and trapped cores were not observed. This contrasts with previous observations to the southwest near Dongsha Island where the wave energies (550 vs. 300 MJ) and bottom slopes were larger. Most of the incident waves can be attributed to remote forcing in the Luzon Strait, although some b-waves were formed locally due to convergence of the internal tide. A new result was the arrival of double a-waves two hours apart near spring tide on June 16-19. Some possible forcing scenarios for these double a-waves will be presented.

2:45

1pUW6. Analysis of low frequency ocean ambient noise induced by internal wave in sand dune region of northern South China Sea. Ruey-Chang Wei (Inst. of Undersea Technol., National Sun Yat-sen Univ., 70 Lienhai Rd., Kaohsiung City 804, Taiwan, rcwei@mail.nsysu.edu.tw), Kuo-Feng Chien, Linus Y.S. Chiu (Inst. of Undersea Technol., National Sun Yat-sen Univ., Kaohsiung City, Taiwan), and Chi-Fang Chen (Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan)

Most powerful internal waves in oceans were found in South China Sea, which were generated in Luzon Strait and mainly driven by tides. In past decades, researches of internal wave's effects on ocean processes have been concentrated on the slope of continental shelf, where transformations of internal waves were significant, so were the induced phenomena. In recent years, large groups of underwater sand dunes were discovered along the upper continental slope, whose formation was speculated to be related to internal wave's energy. In 2014, acoustical and oceanographic moorings were deployed in the sand dune region to study acoustical fluctuations due to the coupled effects of internal waves and sand dunes. Ocean ambient noise data were collected by four SHRUs (Several Hydrophone Recording Unit) and one VLA (Vertical Line Array) at different depths, analysis, and comparison of noise during internal wave events were presented in this study. Low frequency noises were generated by the passing of internal wave, in forms of agitations of sea surface, dramatic fluctuations of water column, and also strumming of mooring structure. Directivity of induced noise helps to identify the mechanism of source, and in addition spatial variation can show the influences from the topography of sand dune.

3:00–3:15 Break

3:15

1pUW7. Acoustic propagation in the South China Sea: Internal waves and sand dunes. D. Benjamin Reeder (Dept. of Oceanogr., Naval Postgrad. School, 73 Hanapepe Loop, Honolulu, HI 96825, dbreeder@nps.edu), Andrea Y. Chang (Inst. of Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan), Chi-Fang Chen (Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), Ching-Sang Chiu (Dept. of Oceanogr., Naval Postgrad. School, Monterey, California), Linus Y. Chiu (Inst. of Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan), Chris W. Miller (Dept. of Oceanogr., Naval Postgrad. School, Monterey, California), Steven R. Ramp (SOLITON Ocean Services, Inc., Carmel Valley, CA), Ruey C. Wei (Inst. of Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan), and Yiing J. Yang (Inst. of Oceanogr., National Taiwan Univ., Taipei, Taiwan)

Very large subaqueous sand dunes were discovered on the upper continental slope of the northeastern South China Sea (SCS) in the spring of 2007 during the ONR 322OA-funded NLIWI Acoustics field experiment. The dunes' formation mechanism is hypothesized to be the internal solitary waves (ISW) which generate from tidal forcing on the Luzon Ridge on the east side of the SCS, propagate west across the deep basin with amplitudes regularly exceeding 125 m, and dissipate large amounts of energy via turbulent interaction with the continental slope, suspending and redistributing the bottom sediment. These internal waves and sand dunes are important acoustical features, as it is expected that they will cause significant anomalies in the acoustical field. Data analysis and modeling are presented to quantify the degree to which these features impact broadband (850-1200 Hz) signals propagating along an acoustic transect oriented perpendicular to the internal wave fronts and sand dune crests.

3:30

1pUW8. Observations of acoustic propagation and geoacoustic inversion affected by subaqueous sand dunes in the South China Sea. Linus Chiu (Inst. of Undersea Technol., No. 70, Lienhai Rd., Kaohsiung 80424, Taiwan, linus@mail.nsysu.edu.tw), Andrea Chang (Asia-Pacific Ocean Res. Ctr., Kaohsiung, Taiwan), Davis B. Reeder, Ching-Sang Chiu (Naval Postgrad. School, Monterey, CA), Yiing-Jang Yang, Chi-Fang Chen (National Taiwan Univ., Taipei, Taiwan), and Chau-Chang Wang (Inst. of Undersea Technol., Kaohsiung, Taiwan)

The large subaqueous sand dunes are expected to affect underwater acoustic propagation. Very large subaqueous sand dunes on the upper continental slope of the northern SCS were discovered in water depths of 160 m to 600 m, which composed of fine to medium sand. In this talk, serial acoustic experiments conducted by Taiwan and United States in the South China Sea during 2012-2014 are overviewed and mid-frequency propagation data/model as well as mid-frequency geo-acoustics are aimed. For mid-frequency propagation, results demonstrate that subaqueous sand dune bedforms fluctuate the distinguishable and dispersive mid-frequency acoustical channel; causing the least distinct arrival patterns in the sand dune area. Numerical simulations using broadband modeling given the adequate initial field and Pade term confirm the observations in the experimental data. This talk also presents experiment results of normal incidence survey tracks, and the errors in reflection coefficient estimation and the resulting sediment properties induced by sand dune bedforms. Results demonstrate that the reflected energy is focused and scattered by different parts of sand dune bedforms, and that they produce significant variation in the estimated reflection coefficients and the inverted geoacoustic properties. [This work was supported by the Taiwan MOST and the US ONR.]

3:45

1pUW9. Modeling study of underwater acoustic propagation over sand dunes in the South China. Stephanie Yang, Chi-Fang Chen (National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, r03525010@ntu.edu.tw), Ching-Sang Chiu, and Christopher W. Miller (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA)

There is an extensive field of underwater sand dunes located at depths between 160 m and 600 m in the northeastern South China Sea. The largest amplitude of these sand dunes is about 16 m with horizontal length scales between 200 and 400 m. Underwater acoustic transmission can be affected by these topographic features. Two-dimensional (2-D) and three-dimensional (3-D) underwater acoustic propagation models, RAM and FOR3D, were employed to simulate the acoustic propagation over these sand dunes from a moored source to a vertical-line hydrophone array deployed in an experiment in 2014. Environmental input to the models were the measured bathymetry and sound speed profiles. To prevent model over initializations at steep angles in an ocean with a high-impedance, low-loss bottom, the generalized Gaussian starter was employed and tuned to match the angular beam-pattern of the moored source to give a proper starting field. Simulation results pertaining to the temporal fluctuations, 2-D and 3-D propagation effects in relation to the sand dunes and the water-column variability are presented and discussed. Simulation results are also compared to the measured transmission loss. [This research was supported by both the Ministry of Science and Technology of Taiwan and the Office of Naval Research of the USA.]

4:00

1pUW10. Influence of internal wave on the space-time interference patterns of broadband acoustic field. Tao Hu, Shengming Guo, Li Ma (Inst. of acousitcs, Chinese Acad. of Sci., Beisihuanxi Rd. 21, Beijing 100190, China, hutao76@aliyun.com), and Wenhua Song (Ocean Univ. of China, Qingdao, China)

The frequency shift of the acoustic interference striations is caused by the depth variation of the thermocline in shallow water waveguide. Based on the perturbation theory, a relationship between the frequency shift of the interference striations and the contoured sound speed profile is established. In a given frequency band, if the modal horizontal wave number difference

acts as a monotonic function of frequency, the frequency shift of interference striations has a direct correlation with the contoured SSP. This relationship is verified by simulation calculation as well as experiment data. It is found that if the propagation direction of sound is parallel to the wave front of internal wave, the frequency shift of the interference striations has a determined relationship with the contoured SSP that passing through the source; while when sound propagates in a direction perpendicular to the internal wave's wave front, the frequency shift is then associated with the contoured mean SSP in the entire sound propagation path.

4:15

1pUW11. Successive explosive sources for geoacoustic inversion: An attempt of particle filtering. Qunyan Ren, Licheng Lu, Jiawei Li (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), Tianjun Liao (State Key Lab. of Complex System Simulation, Beijing, China), Shengming Guo, and Li Ma (Key Lab. of Underwater Acoust. Environment,Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

This paper presents the results of sequential geoacoustic characterization of ocean bottom in shallow waters from geoacoustic measurement of successive explosive sources. The recording system is a moored vertical line array (VLA) composed of 32 elements, and the bomb sources were successively deployed from a moving ship. The ship straightly sailed at constant speed of 5 knots, and a total number of 15 bombs were deployed with a time interval of 3-min for adjacent bombs. A particle filtering (PF) technique is adapted here to estimate\track the spatial distributions of the geoacoustic parameters by filtering the complex pressure fields recorded on the VLA. As for references, independent batch inversion of conducted by Simulated Annealing (SA) algorithm are applied on bombs at different ranges. The general agreement between PF and SA suggests that, the PF is capable of mapping spatial geoacoustic parameter variability in complex ocean environments.

4:30

1pUW12. Sequential source localization in coastal water under the circumstances of time-evolving sound speed profiles. Lin Su, Shengming Guo, Qunyan Ren, and Li Ma (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, sulin807@mail.ioa.ac.cn)

The sound speed profile (SSP) in water column is highly variable in time and space and can not be accurately known among measurement periods in coastal water. These factors will lead to the modeling mismatch and influence the performance of source localization of matched filed processor (MFP). The possibility of sequential tracking of source range and depth under the circumstances of time-evolving sound speed profiles is discussed. In this paper, we attempt to describe the time evolution characteristics of SSPs by analyzing the oceanographic dynamics which is the physical reason of the SSP's change. A sequential approach considering the oceanographic dynamics model into the state-space equation is developed and applied. Synthetic simulations are carried out with experimental SSP data and acoustic pressure data from a vertical line array. Result suggests the sequential approach based on oceanographic dynamics model can improve the tracking capabilities.

4:45

1pUW13. A comparison of finite element and hybrid ray-tracing propagation models for shallow water environments. Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu), Jacob George, and David Harvey (Naval Oceanographic Office, Stennis Space Ctr., MS)

It has been found that the bottom loss can be significantly frequency dependent in shallow water waveguides due to scattering from the rough interface and the volume. This can cause transmission loss to increase with frequency for the same shallow water environment. One way of predicting this frequency dependence is through finite element (FE) propagation modeling which solves the Helmholtz equation exactly within the resolution of the discretization for a completely customizable domain. However, finite element models are difficult to compute for high frequencies and large regions. In this work, two hybrid models are considered and compared with the finite element exact solution. The first model uses a bottom loss calculated using FE for a rough interface as input into a ray based model. The second model uses an even simpler approach by characterizing the bottom loss with the Eckart formulation for use in the ray-based model. The performance of these models as a function of frequency and source and receiver geometry will be assessed. If successful, the simple Eckart formulation could be used to characterize the bottom loss for databases since depends on few parameters. [Work sponsored by the Naval Oceanographic Office and ONR, Ocean Acoustics.]

5:00

1pUW14. Quantifying the spatial variability of low frequency acoustic propagation in the Northern East China Sea. Sungho Cho, Donhyug Kang, Byoung-Nam Kim, Seom-Kyu Jung (Maritime Security Res. Ctr., Korea Inst. of Ocean & Sci. Technol., 787 Haean-ro(st), Sangnok-gu, Ansan 426-744, South Korea, shcho@kiost.ac.kr), and Jee Woong Choi (Dept. of Marine Sci. and Convergence Eng., Hanyang Univ., Ansan, Korea (the Republic of))

During a period 14-28 May, 2015, the Shallow-water Acoustic Variability EXperiment 2015 (SAVEX15) was made in the northern East China Sea (ECS) to obtain acoustic and oceanographic data for studying the coupling of physical and geophysical parameters, which could affect the variability of acoustic propagation. A strong underwater sound channel (USC) had been existed at depths ranging from 30 to 50 m with channel axis at ~40 m during the SAVEX15 period. Two types of mid-range propagation measurements were conducted in shallow water (nominal water depth of ~100 m) using simultaneously both continuous waves superimposed at several fixed frequencies below 1.6 kHz and impulsive broadband signals transmitted by sparker system. A vertical line array composed of temperature and pressure sensors was moored for measuring the acoustic signals and vertical sound speed profiles in time. And a marine geophysical survey using a chirp sonar, sparker system, and sediment cores conducted at the experimental site showed that there was a thin surficial layer of O(1 m) overlaying multiple layers under the water-seabed interface. The measurement results of transmission loss are presented and compared to the model predictions using the measured oceanographic data. [KIOST (Project Code No. PE99331).]

5:15

1pUW15. Range estimation of a broadband source using the array/waveguide invariant and a vertical array. Hee-Chun Song and Chomgun Cho (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu)

During a recent shallow-water experiment conducted in the northern East China Sea (SAVEX15), a low-frequency (0.5-2 kHz) source (SeaNos) was towed at the mid-depth (~50 m), being close to the channel axis of the underwater sound channel (USC) observed unexpectedly in a shallow water waveguide environment. It provided a variety of distances and propagation directions between the source and two moored vertical line arrays (VLAs). The channel impulse responses from the SeaNos indicate that the earlier reflected and refracted arrivals from the USC are followed by surface/bottom reflected arrivals. The later reflected arrivals are utilized for source-range estimation using the array/waveguide invariant based on the beam-time migration and the VLAs.

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: www.aip.org/

Aqua Sound, Inc.: aqua-sound.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/

1eMU

Musical Acoustics: Special Presentation of the History, Culture, Practice, and Performance of Hawai'ian Music

Andrew C. Morrison, Chair
Natural Science Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Chair's Introduction—7:00

Invited Paper

7:05

1eMU1. Hoʻolono i ka Leo o ka Makani (*Hear the Voice of the Wind*). Keawe Lopes, Puakea Nogelmeier, Jon Osorio (School of Hawaiian Knowledge, Univ. of Hawaii, Manoa, Spalding Hall, Honolulu, HI 96822, rlopes@hawaii.edu), and Aaran Salā (Univ. of Hawaii, Windward Community College, Kāneʻohe, HI)

This is a performance of oli, mele, and hula that explores the wide range of chant, song, and dance that Kanaka Maoli have both created and adopted to forge the unique and celebrated music of Hawaii. Scholar/musicians Keawe Lopes, Puakea Nogelmeier, Jon Osorio, and Aaron Sala combine their diverse talents, musical experience and love for the language in this presentation of music, dance, and conversation. Our music is in our nature. Hoʻolono i ka leo o ka makani.