Session 2aABa

Animal Bioacoustics: Tropical Bioacoustics I

Thomas F. Norris, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

2aABa1. Beaked whale species occurrence in the central Pacific and their relation to oceanographic features. Simone Baumann-Pickering, Anne E. Simonis, Jennifer S. Trickey (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., San Diego, CA), and Erin M. Oleson (Pacific Islands Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, Honolulu, HI)

Mesoscale oceanographic features are a major force in structuring the marine environment through processes such as eddy-induced upwelling, and as such effect distribution and aggregation patterns of all organisms along the food chain. It has been shown that top pelagic predators such as cetaceans react to these environmental changes in different ways. We present analysis of frequency-modulated (FM) echolocation pulses of Cuvier's beaked whale (*Ziphius cavirostris*), Blainville's beaked whale (*Mesoplodon densirostis*), and an unknown beaked whale species producing FM pulse type "BWC," possibly ginkgo-toothed beaked whale (*M. ginkgodens*), at five locations in the central Pacific. The recordings were collected at Pearl and Hermes Reef (Northwestern Hawaiian Islands), Kona (Main Hawaiian Islands), Wake Atoll, Tinian, and Saipan (Northern Mariana Islands) between 2008 and 2015, ranging from 4 to 8 years per site. All three beaked whale species were present at all sites in different proportions throughout the recording periods, with a strong nocturnal pattern only for the "BWC" pulse type, yet without seasonal pattern. We examine the varying presence in the context of remotely sensed oceanographic data, including sea surface height deviation, temperature, and salinity, as well as chlorophyll *a* and derived primary productivity.

8:25

2aABa2. Echolocation systems of leaf-nosed bats in Taiwan and Iriomote island, Japan. Hiroshi Riquimaroux (SDU-VT Int. Lab., Shandong Univ., 27 Shanda Nanlu, Jinan, Shandong 250100, China, hiroshi_riquimaroux@brown.edu)

The leaf-nosed bats, *Hipposideridae*, are one of the most common tropical bats found in Asia. Two species of leaf-nosed bats in Iriomote island, *Hepposideros turpis,* and Taiwan, *Hipposideros terasensis*, were compared. They emit short echo location pulses (4-7 ms in duration) consisted of constant frequency (CF) component followed by downward frequency modulated (FM) component. The second harmonics are the most intense frequency component. Comparison lists for two species are shown below. Taiwanese leaf-nosed bats, *Hipposiderous terasensis* CF2: 70 kHz (CF1: 35 kHz) 50-60 g in weight Japanese leaf-nosed bats, *Hepposideros turpis* CF2: 82 kHz (CF1: 41 kHz) 20-30 g in weight threatened species (Red list). Both Taiwan and Iriomote island are at around the same North latitude 25 degrees. Findings from *H. terasensis* and *H. turpis* are introduced and discussed. [Research supported by MEXT, Japan, and Shandong University, China.]

8:45

2aABa3. Melody in my head, melody in my genes? Acoustic similarity, individuality, and genetic relatedness in the indris of Eastern Madagascar. Marco Gamba, Valeria Torti, Giovanna Bonadonna (Life Sci. and Systems Biology, Univ. of Torino, Via Accademia Albertina 13, Torino 10123, Italy, marco.gamba@unito.it), Rose M. Randrianarison (Groupe d'Etude et de Recherche sur les Primates de Madagascar, Antananarivo, Madagascar), Olivier Friard, and Cristina Giacoma (Life Sci. and Systems Biology, Univ. of Torino, Torino, Italy)

Indris (*Indri indri*) are the only singing lemurs and produce different types of songs that can be differentiated according to their temporal patterns. The most distinctive portions of the songs are "descending phrases" consisting of 2-5 units. In our study, indri songs were recorded in the Eastern rainforests of Madagascar from 2005 to 2015. All the recordings were made when the recorder operator was in visual contact with the singing social group and by recognizing individual indris using natural markings. Because the individual songs frequently overlap during the chorus, we extracted the pitch contour of 1084 descending phrases using the program PRAAT. We tested whether the structure of the phrases could provide conspecifics with information about sex and individual identity. We also examined

whether the structure of the phrases was related to the genetic relatedness of the indris. The results suggest that the songs have consistent sex-, group-, and individual-specific features. We have also found a significant correlation between the genetic distance and the acoustic similarity. The descending phrases may be used by the indris to convey information of sex and identity, and genetic relatedness may play a role in determining song similarity to a larger extent than previously assumed.

9:05

2aABa4. Recent bioacoustics researches on biosonar, hearing, and noise effect on the Indo-Pacific humpback dolphins (*Sousa chinensis*). Songhai Li (Inst. of Deep-Sea Sci. and Eng., Chinese Acad. of Sci., 28 Luhuitou Rd., Sanya 572000, China, LISH@IDSSE.AC.CN) and Ding Wang (Inst. of Hydrobiology, Chinese Acad. of Sci., Wuhan, China)

The Indo-Pacific humpback dolphins (*Sousa chinensis*) are widely distributed in the coastal waters of the tropical and sub-tropical ocean in Asia, including southeast China. Conservation of the humpback dolphins in Chinese waters has been on the agenda of local scientific and conservation communities since the 1980s, despite little research including bioacoustics research has been conducted. Our recent bioacoustics studies indicated that the biosonar clicks of the wild humpback dolphins were short-duration, broadband, ultrasonic pulses, similar to those produced by other whistling dolphins of similar body size. However, their click source levels with an average of around 190 dB re: 1μ Pa in peak-to-peak, appear to be much lower than those of other whistling dolphins. Hearing sensitive frequency range of the humpback dolphins is generally higher than 5 kHz and lower than 120 kHz with possible age-related hearing loss for old dolphins. The humpback dolphin could therefore be characterized as a mid-frequency cetacean, which operates sounds in mid- to high-frequency range. Any sufficiently intense sounds with mid- to high-frequency components could have deleterious effects on the humpback dolphins through interference on animals' behavior and with the animals' ability to detect signals from conspecifics, and echoes of echolocation clicks.

9:25

2aABa5. Acoustic response of Indo-Pacific humpback dolphins to the variability of marine soundscape. Tzu-Hao H. Lin (Res. Ctr. for Information Technol. Innovation, Academia Sinica, No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, schonkopf@gmail.com), Chih-Kai Yang, Lien-Siang Chou (Inst. of Ecology and Evolutionary Biology, National Taiwan Univ., Taipei, Taiwan), Shih-Hau Fang (Dept. of Elec. Eng., Yuan Ze Univ., Taoyuan, Taiwan), and Yu Tsao (Res. Ctr. for Information Technol. Innovation, Academia Sinica, Taipei, Taiwan)

Marine mammals can adjust their vocal behaviors when they encounter anthropogenic noise. The acoustic divergence among different populations has also been considered as the effect of ambient noise. The recent studies discover that the marine soundscape is highly dynamic; however, it remains unclear how marine mammals alter their vocal behaviors under various acoustic environments. In this study, autonomous sound recorders were deployed in western Taiwan waters between 2012 and 2015. Soundscape scenes were unsupervised classified according to acoustic features measured in each 5 min interval. Non-negative matrix factorization was used to separate different scenes and to inverse the temporal occurrence of each soundscape scene. Echolocation clicks and whistles of Indo-Pacific humpback dolphins, which represent the only marine mammal species occurred in the study area, were automatically detected and analyzed. The preliminary result indicates the soundscape scenes dominated by biological sounds are correlated with the acoustic detection rate of humpback dolphins. Besides, the dolphin whistles are much complex when the prey associated scene is prominent in the local soundscape. In the future, the soundscape information may be used to predict the occurrence and habitat use of marine mammals.

Contributed Papers

9:45

2aABa6. Navigating the soundscape: The role of acoustic cues in the settlement behavior of larval reef fishes. Andria K. Salas (Integrative Biology, Univ. of Texas at Austin, Austin, TX), Preston S. Wilson, Megan S. Ballard (Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 1 University Station C2200, Austin, TX 78713, pswilson@mail.utexas.edu), Andrew H. Altieri (Smithsonian Tropical Res. Inst., Panama City, Panama), and Timothy H. Keitt (Integrative Biology, Univ. of Texas at Austin, Austin, TX)

Nearly all coral reef fishes pass through a pelagic larval stage that concludes when they locate and select appropriate reef habitat. At least some species use the reef soundscape—the collection of sounds produced by reef-dwelling organisms—to achieve this task that is necessary for survival. Since larval fishes in the open ocean are inherently difficult to study, investigating their acoustic environment can allow predictions about their behavior in response to these cues. Further, we can predict the roles of different frequency bands in aiding long-distance navigation or short-distance habitat selection. We recorded the soundscapes at four reefs in Caribbean Panama for six weeks to inform the identity and variability of the most common sounds. We next predict the frequency-dependent transmission loss using an acoustic propagation model calibrated with acoustic transects. Including knowledge of source sound levels allows us to elucidate the sound fields created by low frequency fish and high frequency invertebrate sounds originating at the reef. We find that larval fishes are likely to encounter highly complex acoustic environments as a result of frequency-dependent acoustic structure in the water column coupled with temporal and spatial variation in the sources of the potential cues.

10:00–10:15 Break

10:15

2aABa7. Macro- and micro-scale spatial variation in the acoustic activity of snapping shrimp on coral reefs in the Pacific. Eden Zang, Marc Lammers (Oceanwide Sci. Inst., 3620 Baldwin Ave, Ste. 206-B, Makawao, HI 96768, ezang@oceanwidescience.org), Max Kaplan, T. A. Mooney (Woods Hole Oceanographic Inst., Woods Hole, MA), Pollyanna Fisher-Pool, and Russell E. Brainard (NOAA Fisheries, Honolulu, HI)

Coral reef soundscapes are increasingly becoming recognized as critical factors in the study of reef dynamics, from the role they play in larval recruitment to the assessment of coral reef biodiversity and ecosystem stability. Snapping shrimp produce the predominant source of sound on most coral reefs at frequencies between 2 and 20 kHz. Their activity is influenced by a variety of abiotic factors. As such, coral reef soundscapes are prone to considerable flux and variation. However, this variation is still poorly understood on a variety of spatial and temporal scales, making it difficult to draw meaningful comparisons between the soundscapes of different reefs. We report on an effort to quantify the acoustic activity of snapping shrimp across 12 coral reef sites in the Pacific Ocean separated by distances ranging from hundreds of meters to thousands of kilometers, including reefs across the Hawaiian archipelago, the Northern Mariana Islands, and American

Samoa. We use data obtained from long-term, bottom-moored acoustic recorders to document the variability in snapping shrimp activity observed on multiple temporal scales and examine factors correlated with this variability at each location.

10:30

2aABa8. A new baleen whale call recorded in the Mariana Trench Marine National Monument. Sharon L. Nieukirk, Selene Fregosi, David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, sharon.nieukirk@oregonstate.edu), and Holger Klinck (BioAcoust. Res. Program, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY)

In fall 2014 and spring 2015, passive acoustic data were collected via autonomous gliders east of Guam in an area that included the Mariana Trench Marine National Monument. A short (2-4 s), complex, unknown sound was recorded. The call began with a brief 0.4 s long tone centered at 60 Hz, followed by a 2 s low-frequency moan that sweep from 44—30Hz. The moan appeared to have both amplitude modulation and strong harmonics. The moan was followed by two ~60-150 Hz upsweeps lasting 0.5s that often had diffuse energy up to 1000 Hz. The call ended with metallic-sounding upsweeps with most energy between 700 and 800 Hz, but up to 7.5 kHz. These sounds were recorded regularly during both fall (326 calls during 38 glider dives) and spring (110 calls during 16 glider dives). Many components were not visible in low signal to noise ratio calls. Calls were typically 5-6 minutes apart and often occurred in long sequences. Aurally, the sound is quite unusual and most resembles the minke whale "Star Wars" call. We propose this sound is biological and is likely produced by a baleen whale.

10:45

2aABa9. Passive acoustic monitoring of Pygmy blue whales *(Balaenoptera musculus brevicauda)* in Southern Sri Lanka. Annukka Pekkarinen (Maritime Environ. Res. Group, World Maritime Univ., PO Box 500, Malmö 20124, Sweden, ap@wmu.se), Yeelen Olive (VisioTerra, Champs Sur Marne, Paris, France), and Marianne H. Rasmussen (Univ. of Iceland, Husavik, Iceland)

Two ecological acoustic recorders were deployed in Southern Sri Lanka to study the presence of the assumed resident population of pygmy blue whales. A major shipping lane runs through this feeding area of the whales and it has been recognized by the International Whaling Commission as well as International Maritime Organization as a risk area for ship strikes. To implement any management practices to mitigate ship strikes, a solid scientific basis is needed. Such data must include information on migration and movement patterns of the whale population. Therefore, combining the passive acoustic monitoring data with existing visual transect survey results gives further insight to choosing suitable management practices to address the issue. Passive acoustic monitoring data indicate that at least part of the population is indeed resident in the area throughout the year but there is both seasonal and inter-annual variation in the acoustic activity of the whales. The acoustic data were further combined with data from oceanographic models and the results indicate there is no single parameter explaining the variation, although the acoustic activity has weak correlation with rainfall patterns. The pygmy blue whale call detections also appear to increase during periods of stronger upwelling in Southern Sri Lanka.

11:00

2aABa10. Spatiotemporal variation in habitat utilization of humpback dolphins *(Sousa chinensis)* potentially affected by freshwater discharge. Chia-Yun Lee, Tzu-Hao Lin (Inst. of Ecology and Evolutionary Biology, National Taiwan Univ., Taiwan, No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, janiceli0918@gmail.com), Tomonari Akamatsu (Fisheries Res. Agency, National Res. Inst. of Fisheries Sci., Kanagawa, Japan), and Lien-Siang Chou (Inst. of Ecology and Evolutionary Biology, National Taiwan Univ., Taiwan, Taipei, Taiwan)

Dynamics of habitat utilization of marine mammals are important for the conservation management of coastal ecosystems. Previous studies have shown that seasonal changes of humpback dolphin distribution are associated with seasonal changes of upstream rainfall. However, temporal variations of dolphin behavior at an estuarine habitat remain unclear. In this study, echolocation clicks of humpback dolphins were recorded at the Xinhuwei River Estuary, western Taiwan, between July 2009 and November 2015. Sound source bearing angles to identify different sound sources, intensity, and inter-click intervals (ICIs) as an index of the sensing distance were measured to investigate the sensing and movement of dolphin groups. Humpback dolphins searched for shorter sensing distances and moved back and forth near the estuary during spring and summer, which suggests foraging-related behaviors were much frequently observed during wet seasons. However, humpback dolphins changed to focus on longer detection ranges (ICIs changed from 40 to 50 ms to 50 to 70 ms) and move in straight forward when upstream rainfall increased during this period. The present result indicated the dynamics of habitat utilization of humpback dolphins at an estuary is likely driven by the distribution and abundance of prey resource which is influenced by the temporal change of river runoff.

11:15

2aABa11. Source level of fin whale calls from the Equatorial Pacific Ocean. Jennifer L. Miksis-Olds (School of Marine Sci. & Ocean Eng., Univ. of New Hampshire, PO Box 30, Mailstop 3510D, State College, PA 16804, j.miksisolds@unh.edu)

Knowledge of source levels is essential for determining the range of effective fin whale communication and for estimating fin whale density from passive acoustic recordings. Source level calculations were performed on vocalizations automatically detected from spectrograms of Comprehensive Nuclear Test-Ban Treaty Organization data recorded at Wake Island in the Equatorial Pacific Ocean. Received levels were calculated, and transmission loss (TL) was determined using a season- specific OASIS Peregrine parabolic equation model for a 20 Hz signal. The model incorporated the location of the sensor in the deep sound channel, bathymetry of the area, and local sound speed profiles. TL was modeled for 360 bearings with a 1^{o} resolution. TL values between the sensor and source were found for individual vocalizations using ranges and bearings calculated through hyperbolic localization. The exact whale depths were unknown but assumed to be 50 m. Over 7500 localizations of the fin whale 20 Hz call were identified from 2007 to 2009 with an average source level of 184.8 dB +/- 8.6 dB. When the 7500 calls were subsampled at 6 hour intervals to reduce the bias of loud, persistent singers, the average source level was 168.5 dB +/- 4.4 dB (n = 174). [Work supported by ONR.]

Session 2aABb

Animal Bioacoustics and Signal Processing in Acoustics: Anthropogenic Transient Noise Sound Field and Its Effects on Animal Communication I

Shane Guan, Cochair
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Satoko Kimura, Cochair
Field Science Education and Research Center, Kyoto Univ., Kyoto, Japan

Chair's Introduction—8:45

Invited Papers

8:50

2aABb1. Quantitative measurements of seismic airgun reverberation in the shallow-water Beaufort Sea. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Melania Guerra (APL, Univ. of Washington, Seattle, WA), Susanna Blackwell (Greeneridge Sci., Santa Barbara, CA), and A. Michael Macrander (Shell Exploration and Production Co., Anchorage, AK)

Shallow-water airgun survey activities off the North Slope of Alaska generate impulsive sounds that have been the focus of much regulatory attention. Reverberation from repetitive airgun shots, however, also increases the background noise levels, which can decrease the detection range of nearby passive acoustic monitoring (PAM) systems and potentially mask communication signals between animals. Typical acoustic metrics for impulsive signals provide no quantitative information about reverberation. Two metrics are suggested here for quantifying reverberation: a "minimum level" metric that measures reverberation levels between airgun pulse arrivals, and a "reverberation metric" that estimates the relative magnitude of reverberation vs. expected ambient levels in the hypothetical absence of airgun activity, using satellite-measured wind data. The metrics were applied to acoustic data measured by autonomous recorders in the Alaskan Beaufort Sea in 2008. They demonstrate how a 3000 cu. inch seismic survey in 50 m deep water can increase the background noise over natural ambient levels by 30-45 dB within 1 km of the activity, by 10-25 dB within 15 km of the activity, and by a few dB at 128 km range. Shallow-water reverberation can thus substantially reduce the performance of PAM systems several kilometers of shallow-water seismic surveys. Other impulsive activities such as pile driving may face similar issues.

9:10

2aABb2. Inter-pulse sound field from a marine seismic survey in the Arctic and its potential effects on marine mammal acoustic masking. Shane Guan (Office of Protected Resources, National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov) and Joseph F. Vignola (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC)

It has been documented that reverberant fields from marine seismic surveys elevate sound levels, and thus has the potential to mask acoustic signals that are important to marine mammals. In this study, we investigated one-third-octave (OTO) bands of airgun inter-pulse sound fields from a shallow water ($<$15 m) seismic survey in the Beaufort Sea. The results show that at close distances less than 3 km, peak energy of the OTO band levels were concentrated around 80-110 Hz and the median sound levels were 40-50 dB above ambient. The higher frequency components ($>$3 kHz) of inter-pulse sound field at these close distances ranged 5-20 dB above ambient. At distances beyond 3 km, the received acoustic energy becomes more uniform up to 1000 Hz. The median OTO band levels at these distances were approximately 10-20 dB above ambient. For higher frequencies, sound levels at these distances were approximately the same as ambient. These results imply that low-frequency bowhead whales could be affected by acoustic masking from seismic airgun inter-pulse sound fields, especially at relatively close distances. However, potential acoustic masking effects to mid-frequency beluga whale echolocation signals are expected to be less profound.

9:30

2aABb3. Long range airgun inter-pulse noise field in offshore northern Australian waters. Craig McPherson (JASCO Appl. Sci. (Australia) Pty, Ltd., Unit 4, 61-63 Steel St., Capalaba, QLD 4157, Australia, craig.mcpherson@jasco.com), Bruce Martin (JASCO Appl. Sci. (Canada) Ltd., Dartmouth, NS, Canada), Michael Wood (JASCO Appl. Sci. (UK) Ltd., Droxford, Hampshire, United Kingdom), and Alexander MacGillivray (JASCO Appl. Sci. (Canada) Ltd., Victoria, BC, Canada)

Elevated ambient noise levels have been shown to affect marine fauna communication. The effects due to increased noise levels from the multipath propagation and reverberation of airgun pulses over large distances is a relatively new area of interest. This research examines a distant seismic survey opportunistically recorded at two locations during a long-term monitoring program in the Timor Sea, Australia. During the analyzed period, the survey was between 168 and 235 km from the Autonomous Multichannel Acoustic Recorders (AMAR). The noise levels between airgun pulses were examined using an incremental computation method based on calculating the root-mean-square sound pressure level in 125 ms sub-intervals. These were compared to ambient noise levels from the month prior to commencement of the seismic survey and during line turns. The received pulse levels and length differences, and therefore inter-pulse noise field, between the recorders were greater than expected given the proportional distance to source difference. The ambient soundscape included biological contributors such as Omura's whale, fish and benthic crustaceans. Propagation modeling was conducted to examine the effect of the propagation path between source and receiver on the received signals.

9:50

2aABb4. Influence of the ocean-bottom structure in the long-distance propagation by seismic survey source. Toshio Tsuchiya (Japan Agency for Marine-Earth Sci. and Technology/Tokyo Univ. of Marine Sci. and Technol., 2-15 Natsushimacho, Yokosuka, Kanagawa 2370061, Japan, tutiyat@gmail.com), Shinpei Gotoh, Yukino Hirai, and Etsuro Shimizu (Tokyo Univ. of Marine Sci. and Technol., Koto-ku, Tokyo, Japan)

A source of Multi-Channel Seismic survey system (MCS) is the air-gun array which generates a high level and short pulse sound. However, there was anxiety which has an influence on behavior of a marine mammal by very high acoustic pressure. Therefore, JAMSTEC have decided guidance of regulation along a guideline of NOAA (TTS) in safe purpose of a marine mammal. We'll hope for a MCS investigation by a new JAMSTEC R/V "KAIMEI" (5,500GT) in the Hawaii Islands offing jointly with an U.S. research institute by geological interest in the future. So, long-distance propagation (about 100 km) from air gun source was calculated by a Parabolic Equation simulation for safety ensuring of marine mammals. The simulation results are as follows: 1) When the ocean-bottom of the long-distance transmission path is basalt, for a reflected pulses to pile up from the bottom in various courses, a received pulse width becomes very long. 2) The received level was lower than the TTS level in the watch possible distance from the marine mammal watch room installed in R/V "KAIMEI." 3) However, when the low restriction level (the behavior obstruction level) was applied for marine mammal protection in the future, we have to lower the air gun transmission level substantially.

10:10–10:25 Break

10:25

2aABb5. An attempt to estimate ship noise effect on humpback whales in Japan. Tomonari Akamatsu (National Res. Inst. of Fisheries Sci., Fisheries Res. Agency, 7620-7, Hasaki, Kamisu, Ibaraki 314-0408, Japan, akamatsu@affrc.go.jp), Ryosuke Okamoto (Ogasawara Whale Watching Assoc., Ogasawara, Japan), Kyoichi Mori (Teikyo Univ. of Sci., Uenohara, Yamanashi, Japan), Yoko Mitani, Kouki Tsujii (Hokkaido Univ., Hakodate, Japan), Toshio Tsuchiya (JAMSTEC, Yokosuka, Japan), Takahiro Kijima (Ministry of Land, Infrastructure, Transport and Tourism, Tokyo, Japan), and Naoya Umeda (Osaka Univ., Osaka, Japan)

Ocean noise pollution is getting to be a major issue for the environmental assessment of maritime transportation and engineering. Evidences to estimate the effect of noise on marine creatures are urgently required. A Japanese team consist of government agency, universities, and research institutions launched a new project to observe possible effects of ship noise on humpback whales in Ogasawara archipelago. Radiated noise from a ship was measured according to the ISO standard protocol in deep water. The position and operational conditions of the ship during the daily voyages were precisely monitored onboard. Sound field within 10 km from the voyage route was calculated by numerical simulation. In parallel, land-based visual observers tracked humpback whales by a theodolite to locate surface position so that the exposure level at the animal can be estimated. Two autonomous stereo recording systems were deployed in the focal area to monitor the phonation behavior of singing whales simultaneously. As the first year result, temporal termination of song sequence was occasionally observed although avoidance behavior from the ship was not quite clear. This project would[NU1] last two more years to provide a scientific evidence of the minimum exposure level to elicit behavioral reaction of the whales.

10:45

2aABb6. Effects of anthropogenic noise on migrating bowhead whales in the Beaufort Sea. Katherine H. Kim, Susanna B. Blackwell (Greeneridge Sci., Inc., Greeneridge Sci., Inc., 90 Arnold Pl, Ste. D, Santa Barbara, CA 93117, khkim@greeneridge.com), and Aaron M. Thode (Scripps Inst. of Oceanogr., La Jolla, CA)

Greeneridge Sciences, Inc. (Greeneridge), has conducted multi-year, underwater, acoustic measurements at two study areas offshore of the North Slope of Alaska in the Beaufort Sea. The primary objective of these studies was to measure industrial sounds, e.g., those produced by oil production, seismic exploration, and drilling activities, and to assess their potential effects on the behavior of bowhead whales during their annual fall migration. To meet this objective, during open-water season every year from 2001 to 2016 (BP, Hilcorp) and 2007 to 2014 (Shell), Greeneridge deployed passive acoustic recorders equipped with directional sensors (DASARs) along the continental shelf. Over the course of these studies, millions of bowhead calls were localized, while ambient noise and various types of anthropogenic noise, such as tonal sounds associated with machinery and vessels and seismic airgun pulses, were quantified. The localization capabilities of the DASARs, large numbers of observations, and multi-year time series measurements together permitted, with high statistical power, quantitative evaluation of the effects of anthropogenic noise on bowhead behavior. Here, we'll review some of our major findings, including the apparent displacement of calling bowhead whales and changes in their calling rate in response to industrial

activities, consistency in their call source levels, and a shift in the frequency content of their call repertoire. [Work supported by BP Exploration Alaska, Shell Alaska Venture, and Hilcorp Alaska.]

11:05

2aABb7. Predicting masking by impulsive noise sources: An experimental evaluation. Jillian M. Sills (Long Marine Lab., Inst. of Marine Sci., Univ. of California at Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95050, jmsills@ucsc.edu), Brandon L. Southall (SEA, Inc., Aptos, CA), and Colleen Reichmuth (Long Marine Lab., Inst. of Marine Sci., Univ. of California at Santa Cruz, Santa Cruz, CA)

Impulsive noise sources pose significant challenges in terms of predicting auditory masking using conventional methods. Aside from their obvious time-varying structure, another complicating factor is the influence of propagation on the spectral and temporal characteristics of the noise, especially in water where such sounds may travel considerable distances. To address this, we developed a psychophysical paradigm to quantify masking in Arctic seals during different time intervals of seismic maskers, which were recorded either close to (1 km) or far from (30 km) an operational air gun array. Signal-to-noise ratios at threshold (50% correct detection rate) were obtained behaviorally for trained seals, and were compared to conventional masked threshold predictions based on average noise levels and critical ratio measurements for the same individuals. The experimental data showed that masking predictions were poorest in the time intervals where noise exhibited the greatest amplitude variation. These findings provide insight into whether and how the dynamic sound field surrounding seismic surveys constrains the ability of seals to detect relevant signals, and show how predictive models of masking for transient noise sources can be improved by incorporating time-based analyses of signals and noise. [Work supported by OGP-JIP.]

11:25

2aABb8. Noise field characterization in the habitat of the East Taiwan Strait Indo-Pacific Humpback Dolphin during the pile driving activity of demonstration offshore wind farm. Chi-Fang Chen (Dept. of Eng. Sci. and Ocean Eng. / Ocean Technol. Res. Ctr., National Taiwan Univ., No. 1 Roosevelt Rd., Sec.#4, Taipei 106, Taiwan, chifang@ntu.edu.tw), Shane Guan (NOAA/NMFS, Office of Protected Resources, Silver Spring, MD), Lien-Sian Chou (Inst. of Ecology and Evolutionary Biology, National Taiwan Univ., Taipei, Taiwan), Ruey Chang Wei (Inst. of Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan), William W. Hu, Jeff C. Wu, Nai-Chang Chen, Wei-Shien Hwang (Dept. of Eng. Sci. and Ocean Eng. / Ocean Technol. Res. Ctr., National Taiwan Univ., Taipei, Taiwan), Sheng-Fong Lin (GEL, Industrial Technol. Res. Inst., Hsin-Chu, Taiwan), and Derrick Lin (Swancor Renewable Energy Co., Ltd., Taipei, Taiwan)

The Eastern Taiwan Strait (ETS) population of Indo-Pacific humpback dolphin (*Sousa chinensis*) is listed critically endangered in the Red List of Threatened Species by the International Union for Conservation of Nature due to its small population size and narrow distribution. The humpback dolphin habitats off the coast of Miaoli and Changhua are sites selected for future wind farms, therefore, the noise impact of pile driving on this critically endangered population is expected to be serious. This paper presents works done in (1) characterizing the sound field during the test pile driving and associated activities in the humpback dolphin habitat; (2) identifying dominant anthropogenic noise sources of the dolphin habitat during the construction of demonstration wind turbines and associated activities; and (3) examining the implications of the sound field from wind turbine construction and associated activities in relation to humpback dolphins' hearing and communication. The results from the study can provide critical information and conservation recommendations for an environmental impact analysis for the full scale wind farm construction in 2017.

Session 2aAO

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

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

Invited Papers

8:30

2aAO1. A critical review of geoacoustic inversion: What does it really tell us about the ocean bottom? N. Ross Chapman (School Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3065, Victoria, BC V8P 5C2, Canada, chapman@uvic.ca)

Estimation of parameters of geoacoustic models from acoustic field data has been a central theme in acoustical oceanography over the past three decades. Highly efficient numerical techniques based on Bayesian inference have been developed that provide estimates of geoacoustic model parameters and their uncertainties. However, the methods are model-based, requiring accurate knowledge of the acoustic propagation conditions in the ocean to carry out the inversion. More recent research has revealed fundamental limitations of model-based inversion methods in conditions of unknown temporal and spatial variations in the water. In addition, the inversions can generate only effective models of the true structure of the ocean bottom, which are generally highly variable over relatively small spatial scales. There are other questions about the theory for sound propagation in porous sediment media that raise doubt about the validity of inversion results. In most inversions, a visco-elastic theory is used, but is this correct? This paper reviews successes and failures of geoacoustic inversion to understand the limitation of model-based methods. Research directions are suggested in conclusion that show promise for development of new approaches. [Work supported by ONR.]

8:50

2aAO2. Physics-based inversion of multi-beam echo sounder data for seafloor properties. Darrell Jackson, Brian T. Hefner, and Anatoliy Ivakin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, drj@apl.washington.edu)

This is an extension of work on physics-based seafloor inversion for seafloor properties, pioneered by Pouliquen and Lurton and by Sternlicht and De Moustier. This approach fits model time series for backscattered intensity to data, with the best-fit model parameters constituting the inversion output. In the present case, use of multi-beam sonar provides a large number of time series and consequent stronger constraints on model parameters. The method has been tested at two experiment sites, one with sandy and shelly areas and one with a thin layer of mud over sand. Ground truth data were gathered on seafloor roughness using a laser line scanner, on layering using a conductivity probe, on grain-size distribution using diver cores, and on sediment sound-speed and attenuation using acoustic probes. The inversion process involves three stages. First, a sonar-equation model is used to generate echo intensity time series including scattering by both seafloor roughness and volume heterogeneity. Model-data fit provide estimates of acoustic attenuation, volume scattering strength, and interface scattering strength. Next, physics-based models are fitted to the interface and volume scattering strengths, and finally, regression relations are used to provide a set of geoacoustic parameters sufficient to constrain standard reverberation simulations. [Work supported by SERDP.]

Contributed Papers

9:10

2aAO3. Acoustic remote sensing of double-diffusive instabilities and intrusions in ocean frontal zones. Timothy F. Duda and Andone C. Lavery (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu)

Small-wavelength sound has been observed to scatter from turbulent microstructure and plankton in the ocean. Additional scattering from salt fingers and diffusive-convective flows that result from double diffusive instabilities (DDI) is also theoretically possible in this same wavelength band, and has been measured in a laboratory setting, though with artificially high gradient parameters. If measurable, small-wavelength backscatter would enable remote sensing of water mass interleaving and mixing from intrusive flows at fronts, which spawn DDI. The factors controlling the visibility of echoes from DDI features is the strength of the features in terms of density and sound-speed anomaly, and the level of masking return from plankton, which may merely interfere or act as passive tracers of these features. Microstructure data collected in double-diffusive instability features are used to compute the expected monostatic backscatter signals. These results suggest that DDI microstructure, from both salt fingering and convection, would be acoustically observable in areas with low to average plankton. The results also suggest that density interfaces (layering structures) associated with DDI would be visible. Acoustic backscatter data show echoes from both the interfaces and DDI microstructure in a frontal zone

across a range of wavelengths. [Work supported by the Office of Naval Research.]

9:25

2aAO4. Correlation among multipaths for long rang propagation. Arthur B. Baggeroer (Mech. and Elec. Eng., Massachusetts Inst. of Technol., Rm. 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu) and John A. Colosi (Dept. of Oceanogr., Navy Postgraduate School, Monterey, CA)

The multipath arrivals in ocean acoustic tomography are the important observations for the inversion to a sound speed profile and then to temperature. They are also implicit in any matched field processing beamforming. While the covariance of ray path travel times has been examined (Flatte and Stoughton, JGR 91, C6), the correlations among the waveforms of these multipaths has never been determined. There are two conflicting hypotheses: i) the paths radiate from a source, so they must be correlated. ii) Alternatively, the small scale ocean inhomogeneities randomize the paths since they traverse different ocean masses. At short ranges and low frequencies, the paths are correlated, whereas at long ranges and high frequencies, they are uncorrelated. Reality is somewhere in between. We have a power law medium which at large scales correlates paths but decorrelates them at small scales. The key quantity distinguishing the scales is the Fresnel zone extent. We analyze the correlation from two perspectives. A theoretical one based on path integral formulations and a experimental one on numerical Monte Carlo simulations on the massively parallel MIT Lincoln Laboratory LLGrid using adaptive beamforming for path resolution.

9:40

2aAO5. Reconstruction of the dynamic surface roughness with acoustic imaging technique. Giulio Dolcetti and Anton Krynkin (Mech. Eng., Univ. of Sheffield, Mappin St., Sheffield S1 3JD, United Kingdom, gdolcetti1@sheffield.ac.uk)

We present a novel technique which recovers the three dimensional profile of the free surface based on the scattering of the narrow-band ultrasonic waves recorded by a two-dimensional array of receivers. The method is based on the inversion of the discretized Kirchhoff integral. The inverse technique implements the Tikhonov regularisation and the generalised cross validation technique in order to obtain solution to an under-determined system of equations. Using airborne ultrasound, the technique has been validated experimentally to reconstruct a two-dimensional composite static surface and numerically to recover the three-dimensional dynamic surface. The application to the typical three dimensional patterns found on the free surface of shallow turbulent flows is illustrated and discussed. The method has important applications for the remote monitoring of natural rivers, where recent studies confirm the direct link between the fundamental hydraulic properties of the flow and the dynamic behavior of the free surface. The potential significance in the oceanic environment is also discussed.

9:55–10:10 Break

10:10

2aAO6. Detection of long term trends in underwater noise levels. Claire F. Powell (Cefas, Cefas, Pakefield Rd., Lowestoft NR33 0HT, United Kingdom, claire.powell@cefas.co.uk) and Nathan D. Merchant (Cefas, Lowestoft, Suffolk, United Kingdom)

The risk of adverse impact to marine ecosystems from underwater noise pollution is increasingly recognised by scientists, policymakers, and wider society. Deep water measurements from the Northeast Pacific indicate that ocean noise has increased substantially over recent decades. Policymakers are now considering establishing noise monitoring programs to determine noise levels and trends in their waters. However, the ability of noise monitoring to detect statistically significant trends is a function of the temporal extent, variance, and autocorrelation of the time series. This has implications for the feasibility of evaluating quantitative policy targets within prescribed time frames and hence should inform the formulation of such targets. The present work demonstrates that methods developed in other environmental science disciplines (e.g., atmospheric temperature measurement) to design long-term monitoring networks and assess their statistical power can be applied to noise monitoring programmes. Example datasets are used to show the application of these methods both to assess the significance of long-term trends in ambient noise, and the required monitoring period to detect a given magnitude of trend (e.g., 3 dB per decade). The implications for the design of noise monitoring networks and target setting for policy purposes are then discussed.

10:25

2aAO7. Lateral diffraction of underwater sounds between Japan and Chile: experimental results and modeling. Kevin D. Heaney (OASIS, Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com), Tomoaki Yamada (Univ. of Tokyo, Tokyo, Japan), Mario Zampolli, and Georgios Haralabus (CTBTO, Vienna, Austria)

One hundred shallow underwater explosions on a 300-km profile along an isoline of 500 m sea depth above a slope zone of off the coast of Japan are received at a water-column hydrophone triplet with 2-km intervals in Chile. The over 16,000-km propagations from the northwest to the southeast Pacific Ocean are classified into two groups. One is from the southern explosions. The data show high sound pressure levels and normal travel times. On the other hand, data from the northern ones reveal 6.5 dB lower sound pressure levels and 6.6 s travel time delays on average. Some islands and seamounts in the central Pacific are located on the halfway from the northern explosions. They affect received sound pressure levels and arrival times, but do not block completely, based on the observation. Modeling of this propagation path, using in-plane 2D modeling, shows nearly complete blockage of the sound by the Northern Hawaiian Island chain. Three dimensional parabolic equation modeling shows that this shadow is filled in from diffraction caused by the Midway Islands. Modeling results quantitatively match the measurements for both energy level and travel time.

10:40–11:10 Panel Discussion

Session 2aBAa

Biomedical Acoustics and Physical Acoustics: Cavitation in Therapeutic Ultrasound II: 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
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Zhen Xu, Cochair
Biomedical Engineering, University of Michigan, 2200 Bonisteel Blvd., Rm 1107 Gerstacker Bldg., Ann Arbor, MI 48109

Contributed Papers

7:50

2aBAa1. Optimisation of ultrasound exposure parameters for extravasation and drug delivery using submicron cavitation nuclei. Christophoros Mannaris, Megan Grundy (Inst. of Biomedical Eng., Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg. (off Roosevelt Drive), Oxford OX3 7DQ, United Kingdom, christophoros.mannaris@eng.ox.ac.uk), Margaret Duffy, Len W. Seymour (Clinical Pharmacology, Dept. of Oncology, Univ. of Oxford, Oxford, United Kingdom), Robert Carlisle, and Constantin C. Coussios (Inst. of Biomedical Eng., Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Sub-micron cavitation nuclei with the ability to passively extravasate through the leaky tumor vasculature have been shown to enhance both extravasation and penetration of pharmaceuticals in tumors. The aim of the present work is to optimize the acoustic parameters and maximize drug delivery mediated by cavitational microstreaming. Either lipid-shell microbubbles, or gas-stabilizing polymeric submicron cups of mean diameter 480 nm are co-administered with a model drug through a flow channel formed in agarose gel. The effect of ultrasound frequency, pressure amplitude, pulse duration, duty cycle, and pulse repetition frequency on the delivery of the drug is first evaluated, using both a single-layer and a dual-layer flow phantom that better represents the leaky tumor vasculature and dense extracellular matrix. The optimal parameters at 0.5 MHz and 1.6 MHz are then selected and used to deliver an oncolytic virus, genetically modified to express the RFP reporter gene, thus allowing the quantification of the delivery, infection, and subsequent spreading of the virus under fluorescence microscopy. We hypothesize that while inertial cavitation within the vasculature leads to extravasation, enhanced penetration is achieved by subsequent cavitation events of the sub-micron nuclei that passively extravasate and propel the drug deeper into the tumor.

8:05

2aBAa2. Giant vesicles as cell models to quantify bio-effects in ultrasound mediated drug delivery. Valerio Pereno, Dario Carugo, and Eleanor P. Stride (Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

The biophysical mechanisms that underpin the interaction between microbubbles and cells in the context of ultrasound-mediated drug delivery are still poorly understood. To aid the identification of these mechanisms, giant unilamellar vesicles (GUVs) were used as cell models to quantify changes in membrane properties as a result of the interaction with ultrasound (1 MHz, 150 kPa, and 60 s continuous wave) and phospholipid-shelled microbubbles (DSPC-PEG40S 9:1 molar ratio), either alone or in combination. The spatial quantification of the vesicle lipid order was performed via spectral microscope imaging, by measuring the contours of generalized polarisation (GP) from the emission spectrum of c-Laurdan, a polarity-sensitive dye. Preliminary data show synergistic mechanical and chemical effects on membranes: ultrasound exposure and shear flow alone generally decrease the vesicle lipid packing, while exposures involving microbubbles reveal contrasting effects depending on the initial vesicle composition and acoustic regime. Results from the present mechanistic study provide an insight into the mechanisms of microbubble-membrane interactions, potentially benefitting the design of effective and predictable microbubble-based ultrasound treatments.

8:20

2aBAa3. Low intensity pulsed ultrasound and lipid-coated microbubbles enhance chondrogenesis of human mesenchymal stem cells in 3D bioprinted scaffolds. Mitra Aliabouzar, Lijie G. Zhang, and Kausik Sarkar (George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu)

Annually, over 6 million people visit hospitals in the United States for issues arising from cartilage damages due to osteoarthritis, acute trauma, rheumatoid arthritis, and sports injury. Articular cartilage is a tissue notoriously hard to regenerate. The objective of this study is to investigate the possibility of facilitating cartilage regeneration using low intensity pulsed ultrasound (LIPUS) stimulation and microbubbles. We have investigated effects of lipid-coated microbubbles (MB) prepared in-house along with LIPUS on proliferation and chondrogenic differentiation of human mesenchymal stem cells (hMSCs) in a novel 3D printed poly(ethylene glycol) diacrylate (PEG-DA) hydrogel scaffold. Our results demonstrate for the first time that LIPUS stimulation in the presence of 0.5% (v/v) MB greatly enhances MSC proliferation for up to 40% after 5 days of treatment. This value is only 18% when excited with LIPUS alone. Furthermore, we optimized acoustic parameters such as excitation intensity, frequency and pulse repetition period for chondrogenic differentiation studies. Synthesis of type II collagen and GAG, which are two key cartilage biomarkers, increased 78% and 17%, respectively, in the presence of LIPUS and MBs, when compared to the controls.

8:35

2aBAa4. Ultrasound-stimulated microbubble enhanced low-dose radiation response. Gregory Czarnota (Sunnybrook Health Sceinces Ctr., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, gregory.czarnota@sunnybrook.ca)

We have recently demonstrated that mechanical perturbation of endothelial cells from ultrasound-stimulated microbubbles (USMB) results in enhanced tumor radiosensitivity at low 2 Gy doses of radiation. Our hypothesis is that USMB-based endothelial membrane perturbations produce ceramide via a sphingomyelinase (ASMase) pathway, and act synergistically with radiation to enhance overall tumor response. Here, we investigate the role of the SMase-ceramide pathway on USMB-based endothelial radiosensitization. Experiments were carried out in wild type (C57BL/6) and ASMase knockout mice, implanted with a fibrosarcoma line (MCA-129). Animals were treated with radiation doses varying from 0 to 8 Gy alone, or in combination with ultrasound-stimulated microbubbles. Treatment response was assessed with Doppler ultrasound vascularity index acquired at 3, 24, and 72 hrs using a VEVO770 preclinical ultrasound system. Staining using ISEL, ceramide, and CD31 immunohistochemistry of tumor sections was used to complement results. In contrast to wild type animals, ASMase knockout mice, or wild-type mice receiving S1P, were found to be generally resistant to the anti-vascular effects of radiation and USMB. Minimal cell death and no vascular shutdown was observed following treatments in those experimental groups. Overall conclusions drawn from this work suggest a mechanotransduction-like effect that results in endothelial radiosensitization.

8:50

2aBAa5. Magnetic targeting of oxygen loaded microbubbles for sonodynamic therapy. Estelle Beguin (Univ. of Oxford, Old Rd. Campus Bldg., Oxford OX3 7DQ, United Kingdom), Jason Sheng, Heather Nesbitt (Ulster Univ., Coleraine, United Kingdom), Joshua Owen (Univ. of Oxford, Oxordf, United Kingdom), Anthony McHale, John Callan (Ulster Univ., Coleraine, United Kingdom), and Eleanor P. Stride (Univ. of Oxford, Oxford, United Kingdom, eleanor.stride@eng.ox.ac.uk)

Previous work has demonstrated that the efficacy of sonodynamic therapy (SDT) can be significantly enhanced by conjugating SDT drugs to oxygen filled microbubbles. For eventual clinical use however, achieving adequate microbubble stability and targeting represents a considerable challenge. This study investigated whether functionalizing microbubbles with magnetic nanoparticles could both improve their longevity and enable localization of the microbubbles to a target site. Phospholipid coated microbubbles containing oxygen and 50 nm spherical magnetite particles were produced by sonication and conjugated to the SDT drug Rose Bengal via an Avidin-Biotin linker. The addition of the nanoparticles was found to substantially enhance the stability of the microbubbles to changes in size and concentration. Orthotopic tumors were induced in BALB/c SCID mice using the BxPc-3 human pancreatic cell line and exposed to ultrasound for 3.5 min (3.5 Wcm^{-2} at 1 MHz center frequency, 100 Hz pulse repetition frequency, and 30% duty cycle) following intravenous injection of the microbubbles with or without application of a 0.1 T magnetic field. 38 days postimplantation the tumors treated with both ultrasound and the magnet were 50% smaller than the control tumors and exhibited a threefold increase in active caspase as a marker of apoptosis. There was no statistically significant effect of ultrasound alone.

9:05

2aBAa6. Use of pulse repetition frequency to augment acoustic droplet vaporization in vivo. Robinson Seda, Jonah Harmon (Biomedical Eng., Univ. of Michigan, 2225 Lurie Biomedical Eng., 1101 Beal Ave., Ann Arbor, MI 48105, robseda@umich.edu), Jeffrey B. Fowlkes (Radiology, Univ. of Michigan, Ann Arbor, MI), and Joseph Bull (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Tumor infarction shows promise as a cancer treatment. Gas embolotherapy tries to achieve this by locally occluding blood vessels supplying tumors using bubbles resulting from vaporization of liquid droplets. Sufficient occlusion may be necessary in order to starve tumors to death. The purpose of this study is to increase vaporization of droplets *in vivo* by changing the pulse repetition frequency (PRF) in order to improve the therapy. One milliliter of a solution containing 3×10^7 lipid-coated perfluorocarbon droplets was injected through the tail vein of a rat at a rate of 0.1 mL/min. Acoustic droplet vaporization (ADV) was induced in the feeder vessel of the exposed cremaster muscle using a focused, 7.5 MHz transducer and 3 MPa rarefactional pressure. A pulse length of $1\mu s$ was used, while the PRF was varied from 10 to 1000 Hz. Recordings of ADV showed substantial vaporization in both arterial and venous sides at higher PRFs. Vessel rupture and RBC extravasation were also observed. In other instances, blood vessels experienced a narrowing of the lumen at the site of vaporization (25% reduction), suggesting an ultrasound/droplet/bubble interaction effect. Bubbles resulting from vaporization at high PRF were more likely to occlude small blood vessels, possibly due to coalescing.

9:20

2aBAa7. Enhanced chondrocytes growth in 3-D alginate scaffolds with improved porosity and permeability induced by low intensity pulse ultrasound. Juan Tu (Phys., Inst. of Acoust., Nanjing Univ., #22 Hankou Rd., Nanjing 210093, China, juantu@nju.edu.cn), Xiasheng Guo, Dong Zhang (Phys., Nanjing Univ, Inst. of Acoust., Nanjing, Jiangsu, China), Qingyu Ma, Gepu Guo (Phys., Nanjing Normal Univ., Nanjing, China), and Huan Xu (National Inst. of Metrology, Nanjing, China)

Recent development in applications of new biomaterials and biomedical engineering enable the tissue engineering to become a promising cartilage repair technique. Here, a 3-D alginate scaffold was fabricated by a cross-linked method. Experiments were performed to investigate how the porosity and permeability of the 3-D scaffold, as well as the proliferation rate of seeded cells, were affected by the ultrasound exposure parameters. The scanning electron microscopy and fluorescence imaging were used to examine the micro-structure, porosity, and permeability of the scaffolds, and biochemical analyses were applied to assess the cell growth in the scaffold. The optimum low intensity pulsed ultrasound (LIPU) driving parameters that benefit the enhancement of scaffold porosity and cell proliferation were also explored. The results suggest that, for the scaffold exposed to LIPU, its porosity and permeability could be significantly enhanced by the increasing LIPU amplitude, which might be induced by the microstreaming shear stress generated by ultrasound-driven microbubble oscillations. The assessments of cell proliferation and collagen II expression confirmed that, with appropriately selected LIPUS driving parameters, chondrocytes growth could be effectively promoted in 3-D alginate scaffolds treated by LIPU, because of the improved scaffold porosity and permeability might benefit cell growth space and nutrition supply.

Session 2aBAb

Biomedical Acoustics: High-Frame Rate Imaging with Plane Waves and Diverging Waves

Jeffrey A. Ketterling, Cochair
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Alfred Yu, Cochair
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Hideyuki Hasegawa, Cochair
University of Toyama, 3190 Gofuku, Toyama 9308555, Japan

Chair's Introduction—8:25

Invited Papers

8:30

2aBAb1. High temporal and spatial resolution ultrasonic imaging and application to cardiovascular imaging. Hideyuki Hasegawa (Univ. of Toyama, 3190 Gofuku, Toyama 9308555, Japan, hasegawa@eng.u-toyama.ac.jp)

Recently, high frame rate ultrasound has become used for various imaging scenarios. It creates multiple scan lines at the same time using unfocused transmit beam to reduce the number of transmissions per image frame. Consequently, high frame rate ultrasound can realize a frame rate of over 1 kHz, but the lateral spatial resolution is degraded by using unfocused transmit beams, such as plane and diverging waves. To overcome such a problem, we have developed novel beamforming methods based on adaptive procedures, such as phase coherence imaging and minimum variance beamforming. As a result, high temporal and spatial resolution could be realized simultaneously. The developed high frame rate imaging methods were applied to imaging of cardiovascular dynamics. Blood flow in the carotid artery and cardiac left ventricle were imaged at frame rates of 2500 Hz and 6250 Hz with plane wave and diverging wave transmissions, respectively, and vortices in the carotid bifurcation and left ventricle could be observed. The high frame rate imaging method was also applied to tissue motion imaging. In the carotid arterial wall, pulse wave propagation could be visualized clearly even in a very regional segment of about 10 mm. Also, we could also visualize propagation of the shear wave in the liver, which is spontaneously generated by the beating heart and blood vessels, and its propagation speed could be estimated qualitatively. High frame rate ultrasound would be an indispensable technique for measurement of tissue dynamics.

8:50

2aBAb2. Coherence-based adaptive imaging for high-frame-rate ultrasonic imaging. Pai-Chi Li (Dept. of Elec. Eng., National Taiwan Univ., No.1, Sec. 4, Roosevelt Rd., Taipei 106, Taiwan, paichi@ntu.edu.tw)

Some success has been demonstrated in the studies of adaptive imaging, but these approaches are generally not suitable for high-frame-rate imaging where broad transmit beams are required. This report introduces the signal-to-noise ratio (SNR)-dependent coherence factor (CF), which can be used for adaptive sidelobe suppression in ultrasound (US) imaging. Previous methods employed the minimum-variance-distortionless-response (MVDR)-based CF to achieve remarkable resolution improvement (by MVDR) and to suppress sidelobes (by CF). However, the SNR is often low when using an unfocused acoustic beam, giving such an approach suboptimal performance in these applications since noise also lowers the coherence and thus affects the effectiveness of the sidelobe suppression by these CF-based methods. To overcome this problem, the proposed method takes into account the local SNR in the CF formulation so that the contrast can be restored even when the SNR is low. Simulations show that the proposed method performs well even when the SNR is as low as -10 dB. Compared to the conventional CF, the contrast (CR) and contrast-to-noise ratio (CNR) in clinical US imaging can be improved, e.g., by an average of 27.2% in CR and 11.1% in CNR, with the proposed method.

9:10

2aBAb3. Two-dimensional blood flow vector obtained with high frame rate acquisition of dual-angle Doppler signal. Yoshifumi Saijo, Osamu Akagawa, Kosuke Fukazu, Naoya Tsugita, Sou Yaegashi, and Ryo Nagaoka (Tohoku Univ., 4-1 Seiryomachi, Aoba-ku, Sendai 980-8575, Japan, saijo@idac.tohoku.ac.jp)

Quantitative measurement of flow in blood vessels is important to assess cardiovascular dynamics. The objective of the present study is to obtain two-dimensional (2D) blood flow vector in some carotid artery models and to validate the method in comparison with particle image velocimetry (PIV). Plane wave ultrasound with the repetition rate of 14 kHz was transmitted and received by a 7.5 MHz linear array transducer and a programmable ultrasound data acquisition system. Compound B-mode images and dual-angle Doppler data with the transmission angle of -10° and +10° were obtained. 2D flow vector at arbitrary point was calculated by two velocity components along with each angle. Steady and pulsatile flow in normal and stenosed carotid artery models were simultaneously observed by the proposed method and PIV. 2D velocity vectors were successfully obtained in each flow states and the average error was approximately 5% in comparison with PIV in the carotid artery model. Steady flow in the distal stenosis model showed that the blood flow was influenced by vascular resistance while pulsatile flow seemed to push the blood to the high resistance portion. 2D blood flow obtained with dual-angle Doppler signal may provide important information in understanding cardiovascular dynamics.

9:30

2aBAb4. High frame-rate visualization of blood flow with ultrasound contrast agents. Matthew Bruce, Alex Hannah (Univ. of Washington, 8817 Interlake Ave. N, Seattle, WA 98103, mbruce@uw.edu), Charles Tremblay-Darveau, and Peter Burns (Univ. of Toronto, Toronto, ON, Canada)

Recent breakthroughs in diagnostic ultrasound system architectures have enabled new high frame rate (kHz) capabilities, which are opening new opportunities in blood flow imaging. In order to leverage these increases in temporal resolution broader or unfocused beams are employed. The combination high frame and less focused beams with ultrasound contrast agents introduces both new opportunities and trade-offs. One new opportunity is the ability to combine Doppler based processing to visualize both lower velocity blood flow in the microcirculation and higher velocity flow in larger vasculature not possible with conventional techniques. This can be accomplished by combining nonlinear pulsing sequences to separate microbubble and tissue signals with these less focused beams to separate different microbubble flow velocities. The use of less focused beams to image nonlinear echoes from microbubbles distributes ultrasound energy to microbubbles in temporally and spatially different ways. This work explores the compromises of using these less focused beams to visualize moving microbubbles at higher temporal resolutions. The focus will be on the practical challenges of generating nonlinear signals while balancing microbubble disruption and current system reconstruction capabilities. Both *in-vitro* and *in-vivo* results will be presented.

9:50

2aBAb5. Robust and high-frame-rate visualization of arterial pulse wave propagation dynamics. Alfred C. H. Yu (Elec. and Comput. Eng., Univ. of Waterloo, EIT 4125, Waterloo, ON N2L 3G1, Canada, alfred.yu@uwaterloo.ca), Adrian J. Y. Chee, and Billy Y. S. Yiu (Elec. and Electron. Eng., Univ. of Hong Kong, Pokfulam, Hong Kong)

Crucial to the facilitation of early detection of vascular diseases is the development of new imaging techniques that can track vascular dynamics at sub-millisecond time resolution. Previously, we have designed new high-frame-rate ultrasound imaging solutions that can track complex flow dynamics at >1000 fps frame rate, thereby providing hemodynamic insights for vascular diagnostics. Here, we have expanded our suite of vascular diagnostic tools by developing new imaging modes for mapping the structural dynamics of arterial walls. We have particularly devised a novel framework for robust ultrasound-based mapping of the spatiotemporal dynamics of arterial pulse wave (PW) propagation. Our framework involves: 1) plane wave data acquisition; 2) eigen-based data processing; and 3) phase-based motion estimation. The eigen-processing module represents a novel application of singular value decomposition, and its purpose is to isolate the desired forward PW through identifying the corresponding left singular vector through a frequency analysis algorithm, thus making our framework robust against the biasing impact of PW reflections. Also, to visualize PW propagation, positions of PW front were mapped and trailed with its past positions. Using this new eigen-based PW visualization framework, we can now accurately identify differences in PW propagation dynamics in arterial segments of different wall thicknesses and wall elasticity. Corresponding cineloops will be shown at the meeting.

10:10–10:30 Break

10:30

2aBAb6. Ultrafast plane-wave imaging of ocular anatomy and blood-flow. Ronald H. Silverman (Ophthalmology, Columbia Univ. Medical Ctr., 635 W 165th St., Rm. 711B, New York, NY 10032, rs3072@cumc.columbia.edu), Jeffrey A. Ketterling (Riverside Res., New York, NY), and Raksha Urs (Ophthalmology, Columbia Univ. Medical Ctr., New York, NY)

Ophthalmic ultrasonography is currently performed virtually exclusively with mechanically scanned, focused single-element transducers. This technology provides B-scan images at approximately 10 frames/s and does not offer any capacity for imaging or measurement of blood-flow. Conventional linear array systems often exceed FDA guidelines for ophthalmic exposure, especially in Doppler modes. Plane-wave techniques offer both ultrafast imaging and reduced acoustic intensity, since no focusing is performed on transmit. Using a 128-element, 18 MHz linear array, we adapted compound coherent plane-wave techniques for imaging ocular anatomy and blood flow at up to 20,000 frames/s. We measured acoustic intensity at the elevation focus and demonstrated conformance to FDA ophthalmic guidelines. We imaged human subjects and visualized blood flow by acquiring data continuously for at least one cardiac cycle. By applying a singular value decomposition filter to the spatiotemporal covariance matrix, we formed color-flow images and measured flow velocity over the cardiac cycle in the major retrobulbar vessels, choroid and anterior segment, including the iris and ciliary body. Given the major role of blood flow in diseases such as glaucoma, macular degeneration, diabetic retinopathy, and tumors (among others), this technique offers an important new diagnostic capability in ophthalmology not available using current ophthalmic B-scanners or optical coherence tomography.

10:50

2aBAb7. 4-D ultrafast ultrasound imaging: *in vivo* quantitative imaging of blood flows and tissue viscoelastic properties. Mathieu Pernot, Mafalda Correia, Jean Provost, Clement Papadacci, and Mickael Tanter (Institut Langevin, ESPCI, 1, rue Jussieu, Paris 75004, France, mathieu.pernot@gmail.com)

Very high frame rate ultrasound imaging has enabled the development of novel imaging modalities such as the functional imaging of the brain, cardiac electrophysiology, and the quantitative real-time imaging of the intrinsic mechanical properties of tumors, to name a few. We present here the extension of Ultrafast Ultrasound Imaging in three dimensions based on the use of either diverging or plane waves emanating from a sparse virtual array located behind the probe. High contrast and resolution were achieved while maintaining imaging rates of thousands of volumes per second. A customized portable ultrasound system was developed to sample 1024 independent channels and to drive a 32X32 matrix-array probe. Its capability to track in 3D transient phenomena occurring in the millisecond range within a single ultrafast acquisition was demonstrated for 3-D Shear-Wave Imaging, 4-D Ultrafast Doppler Imaging, and 4D vector flow imaging. These results demonstrate the potential of 4-D Ultrafast Ultrasound Imaging for the volumetric real-time mapping of stiffness, tissue motion and flow in humans *in vivo* and promises new clinical quantitative applications of ultrasound with reduced intra- and inter-observer variability.

11:10

2aBAb8. Role of high-frame-rate imaging for monitoring high-intensity focused ultrasound treatment. Shin-ichiro Umemura (Graduate School of Biomedical Eng., Tohoku Univ., Aoba 6-6-05, Aramaki, Aoba-ku, Sendai 980-8579, Japan, sumemura@ecei.tohoku.ac.jp), Shin Yoshizawa, and Ryo Takagi (Graduate School of Eng., Tohoku Univ., Sendai, Miyagi, Japan)

During high-intensity focused ultrasound (HIFU) treatment, tissue change, and acoustic cavitation as its precursor, should be monitored both in and outside the HIFU focal zone, to ensure the intended therapeutic effect and to prevent potential adverse effects, respectively. Unlike boiling-induced bubbles, cavitation-generated microbubbles are difficult to detect with conventional B-mode imaging because their life is significantly shorter than the typical image repetition period. An image repetition period less than a few milliseconds is required to detect them. This can only be accomplished by high-frame-rate imaging with plane or diverging waves. *In-vivo* as well as ex-vivo experimental results evidencing the effectiveness of such imaging to detect cavitation-generated microbubbles will be shown in this paper. A number of researches are underway to ultrasonically detect HIFU-induced tissue change, which occurs at much lower temperature than boiling. During echo data acquisition for detecting such tissue change, HIFU exposure is typically intermitted. High frame-rate-imaging also has a significant advantage there because the tissue motion by being released from the radiation force by HIFU can be ignored if the intermission is within a few milliseconds. Ex-vivo experimental results evidencing the effectiveness to detect HIFU-induced tissue change by high-frame rate imaging will also be shown.

Contributed Papers

11:30

2aBAb9. High frequency ultrafast flow and strain imaging in the carotid bifurcation: An ex vivo phantom study. Anne E. Saris, Stein Fekkes, Maartje M. Nillesen, Hendrik H. Hansen (Radiology and Nuclear Medicine, Radboud Univ. Medical Ctr., Music 766, PO Box 9101, Nijmegen 6500HB, Nijmegen, N/A, Netherlands), and Chris L. de Korte (Radiology and Nuclear Medicine, Radboud Univ. Medical Ctr., Netherlands, chris.dekorte@radboudumc.nl)

In the Western World, the prevalence of plaques in the carotid is high resulting in stroke and transient ischemic events. Quantification of the plaque size and compositions and the remaining flow and velocity profile is crucial for adequate therapy. We developed a high frequency based method for characterization of plaques and flow. A realistic bifurcation phantom with plaque based on CT scans of a patient was constructed. A realistic pulsatile flow was imposed on the phantom resembling a human flow and pressure profile. Using a Verasonics experimental echomachine with a Visualsonics M250 transducer (Fc = 20MHz), plane wave ultrasound data were acquired at 12.000 fps. A cross-correlation based coarse to fine method was applied on the beamformed data to quantify the axial and lateral strain in the arterial wall and to determine the velocity magnitude and direction of the blood. The strain and flow data were presented side by side for 80 frames covering the full pressure cycle. Strain in the non-diseased carotid wall was moderate with respect to the surrounding tissue. The bifurcation shows large lateral strain due to the force of the pulsating blood. The flow profile shows a turbulent region in the systolic phase at the bifurcation.

11:45

2aBAb10. Plane-wave imaging of in utero mouse embryo at 18 MHz. Jeffrey A. Ketterling (Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., New York, NY 10038, jketterling@riversideresearch.org), Orlando Aristizabal (Skirball Inst. of Biomolecular Medicine, NYU School of Medicine, New York, NY), Colin K. L. Phoon (Div. of Pediatric Cardiology, NYU Langome Medical Ctr. and Fink Children's Ctr., New York, NY), Billy Y. S. Yiu (Dept. of Elec. and Electron. Eng., Univ. of Hong Kong, Pokfulam, Hong Kong), and Alfred C. H. Yu (Dept. of Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

Plane-wave imaging methods allow for high-speed image capture at a time interval equal to round trip acoustic propagation. Plane-wave imaging is ideally suited for cardiovascular imaging where fine-temporal resolution can reveal important information about cardiac mechanics and blood flow patterns. While plane-wave imaging has been demonstrated in humans for cardiovascular studies, its use in mouse models lags because instrumentation is not yet widely available at appropriate ultrasound frequencies. Thus, the amount of functional information that can be mined from mouse models of cardiovascular disease is limited. Here, an 18-MHz linear-array probe was used to acquire plane-wave data at a frame rate of 10 kHz from an in utero, E14.5 mouse embryo. The probe had 128 elements, 1.5 mm elevation aperture, and 8-mm elevation focus. The mother was placed supine on a heated mouse imaging platform, and then, a series of 2D + time data sequences were captured. The data were beamformed using standard delay-and-sum methods, and then, vector flow estimates were obtained at each pixel location using a least-squares, multi-angle Doppler analysis approach. Although not optimally suited for imaging mouse embryos, the 18-MHz data clearly revealed blood flow patterns in the liver, heart, and umbilical cord.

Session 2aBAc

Biomedical Acoustics and Physical Acoustics: Cavitation in Therapeutic Ultrasound III: Tissue Fractionation

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

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

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Communications Engineering, Tohoku University, 6-6-05 Aoba, Aramaki, Aoba-ku, Sendai 980-8579, Japan

Chair's Introduction—9:50

Invited Papers

9:55

2aBAc1. Histotripsy: Transcranial applications. Charles A. Cain, Jonathan Sukovich, Timothy L. Hall, and Zhen Xu (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48109-2099, cain@umich.edu)

Histotripsy produces non-thermal lesions by generating, when an intrinsic threshold is exceeded, dense highly confined energetic bubble clouds that mechanically fractionate tissue. This nonlinear thresholding phenomenon has useful consequences. If only the tip of the waveform (P-) exceeds the intrinsic threshold, small lesions less than the diffraction limit can be generated. Moreover, side lobes from distorting aberrations can be "thresholded-out" wherein part of the relatively undistorted main lobe exceeds the intrinsic threshold producing a clean bubble cloud (and lesion) conferring significant immunity to aberrations. The short very high intensity histrotripsy pulse is significantly backscattered by the cloud it creates, allowing aberration corrections to be made in real time using transmitting elements as receivers. This allows significant increases in peak negative pressures that can be generated through the skull. These same backscattered signals allow a 3D representation of the bubble cloud to be generated providing excellent noninvasive bubble cloud localization. These refinements allow noninvasive transcranial procedures outside of expensive MRI machines.

10:15

2aBAc2. Pilot *in vivo* studies on transcutaneous boiling histotripsy in porcine liver and kidney. Vera Khokhlova, George Schade, Tatiana Khokhlova, Yak-Nam Wang, Julianna Simon, Frank Starr, Adam Maxwell, Michael Bailey, and Wayne Kreider (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, va.khokhlova@gmail.com)

Boiling histotripsy (BH) uses millisecond-long pulses of focused ultrasound (FUS) waves with shocks to mechanically homogenize tissue. Here we report a pilot *in vivo* acute study on transcutaneous volumetric BH ablation of porcine liver and kidney. BH treatment was administered using a 1.5 MHz FUS transducer operating at peak electric power ranging from 0.6 to 4 kW with B-mode ultrasound guidance using an imaging probe mounted coaxially. Sonication protocols delivered 5-30 pulses of 1-10 ms duration and 1% duty factor to focal points spaced 1-1.5 mm apart in a rectangular grid with 5-15 mm linear dimensions. Well-demarcated volumetric BH lesions were successfully generated in both liver and kidney without respiratory gating. The treatment was accelerated 6-fold without affecting the quality of tissue homogenization by using shorter duration BH pulses of larger peak power. These data indicate that transcutaneous volumetric renal and hepatic ablations are feasible in the *in vivo* porcine model. Studies are ongoing to optimize treatment parameters with the goal of clinical BH ablation of renal and hepatic masses. [This work was supported by NIH R01 EB7643, K01 EB015745, NSBRI through NASA NCC 9-58, RFBR 16-02-00653, and Urology Care Foundation.]

Contributed Papers

10:35

2aBAc3. Pilot evaluation of histotripsy treatment for Peyronie's disease.
Yak-Nam Wang, Adam D. Maxwell (APL, CIMU, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ynwang@u.washington.edu), Lynda Brady (Dept. of Mech. Eng., Univ. of Washington, Seattle, WA), George R. Schade, Haider Yasser, Franklin Lee (Urology, Univ. of Washington, Seattle, WA), William R. Ledoux (Ctr. for Limb Loss Prevention and Prosthetic Eng., VA Puget Sound, Seattle, WA), Michael R. Bailey (APL, CIMU, Univ. of Washington, Seattle, WA), and Hunter Wessells (Urology, Univ. of Washington, Seattle, WA)

Peyronie's disease (PD) is a debilitating scarring disorder of the penis which can cause significant curvature, pain, and psychological distress. Intra-lesional injection of specialized drugs to break down the fibrous matrix demonstrates modest results and is extremely expensive; surgery remains the gold standard for PD despite possible complications. We evaluated the feasibility of using histotripsy to disrupt the fibrous plaque and alter the mechanical properties as a novel non-invasive treatment. Freshly excised Peyronie's plaques (n = 5) were separated into individual samples for histotripsy treatment or control tissue. Treatments were conducted using a 1.0 MHz custom-built therapy transducer delivering pulses with repetition frequency of 1000 Hz, duty cycle 0.5%, and peak focal pressures of (98/-17 MPa) under ultrasound guidance. Tissue was either formalin fixed for histological evaluation or frozen for mechanical testing. Histotripsy induced disruption of collagen and elastin fibrils was evident in all fibrous portions of plaques. Histotripsy treatment produced decreased mean modulus, ultimate tensile strength, maximum load, and toughness vs. untreated control samples on mechanical testing. Additionally, the displacement to maximum load ratio was greater in treated vs. untreated control samples. [Work supported by NIH grants K01 DK104854, DK043881, EB007643, NSBRI through NASA NCC 9-58, and VA RR&D grant A9243C.]

10:50

2aBAc4. Treatment envelope of transcranial histoptripsy applied without aberration correction. Jonathan R. Sukovich, Zhen Xu, Timothy L. Hall, Steven P. Allen, and Charles A. Cain (Biomedical Eng., Univ. of Michigan, 1410 Traver Rd., Ann Arbor, MI 48105, jsukes@umich.edu)

Due to the characteristically high absorption, attenuation, and aberrating effects of the skull on ultrasound, transcranial HIFU therapies have typically been restricted to targeting brain regions at depths $\geq$2cm from the interior surface of the skull. Previous studies using histotripsy, a cavitation based ultrasound therapy relying on short duration (<2 acoustic cycles), high negative pressure amplitude (P $\leq$ -30MPa) ultrasound pulses to generate cavitation clouds to fractionate tissue, have shown that targeted bubble clouds can be generated through the skull without aberration correction. Here, we present results from experiments probing the minimum depths with respect to the skull surface at which histotripsy applied without aberration correction can generate lesions. Using a 500 kHz, 256-element transducer, histotripsy was applied through an ex vivo human skull to generate lesions in a tissue mimicking phantom embedded within. Lesions were generated at multiple locations within the skull through skull sections with characteristically different geometries. MRI was used to assess lesion generation following treatment. Using histotripsy without aberration correction, we were able to generate lesions at typical depths of between 6 and 12 mm from the interior skull surface. These results demonstrate the potential of histotripsy to target shallower brain regions than current HIFU therapies.

Invited Papers

11:05

2aBAc5. Non-invasive thrombolysis using microtripsy for deep vein thrombosis and intracerebral hemorrhage. Zhen Xu, Xi Zhang, Jonathan Sukovich, Tyler I. Gerhardson (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., Rm. 1107 Gerstacker Bldg., Ann Arbor, MI 48109, zhenx@umich.edu), Hitinder Gurm (Interventional Cardiology, Univ. of Michigan, Ann Arbor, MI), Gabe Owens (Pediatric Cardiology, Univ. of Michigan, Ann Arbor, MI), Aditya Pandey (Neurosurgery, Univ. of Michigan, Ann Arbor, MI), Tim L. Hall, and Charles A. Cain (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Histotripsy has been demonstrated as a non-invasive, drug-free, image-guided thrombolysis method using cavitation alone. Microtripsy is a new histotripsy approach, where cavitation is generated using 1-cycle ultrasound pulses with negative pressure exceeding a threshold intrinsic to the medium. We investigated microtripsy for treatment of deep vein thrombosis (DVT) and intracerebral hemorrhage (ICH). First, the *in vivo* feasibility of using microtripsy to treat DVT was investigated. Guided by ultrasound imaging, microtripsy thrombolysis treatment was applied to the clot formed in the femoral vein in 14 pigs using 1 μs-long pulses at a pulse repetition frequency of 100 Hz and a peak negative pressure of 30 MPa by a 1 MHz transducer. Blood flow was restored or significantly increased in 13 out of the 14 pigs confirmed by color Doppler. Minor hemolysis was observed. No vessel damage was observed on histology. Second, microtripsy was investigated to liquefy and drain clots through excised human skulls for ICH treatment using a 256-element, 500 kHz phased array transducer. Microtripsy generated individual lesions of 0.1 to 1.5 mm at the geometric location and electronically steered positions out to 20 mm from the geometric focus. By electronically steering the focus to cover a clot volume, microtripsy liquefied 40 mL of clot in 25 min, resulting in clot liquefaction rate of 1.6 mL/min.

11:25

2aBAc6. A preliminary *in-vivo* porcine study of long pulse histotripsy for thrombolysis of intraventricular hemorrhagic clot.
Thomas Looi (Hospital for Sick Children, 555 University Ave., Burton Wing 7142, Toronto, ON M5G1X8, Canada, thomas.looi@sickkids.ca), Vera Khoklova (Univ. of Washington, Seattle, WA), Kullervo Hynynen (Sunnybrook Res. Inst., Toronto, ON, Canada), and James Drake (Hospital for Sick Children, Toronto, ON, Canada)

Intraventricular hemorrhage is primarily a condition of premature babies where 40% will develop a form of bleeding that occupies >50% of the ventricles in the brain. Due to the fragile nature of patient, there is no treatment except the use of shunts for cerebrospinal fluid accumulation. Long pulse histotripsy (LPH) uses focused ultrasound pulses as a method to non-invasively target and mechanically break up the clot. An *in-vivo* IVH procine model has been developed and it is used to test the efficacy and safety of using LPH on a Philips Sonalleve. The IVH model has an average clot volume of 3986 mm^3 and present in both ventricles. To simulate a neonatal patient, a craniotomy has been performed. The acoustic parameters used are as follows: freq. 1.2 MHz, 2x6 mm focus, 10 ms pulse duration, 10,000 cycles, 1% duty cycle, and acoustic power from 325 to 400 W. Sonication points were placed at the center of the IVH clot. Pre-

and post-treatment T1-w, T2-w, and T2*-w MR imaging was completed. Change in the clot volume was measured by segmenting the MR images. The brains were harvested and stained with hematoxylin and eosin for histological examination. Results show that LPH targeted and reduced the clot volume by 28.5-36.7% with a phase change occurring at target. H&E staining showed that visible voids were generated in the clots. Based on early data, it appears that LPH can mechanically reduce the volume of IVH clots. Future work includes increasing the study number and conducting chronic studies to determine the changes to recovery.

Contributed Paper

11:45

2aBAc7. Visualizing the histotripsy process: Bubble cloud-cancer cell interactions in a tissue-mimicking environment. Eli Vlaisavljevich (Univ. of Michigan, 1111 Nielsen Ct. Apt. 1, Ann Arbor, MI 48105, evlaisav@umich.edu), Adam Maxwell (Univ. of Washington, Seattle, WA), Lauren Mancia, Eric Johnsen, Charles Cain, and Zhen Xu (Univ. of Michigan, Ann Arbor, MI)

Histotripsy is an ultrasonic ablation method that uses cavitation to mechanically fractionate tissue into acellular debris. Previous work has led to the hypothesis that the rapid expansion and collapse of histotripsy bubbles fractionate tissue by inducing large strain on the tissue structures immediately adjacent to the bubbles. In this work, the histotripsy fractionation process was visualized at the cellular level for the first time using a custom-built 2 MHz transducer incorporated into a microscope stage. A layer of breast cancer cells were cultured within an optically transparent fibrin-based phantom to mimic cells inside an extracellular matrix environment. The response to single and multiple histotripsy pulses was investigated using high speed optical imaging. Bubbles were generated in the extracellular space, and significant cell displacement/deformation was observed for cells directly adjacent to the bubbles. The largest displacements were observed during collapse for cells immediately adjacent to the bubble, with cells moving more than 150-300 μm in less than 100 μs. Cells often underwent multiple large deformations (>150% strain) over multiple pulses, resulting in the bisection of cells multiple times before complete rupture. These results support our hypothesis and help to explain the formation of the sharp lesions formed in histotripsy therapy.

TUESDAY MORNING, 29 NOVEMBER 2016 — SOUTH PACIFIC 4, 7:45 A.M. TO 12:00 NOON

Session 2aEA

Engineering Acoustics: Transducer Systems

Jason E. Gaudette, Cochair
NUWC Division Newport, 1176 Howell Street, B1371/3, Newport, RI

Yoshinori Takahashi, Cochair
Tokyo Metropolitan College of Industrial Technology, 8-17-1, Minamisenju, Arakawaku, Tokyo 116-0003, Japan

Contributed Papers

7:45

2aEA1. Broadband ultrasound emission from a three-layer thermoacoustic transducer. Olivier Y. Burguiere (Graduate School of Life and Medical Sci., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, dmp5501@mail4.doshisha.ac.jp), Shinji Takayanagi (Graduate School of Eng., Nagoya Inst. of Technol., Nagoya, Aichi, Japan), Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan), and Shizuko Hiryu (Graduate School of Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto, Japan)

A thermophone generates sound with temporal variation of Joule heat which leads to expansions and contractions of a small volume of air near the surface of the metallic film. Consequently, the SPL of the sound is proportional to the input power and the emitted sound is the second harmonic of the input signal.[1] Here, the fundamental characteristics of a three-layer thermophone (200-nm-thick Pt film, 5-μm-thick glass heat-insulating layer, and 500-μm-thick Si heat-releasing layer) were investigated. The device (10×12 mm^2, 2.32 Ω) showed a broadband frequency response up to 140 kHz, and its SPL level at 50 mm was nearly constant (61.5 ± 3 dB) from 20 to 140 kHz at 1.8 W power consumption. The conversion efficiency was 2.68×10^{-5}%, which is 150 times the efficiency of the authors' previous prototype, in which the heat release was less efficient (594-μm-thick alumina layer). Furthermore, by applying both AC and DC currents to the thermophone, it emitted the fundamental and the second harmonics of a 50-20 kHz FM sound, and hence a broadband (100-20 kHz) compound signal. These results and this device's simple manufacturing process suggest that it may be used as a high resolution ultrasound sensor. [1]Shinoda *et al.* (1999), Nature, 400(6747), p.853

8:00

2aEA2. A circular microphone array beamformer based on spherical harmonics. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com) and Jens Meyer (mh Acoust., Fairfax, VT)

A circular microphone array that can easily steer its beampattern(s) to any angle in the horizontal plane around the array would be an attractive microphone array for room audio conferencing. An elegant solution to this problem to design a modal beamformer based on circular harmonics. However, a modal beamformer based on circular harmonics suffers from the loss of control of the beampattern out of the plane of the array. We will present a modified modal beamformer for the circular array that is based on spherical harmonics. This approach allows full control of the beampattern and maintains the efficient steerability known from modal beamformers. Measurements of a specific implementation with 16 sensors will be presented.

8:15

2aEA3. Bio-inspired broadband sonar array prototypes and underwater experiments for two- and three-dimensional acoustic imaging applications. Jason E. Gaudette (Sensors and Sonar Systems, NUWC Div. Newport, 1176 Howell St., B1320, Newport, RI, jason.e.gaudette@navy.mil), Dimitri M. Donskoy (Davidson Lab., Stevens Inst. of Technology, Hoboken, NJ), Caleb J. Martin, Christin T. Murphy (Sensors and Sonar Systems, NUWC Div. Newport, Newport, RI), and James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI)

Underwater sonar imaging relies upon the information contained in time correlation delays across numerous channels. Designing higher resolution imaging systems currently requires increasing the array aperture to wavelength ratio (L / λ), where L is the effective array length and λ is the acoustic wavelength in the medium. This fundamental constraint leads engineers down the path of adding a significant number of channels to a sonar design, which in turn increases array complexity and cost. Our research in bio-inspired sonar has revealed an approach that circumvents this constraint by exploiting bandwidth in addition to time-delay for the angular imaging process. Presented will be the results of a 2-channel underwater prototype tested in an acoustic tank and design work toward a 3-channel prototype for extending imaging to elevation angles. This work represents ongoing efforts toward developing a compact, low-cost broadband underwater sonar for near- to mid-range imaging and classification. Future integration with an autonomous underwater vehicle will demonstrate this technology for simple obstacle detection and avoidance of complex objects. [Work supported by ONR and internal investments by NUWC Division Newport.]

8:30

2aEA4. An ultrasound focusing lens design incorporating high transmission and minimum diffraction. Xiasheng Guo (Inst. of Acoust., Nanjing Univ., No.22, Hankou Rd., Nanjing 210093, China, guoxs@nju.edu.cn), Zhou Lin (Inst. of Acoust., Nanjing Univ., Nanjing, Jiangsu, China), Gepu Guo (School of Phys. and Technol., Nanjing Normal Univ., Nanjing, Jiangsu, China), Juan Tu (Inst. of Acoust., Nanjing Univ., Nanjing, Jiangsu, China), Qingyu Ma (School of Phys. and Technol., Nanjing Normal Univ., Nanjing, Jiangsu, China), and Dong Zhang (Inst. of Acoust., Nanjing Univ., Nanjing, Jiangsu, China)

An improved design of ultrasound focusing lens is reported, in which a periodical array of grooves were carved on the surface of a conventional lens. With vibrations produced by a planar wave transducer coupled to the lens's left surface, each groove could be treated as a point source as long as the groove widths are much smaller than the acoustic wavelength. Several uncommon acoustic effects (*e.g.*, collimation and enhanced acoustic transmission) have been incorporated into the designing, providing the benefits ranging from transferring acoustic energies, suppressing the side-lobes, and minimizing shifting of the focal point. It is demonstrated theoretically and experimentally that acoustic focusing achieved by using the lens can suppress the relative side-lobe amplitudes, enhance the focal gain, and minimize the shifting of the focus. The application of the corrugated lens can make a reduction of about 3 dB in the relative side-lobe amplitudes, a 3.6-dB increase in main-lobe amplitude, and reduction or even elimination of the focus shift which could be enormously valuable in enhancing the safety of noninvasive HIFU therapy. Enlarging the number of grooves could even further enhance the capabilities of improving the focusing efficiency and reducing the relative side-lobe amplitudes.

8:45

2aEA5. The acoustical performance of an ultrasound matrix transducer directly stacked on a silicon chip. Maysam Shabanimotlagh, Shreyas Raghunathan, Nico d. Jong, and Martin D. Verweij (Imaging Phys., Tech. Univ. of Delft, Rm. D210, Bldg. 20, Lorentzweg 1, Delft 2628CJ, Netherlands, m.shabanimotlagh@tudelft.nl)

Clinical studies have shown a high demand for 3D real time ultrasound imaging for accurate diagnosis. One way to scan a volume with high frame rate is to design a 2D matrix transducer allowing beamforming in both lateral and elevation directions. Recent developments in manufacturing technologies allow the integration of a matrix of piezoelectric elements on Application Specific Integrated Circuit (ASIC). The ASIC is responsible for beamforming, amplification, switching, and significant channel count reduction. Traditionally, the backing material for a linear/phased array is composed of a mixture of epoxy with powder of heavy materials, which causes strong attenuation. This avoids energy being reflected back into to the elements and results in short pulses, high bandwidth, and consequently high axial resolution of the image. However, the ASIC acoustically behaves like a hard and non-absorbing backing material. Therefore, the acoustical energy of the piezoelectric pillar propagates into the chip and affects the performance of the neighbouring elements. In this paper we numerically investigate the effect of the ASIC on the acoustical performance of a transducer. In particular, the transmit and receive performance, crosstalk, and directivity pattern are compared with the results for a traditional backing material.

9:00

2aEA6. A micromachined low frequency microphone based on a field effect transistor and an electret. Kumjae Shin, Junsoo Kim, Hoonteak Lee, Donghwan Seo (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), Pohang, Gyungbuk, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), PIRO 405, POSTECH, San31, Hyoja-dong, Nam-gu, Pohang city, Kyungbuk, South Korea, Pohang, Kyungbuk 790784, South Korea, wkmoon@postech.ac.kr)

Recently, several Internet of Things (IoT) devices using low-frequency acoustic sound have emerged as promising sensor applications. Unfortunately, the detection of low-frequency sound using miniaturized microphones is restricted due to the low cut-off frequency of capacitive type transduction. To overcome this limitation, a micromachined microphone based on a field-effect transistor (FET) and an electret was reported and its feasibility as a low-frequency microphone was demonstrated in 2015. However, the proposed microphone was realized by bonding two chips mechanically and the FET was embedded in a membrane, which was disadvantageous for sensitivity enhancement. To realize stable highly sensitive modulation, we devised and fabricated a structure in which the electric field due to an electret embedded in the membrane modulates the channel of the FET. The acoustic signal causes the electret mounted on the membrane to vibrate, which changes the distance between the channel of the FET and the electret. The resulting change in the electric field modulates the conductivity of the channel. The use of an electret embedded on the membrane makes it possible to detect the displacement of the membrane directly and enhances the sensitivity of the microphone. Its feasibility as a low-frequency microphone will be examined experimentally.

9:15

2aEA7. Acoustic lens with exponential horns for selective distant recording. Maho Kobayashi (Tokyo Metropolitan College of Industrial Technol., 2-8-1 Minato-cho, #1811, Funabashi-shi, Chiba-ken 2730011, Japan, mintgreen070503@gmail.com) and Yoshinori Takahashi (Tokyo Metropolitan College of Industrial Technol., Tokyo, Japan)

Many signal processing techniques for sound source separation using microphone array have been studied as a robust audio recording method to ambient noise. However, the implementation in real-time is still hard to be realized in terms of its large computational cost. As an another approach, a directional microphone has a capability to record sound from a specific direction; however, it is impossible to pick up sound at a specific distance. As a part of research on acoustic transducer, on the other hand, a method to broaden the directivity by using the slant-plate acoustical lenses at the front of loudspeaker has been utilized. In addition, there is a report that a slant-plate acoustical lens combined with a conical horn shows a sharper directivity than that by a parabolic microphone. The authors have studied selective distant recording using acoustic lens that synchronously adds the wave fronts after removing the time difference of arrival via the speaking tubes having different lengths. As a result of that the shape of the speaking tube in the acoustic lens was replaced with exponential horn, an improvement in terms of sound quality with a reduction of resonance was obtained.

9:30

2aEA8. Annular 1-3 piezocomposite high intensity focused ultrasound transducer of a concave geometry. Yongrae Roh, Euna Choi, Seongwon Jang, and Minwoo Jo (School of Mech. Eng., Kyungpook National Univ., 80 Daehakro, Bukgu, Daegu 41566, South Korea, yryong@knu.ac.kr)

For medical therapeutic applications, we have developed a concave annular high intensity focused ultrasound (HIFU) transducer that is made of a 1-3 piezocomposite plate and operates at 3 MHz. The transducer has 16 annular channels, and the annuli have been designed to focus at the geometrical center of the sphere. The focal point can be controlled to scan a three dimensional volume by electronic and mechanical means. The focal point is moved along the normal axis of the transducer by electronically steering the sixteen annular channels. The focal point is also rotated back and forth by mechanically wobbling the piezocomposite plate with a kinematic linkage. The kinematic linkage is driven by an electric motor installed inside the transducer. The optimal combination of the number of channels, the width of the kerf between channels, and the width of each channel were determined by means of the OpQuest-Nonlinear programming algorithm and the finite element method. The objective of the optimization was to minimize the side lobe levels while focusing the ultrasound beam to a prescribed position. Based on the design, an experimental prototype of the transducer was fabricated and its performance was measured, which showed excellent agreement with the design.

9:45–10:00 Break

10:00

2aEA9. On implantable sensors for totally implantable hearing devices. Diego Calero, Stephan Paul, and Julio A. Cordioli (Mech. Eng. Dept., Federal Univ. of Santa Catarina, Campus Universitário, Trindade, Florianópolis, SC 88040-970, Brazil, diegocal_p@hotmail.com)

Conventional hearing devices, such as cochlear implants and hearing aids, use a microphone as a sensor for capturing the external sound field. Currently, the microphone is located in an external element, which is also responsible for processing the sound signal, but its presence is the cause of problems like discomfort, impossibility of being used during physical activities and sleeping, and social stigma. These limitations have driven studies with the goal of developing totally implantable hearing devices, and the design of an implantable sensor has been of the main issues to be overcome. Different designs of implantable sensors can be found in the literature and on some commercial implantable hearing aids, including different transduction mechanisms (capacitive, piezoelectric, electromagnetic, etc.), configurations (microphones, accelerometers, force sensor, etc.), and locations (subcutaneous or located in the middle ear). In this work, a detail review of such designs is presented and a general classification is proposed. The technical characteristics of the sensors are presented and discussed in view of the main requirements for an implantable sensor for hearing devices, including sensitivity, internal noise, frequency bandwidth, and energy consumption. The implantation feasibility of each sensor is also evaluated and compared.

10:15

2aEA10. Evaluation of spherical microphone baffle with six hollows for omnibinaural recording. Kento Imamura, Takanori Nishino, Taishi Nakagiri, and Hiroshi Naruse (Graduate School of Information Eng., Mie Univ., 1577 kurimamachiya-cho, Tsu, Mie 514-8507, Japan, imamura@pa.info.mie-u.ac.jp)

A binaural recording system records binaural sounds corresponding to plural listening orientations simultaneously. We devised and evaluated the binaural recording system. A binaural recording is generally performed with a dummy head; however, it can record for single listening orientation only. Moreover, an easy binaural recording system is desired because the dummy head is big, expensive, and difficult to handle. Therefore, we devised a cheaper and smaller omnibinaural recording system. In previous research, a spherical microphone baffle with two hollows was proposed, and the results addressed that it could be considered as a binaural recording system. In this study, a microphone baffle with six hollows was designed and evaluated. The proposed baffle has a spherical shape. Six hollows are located on three orthogonal axes originated from the center of the baffle. The acoustical features were examined by the numerical analysis. The results show that a difference between the previous research and the proposed baffle is small and the proposed baffle can achieve an omnibinaural recording.

10:30

2aEA11. Piezoelectric low frequency shear horizontal guided wave transduction. Guillaume Boivin, Nicolas Tremblay, Ricardo Zednik, and Pierre Belanger (Mech. Eng., Ecole de technologie superieure, 1100, Notre Dame Ouest, Montreal, QC H3C 1K3, Canada, pierre.belanger@etsmtl.ca)

Ultrasonic guided waves are now routinely used in non-destructive evaluation. In plate-like structures, three fundamental modes can propagate, namely, A_0, S_0, and SH_0. Most of the guided wave literature has thus far focused on the use of A_0 and/or S_0 because these modes are easy to generate in plate-like structures using standard piezoceramic transducers. Yet, at low frequency, A_0 and S_0 are dispersive. The consequence of dispersion is that signal processing becomes complex for long propagation distances. SH_0, on the other hand, has the particularity of being the only non-dispersive guided wave mode. Omnidirectional transduction of SH_0 requires a torsional surface stress which cannot be easily generated using standard piezoceramic transducers. This paper compares a transducer concept based on piezoceramic patches assembled to form a discretized circle and a second concept based on a tellurium dioxide disk. The external diameter of the transducers was chosen to be half the SH_0 wavelength at 100 kHz in an aluminium plate. Finite element simulations using the Comsol Multiphysics environment showed that in a 1.6 mm aluminium plate the modal selectivity as well as the omnidirectionality of the tellurium dioxide concept was superior at 100 kHz.

10:45

2aEA12. Thermal saturation and its suppression in high-power, compact carbon nanotube thin-film thermophones. Timothy A. Brungart, James J. Chatterley, Benjamin S. Beck, Brian L. Kline, and Zachary W. Yoas (Appl. Res. Lab., The Penn State Univ., Appl. Res. Lab., P.O. Box 30, State College, PA 16804-0030, tab7@arl.psu.edu)

Carbon nanotube (CNT) thin-film thermophones, at sufficiently high input power levels, suffer from thermal saturation where an increase in the input power does not result in a corresponding increase in the sound pressure level generated. It is believed that high temperature air, trapped in and around the CNT film, inhibits the ability of the CNT film to cool sufficiently between heating cycles, thus limiting the sound pressure output and increasing in severity with both input power and frequency. Thermal saturation appears to be particularly acute for CNT thermophones designed for compactness or when placed inside a protective or loudspeaker enclosure, where natural convection or heat transfer from the film is inhibited. Fan cooling was integrated into a CNT thermophone and demonstrated to both reduce the temperature of the CNT film and suppress, almost entirely, the effects of thermal saturation.

11:00

2aEA13. A new standard for transduction materials. Roger M. Logan (Teledyne, 12338 Westella, Houston, TX 77077, rogermlogan@sbcglobal.net)

A proposal has been made to establish a new (ANSI) standard to replace MIL-STD-1376B (cancelled). This presentation will expand on this proposal and provide a rough outline as a possible starting point for such a standard. Feedback and discussion will be encouraged.

11:15

2aEA14. Development of anti-cavitation hydrophone using a titanium front plate: Durability test in the high intensity focused ultrasound field. Michihisa Shiiba (Dept. of Clinical Eng., Nihon Inst. of Medical Sci., 1276 Shimogawara, Moroyamamachi Irumagun, Saitama 350-0435, Japan, m-shiiba@nims.ac.jp), Nagaya Okada (Honda Electronics Co., Ltd., Toyohashi, Aichi, Japan), and Shinichi Takeuchi (Toin Univ. of Yokohama, Yokohama, Japan)

Our research group has developed new anti-cavitation hydrophones by depositing a hydrothermally synthesized lead zirconate titanate polycrystalline film with 15 μm thickness on the back surface of a titanium front plate with 50 μm thickness and 3.5 mm diameter. A durability test of the anti-cavitation hydrophone was performed when the anti-cavitation hydrophone under test was placed at the focal point of a concave focused ultrasound transducer with 100 mm diameter and at a resonant frequency of 1.75 MHz. The amplified 80 $V_{p\text{-}p}$ (calculated electric input power: about 40 W) signal was applied to the concave ultrasound transducer at the focal point of the focused ultrasound system and high-intensity ultrasound waves were irradiated in water. The irradiated sound pressure at the focal point was about 4 MPa. Through this research, we will report that the fabricated new anti-cavitation hydrophone was robust and was not damaged easily, even in a high intensity focused ultrasound field with sound pressure of where acoustic cavitation occurred. We will make a durability test by increasing the acoustic power at the focal point of the focused acoustic field with sound pressure higher than 15 MPa.

11:30

2aEA15. Piezoelectric particle counter using the resonance vibration modes of a circular plate. Masatoshi Hayashi, Daisuke Koyama, and Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., 1-3 Tatara-Miyakodani, Kyotanabe, Kyoto 610-0394, Japan, dup0308@mail4.doshisha.ac.jp)

In the field of environmental measurements, robust and simple sensors with no electric power supply are required. In the river, the size distributions of sand and small stones in river-bed are important factor for river disaster prevention. In this report, a passive piezoelectric sensor for measurement of the particle distribution in flow was investigated. The sensor consists of an aluminum circular plate (diameter: 50 mm; thickness: 2 mm) and an annular piezoelectric transducer (inner radius: 5 mm; outer radius: 10 mm). Alumina spheres with several diameters (φ = 3, 5, and 8 mm) were employed as small particles. When the particles hit the surface of the sensor, the flexural vibration is excited on the circular plate and the electric power is generated through the piezoelectric effect. Two main peaks at 16.3 and 66.7 kHz appeared in the output signal, and the ratio of these peaks depended on the particle size. From the finite element analysis, it was found that these frequencies of 16.3 and 66.7 kHz correspond with the fundamental and second resonance vibration modes. These results indicate that the particle size can be determined from the frequency spectrum of the output signal.

11:45

2aEA16. Sound recording using optical marker-based motion tracking system: Relation between various constraints. Min-Ho Song and Rolf I. Godøy (Musicology, Univ. of Oslo, Institute for musikkvitenskap ZEB-bygningen 2. etg Sem Sælands vei 2, Oslo 0371, Norway, minho.song@imv.uio.no)

Optical marker-based motion tracking system is the device that can record the motion of moving object using multiple high-speed infrared (IR) cameras. Recent development of the motion capture device enables capturing the detailed motions with high spatial precision of sub-millimeter and high sampling rate up to 10 kHz. Currently, the motion tracking cameras can record the local vibrating movement of an acoustic instrument, which makes it possible to retrieve sound from the visual domain. In this study, two constraints in using marker-based motion tracking cameras for the sound recording are discussed. One is the temporal constraint where high sampling rates cause the detectability problem of the retro-reflective marker. When the cameras are operated with high speed, the cameras cannot radiate sufficient IR light and makes it hard to detect. The other is the amplitude constraint where the low camera calibration accuracy increases the signal-to-noise ratio. The effect of these constraints is observed using professional motion tracking cameras (Qualisys) and their relations over various physical conditions will be given.

Session 2aMUa

Musical Acoustics and Signal Processing in Acoustics: Computational Methods of Simulation for Musical Acoustics

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

Seiji Adachi, Cochair
Department of Acoustics, Fraunhofer Institute for Building Physics, Nobelstr. 12B, Stuttgart 70569, Germany

Chair's Introduction—8:00

Invited Papers

8:05

2aMUa1. Time history theoretical analysis for stick-slip vibration of violin strings. Hideo Utsuno (Mech. Eng., kansai Univ., 3-3-35 Yamate-cho, Suita 564-8680, Japan, utsuno@kansai-u.ac.jp)

The stick-slip vibration of violin string was completely formulated by using CA (Cellular Automaton) method in time history analysis. The local neighborhood rule of the CA method was derived for the travelling wave along the string, both ends of string, and the bow point. The key in success is that the bow point is to be fixed point and also to be velocity excited point at the same time. This means that reflected wave velocity is equal to the incidental wave velocity plus bow velocity. Displacement of arbitrary point of string is calculated and is compared with measured one. Excellent agreement of displacement suggests that CA method can simulate the stick-slip vibration of violin string. Acceleration time of the bow until it reaches a certain speed is also studied to form a complete Helmholtz waves. Step input and acceleration time within fundamental period is not valid to form Helmholtz wave. Acceleration time of more than four times of the fundamental period can substantially form Helmholtz waves.

8:25

2aMUa2. Accurate time-domain modeling of the bowed string. Hossein Mansour (Music Res., McGill Univ., 555 Sherbrooke St. West, Montreal, QC, Canada), Jim Woodhouse (Eng., Cambridge Univ., Cambridge, United Kingdom), and Gary Scavone (Music Res., McGill Univ., Montreal, QC H3A 1E3, Canada, gary@music.mcgill.ca)

An enhanced time-domain, travelling-wave model of the bowed string is presented. The model includes several new features: realistic damping verified by experimental data, detailed coupling of body modes (derived from experimental data) in both polarizations of string motion, coupling to transverse and longitudinal bow-hair motion, and coupling to vibration of the bow stick. The model is designed to allow the various features to be turned on or off, such that their influence on the results can be assessed independently. Schelleng diagrams are computed using different configurations and are compared based on different metrics to reveal trends of behavior.

8:45

2aMUa3. Time domain simulations of a novel lingual organ pipe construction. Péter Rucz, Nóra M. Nagy (Dept. of Networked Systems and Services, Budapest Univ. of Technol. and Economics, 2 Magyar Tuodósok körútja, Budapest H1117, Hungary, rucz@hit.bme.hu), Judit Angster (Dept. of Acoust., Fraunhofer Inst. for Bldg. Phys., Stuttgart, Baden-Württemberg, Germany), Fülöp Augusztinovicz (Dept. of Networked Systems and Services, Budapest Univ. of Technol. and Economics, Budapest, Hungary), and András Miklós (Steinbeis Transfer Ctr. of Appl. Acoust., Stuttgart, Baden-Württemberg, Germany)

In a traditional pipe organ, the dynamic range of both labial (flue) and lingual (reed) pipe ranks are strictly limited as each rank is tuned and voiced to a nominal windchest pressure. Changing this pressure not only affects the amplitude of the radiated sound but both the pitch and the timbre of the pipes. A new pipe construction with a blown open free tongue was proposed recently to overcome this limitation. Prototype pipes of the new construction were built and measurements were carried out on them at different blowing pressures. It was found that the new construction provides a pleasing stability of the pitch and a broad range of playable amplitudes; however, the timbre of the pipes changes significantly with the blowing pressure. To improve the design in the latter aspect, a physical model of the pipes needs to be established first. In this contribution the sound generation of the experimental pipes is simulated by time domain computations. The resonator and its interaction with the vibration of the tongue are simulated using both the truncated impedance model and the reflection function approach. Results of the temporal simulations and their comparison with measurements are presented.

9:05

2aMUa4. Modeling studies of wind instruments using the Navier-Stokes equations. Nicholas Giordano (Phys., Auburn Univ., College of Sci. and Mathematics, Auburn, AL 36849, njg0003@auburn.edu)

Modeling of musical instruments can be done at different levels of sophistication and realism. Depending on the goal of the simulation, it is sometimes possible to treat components of an instrument in an approximate way. For example, in dealing with a wind instrument it may be possible to consider a free running oscillator coupled to a one dimensional resonator with feedback, and ignore the physical details of the oscillator and feedback. However, in some cases, it is necessary to take a more exact approach and apply the fundamental laws of mechanics and fluid dynamics. For a wind instrument, this means the application of the Navier-Stokes equations, a set of nonlinear partial differential equations that require a numerical solution. In recent years, the available computational power has made it possible to apply the Navier-Stokes equations to instrument geometries that are fairly realistic. This talk reviews some of that work and discusses the kinds of questions that can be addressed with simulations of this kind. [Research supported by NSF grant PHY1513273.]

Contributed Papers

9:25

2aMUa5. Numerical study for the function of moving pad on tone hole acoustics. Taizo Kobayashi (Teikyo Univ., Misaki-machi, 6-22, Omuta, Fukuoka 836-8505, Japan, tkoba@cc.kyushu-u.ac.jp), Hidetaka Matsuda (The Phys. Labs., Kyushu Inst. of Technol., Iizuka, Fukuoka, Japan), Toshiya Tatami (Oita Univ., Oita, Japan), and Kin'ya Takahashi (The Phys. Labs., Kyushu Inst. of Technol., Iizuka, Fukuoka, Japan)

In this poster, we will discuss numerical methods for calculating a tone hole model with a moving pad of woodwind instrument by using compressible Large Eddy Simulation (LES), i.e., how to treat moving boundary problems. There are three stages of numerical methods to treat a moving pad on a tone hole. The 1st stage is the simplest method in which a dynamical mesh technique is not employed. It is so called "Stop Motion." The numerical conditions of the distance between the pad and the top of the tone hole are set to adequate points to discuss the effect of the distance, e.g., we set it to 0 mm (i.e., the tone hole is closed by a pad), 0.5 mm, 1 mm, 2 mm, 3 mm, and 5 mm. In the second stage, while a dynamical mesh technique is employed, on the other hand, it does not treat the condition with topological change of the boundary, i.e., in this situation, the distance changes from/into 0 mm. The third stage is a full dynamical mesh technique which provides a method for treating moving boundary problems with topological changes. As the results of calculation, we will also report how the pitch is changed with fingering, and how the fluid field and acoustical field behave around the tone hole when it is changed, comparing with experimental study on the function of tone holes reported by Keefe [1]. Keefe, D.H. ''Experiments on the single woodwind tone hole'', *J. Acoust. Soc. Am.*, **72** 688-699 (1982).

9:40

2aMUa6. Numerical investigation on the influence of wall vibrations on the behavior of the lip excitation in brass instruments. Vincent Fréour (Res. and Development Div., YAMAHA Corp., Yamaha Corp., 203 Matsunokijima, Iwata 438-0192, Japan, vincent.freour@music.yamaha.com) and Hideyuki Masuda (Res. and Development Div., YAMAHA Corp., Iwata, Shizuoka, Japan)

The influence of wall vibrations in brass instruments has been an object of substantial debate in both musician and scientific communities. The excitation of the instrument wall (from the acoustic field inside the instrument and through the direct mechanical excitation from the lips) may influence the acoustic pressure field inside and outside the instrument, potentially resulting into some noticeable effects on the acoustic input impedance and radiated sound. These effects have been the object of numerical and experimental studies on different families of wind instruments. Furthermore, these vibrations are also likely to induce some perturbations on the excitation mechanism itself (the lips of the player), that may contribute to variations of the produced tone, as well as influence the sensations of the player. In this paper, the effects of the this mechanical coupling between the lips and vibrating wall is studied numerically using a physical model of the lips, coupled acoustically to the air-column of the instrument, as well as mechanically with the instrument wall described by a simplified mechanical admittance. Results from time-domain simulations and linear stability analysis will be presented. Some pathological situations, where a structural mode is close to the playing frequency will be particularly discussed.

9:55

2aMUa7. Precision harmonics simulation for vacuum-tube electric-guitar pedals and preamplifiers. Kanako Takemoto, Shiori Oshimo, and Toshihiko Hamasaki (Information Systems and Sci., Hiroshima Inst. of Technol., 2-1-1, Miyake, Saki-ku, Hiroshima 731-5193, Japan, md16009@cc.it-hiroshima.ac.jp)

Recently, vacuum-tube pedals have attracted many electric guitar players, though solid-state circuits or even digital processors are widely accepted. The purpose of this study is to clarify the origin of distorted timbre of the vacuum-tube pedal by circuit simulation with a novel vacuum-tube model. In general, the specific sound of vacuum tube for both hi-fi and electric guitar amplifier is attributed to the even number of harmonics originated from the inherent non-linearity. However, significant difference exists in the input signal amplitude between hi-fi and electric guitar system. The simulation with a conventional non-linear model has been limited within the regulated signal swing of hi-fi audio system. The dynamic non-linear phenomena for large signal, such as an input signal modulation, are not considered. Thus, unrealistic harmonics appear in the pedal simulation due to discontinuity of model parameters. The new model is obtained by physical analysis based on enough measurements, considering the actual input condition of pedal and preamplifier as well. The key property is a seamless modulation of non-linearity influenced by circuit operation. As a result, we have succeeded in simulating changes of the dynamic harmonics accurately for the first time.

10:10

2aMUa8. Electronic cymbals using spectral modeling synthesis with variable timbre by hitting strength and beater hardness. Naoto Wakatsuki and Koichi Mizutani (Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8573, Japan, wakatuki@iit.tsukuba.ac.jp)

Sound source for cymbals in electronic drums based on spectral model synthesis were proposed instead of wavetable based sound sources. In the sound source, the timbre of the synthesized sound can be changed by hitting strength. In the real acoustical cymbals, the timbre change is caused by the coupling among the eigenmodes due to geometrical nonlinearities. In the sound source, it was reproduced using a simple model. As well as the hitting strength, the hardness of beaters hitting a sensor pad are also reflected to the timbre. It was achieved by measuring pulse width in the output analog signal from a commercial sensor pad. The proposed sound source was implemented as a Steinberg's Virtual Studio Technology (VST) plugin. In the talk, the implemented sound source will be simply demonstrated.

Session 2aMUb

Musical Acoustics and Signal Processing in Acoustics: Music Signal Processing I

James W. Beauchamp, Cochair
School of Music and Electrical & Computer Eng., University of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824

Masataka Goto, Cochair
National Institute of Advanced Industrial Science and Technology (AIST), IT, AIST, 1-1-1 Umezono, Tsukuba 305-8568, Japan

Chair's Introduction—10:40

Invited Papers

10:45

2aMUb1. Transcribing piano music in the time domain. Andrea Cogliati, Zhiyao Duan (Elec. and Comput. Eng., Univ. of Rochester, 308 Hopeman, Rochester, NY 14627, zhiyao.duan@rochester.edu), and Brendt Wohlberg (Los Alamos National Lab., Los Alamos, NM)

Automatic music transcription is the process of automatically inferring a high-level symbolic representation, such as music notation or piano-roll, from a music performance. It has many applications in music education, content-based music search, musicological analysis of non-notated music, and music enjoyment. Existing approaches often perform in the frequency domain, where the fundamental time-frequency resolution tradeoff prevents them from obtaining satisfactory transcription accuracies. In this project, we develop a novel approach in the time domain for piano transcription using convolutional sparse coding. It models the music waveform as a summation of piano note waveforms (dictionary elements) convolved with their temporal activations (onsets). The piano note waveforms are pre-recorded in a context-dependent way, i.e., for the specific piano to be transcribed in the specific environment. During transcription, the note waveforms are fixed, and their temporal activations are estimated and post-processed to obtain the pitch and onset transcription. This approach models temporal evolution of piano notes, and estimates pitches and onsets simultaneously in the same framework. Experiments show that it significantly outperforms a state-of-the-art frequency-domain music transcription method trained in the same context-dependent setting, in both transcription accuracy and time precision, in various scenarios including synthetic, anechoic, noisy, and reverberant environments.

11:05

2aMUb2. Unsupervised grammar induction from music data. Kazuyoshi Yoshii (Kyoto Univ., Res. Bldg. #7, Rm. 412, Yoshida-honmachi, Kyoto, Kyoto 606-8501, Japan, yoshii@kuis.kyoto-u.ac.jp)

Music has a lot of similarities to language. Since most languages have clear syntactic structures (e.g., words should be arranged in the SVO order), many linguists have proposed various kinds of grammar theories by carefully and manually investigating language data. The situation is the same with Western music. Although a single musical note (cf. alphabet) has no meaning by itself, a cluster or pattern of multiple musical notes over the quantized time-frequency grid (cf. word) can invoke some impression and such short patterns are concatenated or superimposed (unique to music) to produce more complicated meaning (cf. sentence). We introduce several attempts to discover latent structures underlying music from acoustic or symbolic data (music signals and musical scores) in an unsupervised manner. Integrating statistical acoustic and language models as in speech recognition, for example, it is possible not only to transcribe music but also to discover that particular note combinations can form chords. A key feature of this approach is that both models are trained jointly only from acoustic data. Recently, we have attempted to induce music grammars from polyphonic scores by leveraging the state-of-the-art techniques of natural language processing. This would contribute to automatic music transcription and computational musicology.

11:25

2aMUb3. End-to-end music transcription using a neural network. Paris Smaragdis (CS & ECE, Univ. of Illinois, 201 N Goodwin Ave., Office 3231, Urbana, IL 61801, paris@illinois.edu)

We present various neural network models that learn to produce music transcriptions directly from audio signals. Instead of employing commonplace processing steps, such as frequency transform front-ends or temporal smoothing, we show that a properly trained neural network can learn such steps on its own while being trained to perform note detection. We demonstrate two models that use raw audio waveforms as input and produce either a probabilistic piano roll output or text in music notation format that can be directly rendered into a score.

11:45

2aMUb4. Multiple-timbre note tracking using linear dynamical systems. Emmanouil Benetos (School of Electron. Eng. and Comput. Sci., Queen Mary Univ. of London, Mile End Rd., London E1 4NS, United Kingdom, emmanouil.benetos@qmul.ac.uk)

This work addresses the problem of multi-pitch detection and note tracking in multiple-instrument polyphonic music recordings. A system is developed that extends probabilistic latent component analysis (PLCA) and supports the use of a five-dimensional dictionary of spectral templates per instrument, pitch, deviation from ideal tuning, and sound state (e.g., attack, sustain, decay). A method based on linear dynamical systems (LDS) is introduced for note tracking, which assumes that the output of the PLCA model is the (noisy) observation in an LDS, with the latent states corresponding to the ideal multi-pitch activation output. The LDS-based process supports the tracking of multiple concurrent pitches and can also be integrated within the PLCA-based model, thus guiding the convergence of the multi-pitch detection process. Experiments performed on several datasets of multiple-instrument polyphonic music demonstrate that the LDS-based method leads to significant improvements in multi-pitch detection as compared to using the frame-based PLCA model alone.

TUESDAY MORNING, 29 NOVEMBER 2016 SOUTH PACIFIC 3, 7:45 A.M. TO 12:00 NOON

Session 2aNS

Noise and Architectural Acoustics: Noise Impacts and Soundscapes at Outdoor Gathering Spaces

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

Brigitte Schulte-Fortkamp, Cochair
Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

Chair's Introduction—7:45

Invited Papers

7:50

2aNS1. Nestled in nature but near to noise—The Ford Amphitheatre. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The historic Ford Amphitheatre was relocated in 1920, from what would later become the location of the Hollywood Bowl, across the Hollywood Freeway, into an arroyo with a dramatic natural backdrop. The original wood structure was destroyed by a brush fire in 1929, and rebuilt in concrete in 1931. In the meantime, the freeway has become increasingly noisy. The current renovation to this 1200 seat outdoor amphitheatre includes an expanded "sound wall" that will help to mitigate freeway noise while providing optimal lighting and control positions. The remarkably uniform distribution of ambient noise throughout the seating area, the significantly reverberant character of this outdoor space, and the apparent contributions by the arroyo will be discussed, along with assorted design and construction challenges.

8:10

2aNS2. Acoustics upstaged: Challenging conundrums at the Starlight Bowl. David A. Conant (McKay Conant Hoover, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, dconant@MCHinc.com)

Since its debut in 1935, the 3600-seat Starlight Bowl in San Diego's renowned Balboa Park has been increasingly impacted by environmental noise, principally from aircraft landing at nearby Lindbergh Field. Traditionally, performers would temporarily "freeze" on cue from the director as a jet approached, then resume as though nothing had happened. Although amusing initially, audiences soon were just annoyed. MCH was retained to identify and rank prospective mitigation measures as the principal tenant was losing audience. In addition to collecting the usual ambient and fly-by noise level data, MCH examined the efficacy of partial and full enclosure options, and associated implications on audio system planning and operation. Ultimately, the myriad of acoustical/audio noise-mitigating options developed could not be reconciled with the historic venue's name, as the visual impact alone, of large jets flying within 600 ft would forever be up-staging any on-stage performance. We will report on our process, observations, and recommendations.

8:30

2aNS3. "Shhh." and other methods for controlling crowd noise. Shane J. Kanter and Jonathan Laney (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, skanter@thresholdacoustics.com)

As designers of venues for performance, we spend a considerable amount of time and effort on maintaining a low background noise level while the acoustically critical spaces are in use. The typical noise sources, such as mechanical systems and light fixtures, are predictable and controllable. Just as background noise is inferential, audience noise within the common classical music venue is generally understood and under control. Effects of outdoor background noise are explored—such as city noise, road noise, and cicadas at two Chicago outdoor venues: Jay Pritzker Pavilion and Ravinia Festival. Second, because both of these venues have a fixed section of seating near the stage and a large section of general admission law seating, the lawn seating typically fosters a relaxed atmosphere where patrons feel free to converse with one another during performances. Ravinia and the Pritzker Pavilion differ when it comes to how they deal with chatting patrons during performances. Special attention will be paid to the audience noise in both the fixed seating and lawn seating at each venue and through the use of patron interviews, and exploration the effect of audience noise on the performance experience.

8:50

2aNS4. Amphitheaters: The noise within. Richard H. Talaske (TALASKE | Sound Thinking, 1033 South Blvd., Oak Park, IL 60302, rick@talaske.com)

The continuing success of outdoor performance facilities demonstrates that music and theatre can be enjoyed within environments which are much nosier than the silent indoor concert halls and theaters we know and love. However, there are limits to how much noise is acceptable. Design methods can be implemented to control the noise and significantly improve the listening experience. This presentation offers insight into the design of outdoor music venues and the management of noise as heard by the patron and performers. Numerous amphitheaters will be discussed, including the Jay Pritzker Pavilion, American Players Theatre, and Aurora's River's Edge, thereby offering examples of facilities designed for symphonic music, theatre, and popular music. The importance of acoustic considerations such as signal-to-noise, frequency response, and direction of arrival of sound will be discussed.

9:10

2aNS5. Overcoming challenges to future outdoor entertainment performances due to disruptive commercial vendor noise intensity and the resulting community outrage. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

Outdoor entertainment and concerts are commonplace historically. Many of these public attractions also include commercial, concessions, and other vendor booths. Typically, such events are regulated by local government. During the inaugural performance of a successful music festival, whose sponsors had contractually agreed to the community permitting requirements, a commercial vendor proceeded to generate such loud and intense sounds that the entire festival and closest residential community were bombarded with object vibrating sounds. The public outcry from the local residential community was immediate and so was the reaction of the festival organizers. The commercial vendor was ordered to vacate the venue. However, the local community was so outraged that the festival organizers were put on notice that the festival had violated its contract and would not be permitted again. After many hours of community outreach and public hearings, the festival organizers were given a highly conditional permit to hold the festival again the following year. Strict A-weighted and C-weighted noise limits and heavy violation fines were imposed, as well as a requirement for continuous noise monitoring at multiple sites at the venue and in the residential community. This paper reports the concert performance results of the past few years.

9:30

2aNS6. Changes to soundscapes at outdoor gathering spaces in Fukushima, Japan, caused by the severe nuclear power plant accident. Koji Nagahata (Fukushima Univ., 1 Kanayagawa, Fukushima 960-1296, Japan, nagahata@sss.fukushima-u.ac.jp)

Five years have passed since the accident at the Fukushima Daiichi Nuclear Power Plant, which caused changes to daily life in the city of Fukushima, Japan. These daily life alterations caused changes to the soundscapes of Fukushima, which are still evolving now. In this study, the soundscape change at Fukushima's outdoor gathering spaces is discussed using field recordings by the author. Shortly after the accident, few human voices could be heard at outdoor gathering spaces, but natural sounds such as birdsongs could be heard as usual per the time of year. Park decontamination started during summer 2011. In some parks where decontamination was successfully completed, people's voices and sounds of children playing returned during spring 2012. However, in other parks where decontamination was done but ineffective, the lack of human voices and artificial sounds continued until radiation levels decreased sufficiently. In this way, soundscape change at outdoor gathering spaces represents people's attitude toward the radioactive contamination; soundscape recording documents not only sonic environments but also people's lives.

9:50–10:05 Break

10:05

2aNS7. E-participation in the context of the evaluation of the sonic environment. André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de), Brigitte Schulte-Fortkamp (ISTA Institut für Technische Akustik, Technische Universität Berlin, Berlin, Germany), and Klaus Genuit (HEAD Acoust. GmbH, Herzogenrath, Germany)

E-participation methods are increasingly applied to promote public participation and to achieve greater awareness and involvement by citizen by providing the opportunity to directly report to, e.g., governmental organizations. Within the framework of the German Year of Science 2015, a special initiative, called "Sound of the City," was carried out to encourage the public to report on how their cities typically sound. Participants could upload audio recordings of acoustical environments and had to relate the respective audio files to the corresponding places. Moreover, the participants could additionally describe the uploaded sounds. This action lasting several months led to a comprehensive database of audio files including related places, descriptions, and ratings. The database was systematically analyzed with respect to several issues, such as general participation level, hot spot identification, and common features of uploaded sounds. The paper will highlight the benefit and limitations of such actions and initiatives applying e-participation with respect to community noise issues and urban planning. [The "Sound of the City" initiative took place within the framework of the Year of Science 2015 and was organized and supported by the Federal Ministry of Education and Research (BMBF).]

10:25

2aNS8. Next generation soundscape design using virtual reality technologies. Andy Chung (Smart City Maker, Hong Kong Plaza, Hong Kong HKSAR, Hong Kong, ac@smartcitymaker.com), Wai M. To (Macao Polytechnic Inst., Macao, Macao), and Brigitte Schulte-Fortkamp (TU, Berlin, Germany)

Sound quality has been demanded by the community as part of the smart city initiatives to enjoy a real liveable environment. While a lot of governments pay attention to environmental issues such as air quality and waste, those who care about the sonic environment still rely primarily quantitative levels for either compliance or improvement. Sound quality is most likely ignored. There is at present the international standard ISO 12913-1:2014 providing the definition and the conceptual framework of soundscape. Despite the situation, sound quality is a subjective matter after all and relies very much on human perceptions, as well as the contextual environment. Sound walk, questionnaire, and lab test are common tools used in soundscape studies. These tools allow a good understanding of the perception of the prevailing sonic environment. To figure the sound quality of different design options, though, we can immerse the participants in a virtual, but photorealistic environment, so that they can perceive, compare, and give feedback as if they were in the real environment. A mobile VR application has been developed for iterating soundscape design and fine-tuning. This paper presents this application with case studies.

Contributed Papers

10:45

2aNS9. On-site and laboratory soundscape evaluations of three recreational urban spaces. Anna Josefine Sørensen (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads 352, Kgs. Lyngby 2800, Denmark, annajosefine@gmail.com), Thea Mathilde Larsen, Lærke Cecilie Bjerre (Dept. of Civil Eng., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark), Sébastien Santurette (Hearing Systems, Dept. of Elec. Eng., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark), and Cheol-Ho Jeong (Acoust. Technol., Dept. of Elec. Eng., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

Soundscape quality was evaluated using four subjective psychological rating factors in three recreational urban spaces in which water and a variation of other natural and anthropogenic sound sources were present. The noise level was measured at each site during occupant peak flows and recordings for listening experiments were made simultaneously. Listeners answered questionnaires either on site or following playback of the recordings in the laboratory, with or without access to each site's visual context. They rated their perception of loudness, acceptance, stressfulness, and comfort, along with their preference toward eight sound sources. The comfort ratings were negatively correlated with loudness and stressfulness and positively correlated with acceptance. The sound level was found to be a good predictor of these subjective parameters in the laboratory, but not on site. Moreover, the availability of the visual context in the listening experiment had no effect on the ratings. The presence of trees and water was also found to increase on-site comfort. Generally, the participants were more positive towards natural sound sources on-site. Overall, the results suggest that on-site context plays an important role for evaluating acoustic comfort in urban recreational areas.

11:00

2aNS10. Real-time natural soundscape generation based on current weather conditions for workspace voice-masking. Caitlin Riggs and Jonas Braasch (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, riggs.cait@gmail.com)

Sound masking in work spaces has been implemented to decrease likelihood of distraction from work tasks and improve speech privacy. Current uses of noise as maskers commonly apply broadband white or pink noise signals due to their perceived neutrality. This research combines work into the restorative properties of exposure to nature/natural sounds and pilot studies of natural sounds as maskers to suggest a noise-masking system of "natural" sounds. This system composes a natural soundscape in real time determined by the current weather and time of day such that the masking audio is aesthetically pleasing and informative about the outside world in addition to providing improved speech privacy and reducing distraction when compared to a setting with no masking system. Currently, there is no experimental foundation to suggest that restoration or slowed attentional fatigue can occur if this type of alternative masking sound is presented during a task. This implementation of a dynamic, immersive soundscape masker begins the investigation into the practical efficacy of such a masking system.

11:15

2aNS11. Evaluation of the noise disturbance and noise limits of a riding horse shooting competition. Juan C. Montoya (Cross Spectrum Acoust., 41 Churchill St., Springfield, MA 01108, jmontoya@csacoustics.com)

Shooting ranges have turned into an unwanted activity in rural communities due to high levels of impulse sound pressure levels. Permanent complaints from neighbors or neighborhood associations pertaining health concerns and noise pollution around areas nearby shooting ranges are a driving force to create state noise control laws. As such, examining zoning will limit (restricting licenses) to the use of agricultural or residential land to be used for shooting events. Gun noise is measurable and therefore is able to be quantified and characterized. This study evaluates with field noise measurements the events from a "shooting ranch" which currently operates offering a shooting competition while riding a horse emulating the Old West. The analysis of field data pretends to look at metrics such as LAeq (A-weighted equivalent sound level), L90 (noise level exceeded for 90% of the measurement period), and Lmax (maximum level if the measurement period from 50 caliber single action revolver gun sounds while shooter is in movement riding a horse. The field measurements were conducted in different nearby residences. The impulse noise lead to examine instead individual gunshots which exceeded background levels by 14 to 34 decibels. Noise levels from Shooting Ranch events are sometimes more than three times the background noise level. Leq and L90 metrics are applicable within individual events.

11:30–12:00 Panel Discussion

Session 2aPA

Physical Acoustics and Noise: Acoustics of Supersonic Jets: Launch Vehicle and Military Jet Acoustics I

Seiji Tsutsumi, Cochair
JEDI Center, JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara 252-5210, Japan

Kent L. Gee, Cochair
Brigham Young University, N243 ESC, Provo, UT 84602

Janice Houston, Cochair
Jacobs ESTS Group, 1500 Perimeter Pkwy, Suite 400, Huntsville, AL 35806

Chair's Introduction—8:15

Invited Papers

8:20

2aPA1. Influence of launch platform cut-outs on flow and acoustic behavior of rocket exhaust. Karthikeyan Natarajan and Lakshmi Venkatakrishnan (Experimental AeroDynam. Div., CSIR-National Aerosp. Labs., EAD, PB 1779, Old Airport Rd., Bangalore 560017, India, nkarthikeyan@nal.res.in)

A launch vehicle experiences intense acoustic loading in the initial phase of its lift-off which affects the launch vehicle structure, sensitive payloads, and electronics on board. There is immense interest in alleviation of acoustic loads resulting in reduced need for strengthening of the vehicle structure. The effect of jet blast deflector shape on the acoustic loading has been extensively investigated, both computationally and experimentally, by simulating jet(s) impinging on a flat plate. However, contributions from the launch vehicle environment, such as the launch platform, are often ignored. The motivation for this study is that the flow over the launch platform is likely to be significantly influenced by the cut-outs made in the platform for the nozzles. As the nozzles emerge out from the cut-outs during lift-off, the jet exhaust grows and interacts with the launch platform, contributing to the overall acoustic loads experienced by the vehicle. This paper presents an experimental investigation of rocket exhaust interaction with the launch platform using single and twin jets impinging on a flat plate with cut-outs. The measurements include high speed shadowgraphy and microphone measurements in the near and far-field to enable flow and acoustic characterization.

8:40

2aPA2. A design method of fairing acoustic noise mitigation for Korean space launcher. Soon-Hong Park and Sang-Hyun Seo (Launcher Structures Dept., Korea Aerosp. Res. Inst., 169-84 Gwahangno, Yuseong-gu, Daejeon 305806, South Korea, shpark@kari.re.kr)

In this presentation, fairing acoustic protection system (APS) for Korean space launcher is introduced. The design of APS for payload fairing of KSLV-II is summarized. The current APS consists of acoustic blankets for midfrequency range absorption and acoustic resonators for low frequency noise absorption. Detailed design procedures of APS, which include external acoustic loading prediction based on the modified NASA SP-8072 monograph and fairing system vibro-acoustic analysis, are presented. An acoustic test of a cylindrical composite structure with APS is also presented. The effect of spatial correlation of external acoustic excitation on the vibro-acoustic response of fairing structure and its interior volume is discussed. The concept of future APS by using the micro-perforated panel absorber combined with Helmholtz resonators is also introduced. Effects of high pressure environment on the acoustic absorption of micro-perforated panel is discussed.

9:00

2aPA3. Impact of drift on the vehicle liftoff acoustic environments. Clothilde Giacomoni (All Points Logistics, LLC, M.S. ER42, Bldg. 4203, Huntsville, AL 35812, clothilde.b.giacomoni@nasa.gov) and R. Jeremy Kenny (Marshall Flight Ctr., NASA, Huntsville, AL)

During liftoff, a vehicle can drift due to wind, nozzle gimbaling, fly away maneuver, etc. This drift can cause the exhaust plumes to impinge on the deck and cause the noise levels experienced by the vehicle to increase. A small increase in the plume impingement can have a dramatic effect on the noise levels when the vehicle is only a few nozzle diameters from the deck. As the vehicle lifts off the deck the increase in noise levels lessens as the plume impingement increases. Several scale model acoustic tests have been undertaken at Marshall Space Flight Center which had test cases that were used to define the relationship between drift and the noise levels experienced by the vehicle.

9:20

2aPA4. Analysis of nozzle geometry effect on supersonic jet noise using Schlieren. Yuta Ozawa (Tokyo Univ. of Sci., Niijuku 6-3-1, Katsushika-ku, Tokyo 125-8585, Japan, ozawa@flab.isas.jaxa.jp), Akira Oyama (ISAS/JAXA, Sagamihara-shi, Kanagawa, Japan), Masayuki Anyoji (Kyusyu Univ., Kasuga-shi, Fukuoka, Japan), Akira Oyama (ISAS/JAXA, Sagamihara-shi, Kanagawa, Japan), Hiroya Mamori, Naoya Fukushima, Makoto Yamamoto, and Kozo Fujii (Tokyo Univ. of Sci., Katsushika-ku, Tokyo, Japan)

Strong acoustic waves emitted from rocket plume might damage to rocket payloads because rocket payloads consist of fragile structure. Therefore, it is important to predict acoustic directivity and reduce its intensity level. In this study, we conduct experiments of supersonic jet flows and investigate an influence of the nozzle geometry on acoustic waves by means of Schlieren method and microphone measurement. Three different nozzles are examined: a conical nozzle, a convergent-divergent nozzle (referred as C-D nozzle), and a tab-C-D nozzle. Tabs are equipped in the nozzle inside and turbulence is generated in the tab-C-D nozzle case. The Schlieren visualization shows that the strong shock trains are observed in the potential core of the jet for the conical nozzle case, while the shock waves are relatively weak since the nozzles are in the nearly ideal expanded condition in the C-D nozzle and tab-C-D cases. The distribution of near field OASPL (over all sound pressure level) obtained by microphone measurement shows strong directivity in the downstream direction for all the cases. This directivity seems to be the Mach wave radiation. Moreover, conical nozzle cases have strong acoustic intensity level caused by shock associated noise.

9:40

2aPA5. Visualization movie analysis of acoustic phenomena of a supersonic jet and Its comparison with intensity vectors. Masahito Akamine, Koji Okamoto (Graduate School of Frontier Sci., Univ. of Tokyo, Kashiwanoha 5-1-5, Okamoto lab., Dept. of Adv. Energy, Graduate School of Frontier Sci., Kashiwa, Chiba 277-8561, Japan, akamine@thermo.t.u-tokyo.ac.jp), Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Susumu Teramoto, Takeo Okunuki (Graduate School of Eng., Univ. of Tokyo, Bunkyo-ku, Tokyo, Japan), and Seiji Tsutsumi (Japan Aerosp. Exploration Agency, Sagamihara, Kanagawa, Japan)

The authors have studied the acoustic wave from an unheated, Mach 1.8 ideally expanded jet by using the acoustic intensity vector measurement and analysis of high-speed schlieren visualization movies (for example, the Fourier transform and wavelet-based conditional sampling). Both these techniques reveal the characteristics of the acoustic wave, such as the propagation direction and location of a source region. These techniques have their own advantages: the quantitative data can be obtained by using the acoustic intensity vector measurement, whereas the acoustic field including a close region to the jet can be visualized with high spatial resolution by using the Schlieren movie analysis. Therefore, their comparison is a meaningful approach to understand the acoustic phenomenon. This presentation compares these two techniques and describes what can be discussed with their comparison, considering the advantage and disadvantage of each measurement technique.

10:00–10:20 Break

10:20

2aPA6. Results of scale model acoustic tests using supersonic cold jets for H3 launch vehicle. Wataru Sarae, Atsushi Sawada, Keita Terashima (JAXA, 2-1-1 Sengen, Ibaraki, Tsukuba 305-8505, Japan, sarae.wataru@jaxa.jp), Takanori Haga, Seiji Tsutsumi (JAXA, Kanagawa, Japan), Tatsuya Ishii (JAXA, Tokyo, Japan), and Tetsuo Hiraiwa (JAXA, Miyagi, Japan)

Acoustic test using Mach 2 cold jet was conducted for the H3 launch vehicle currently being developed in Japan. Effect of the clustered engines and the newly built movable launcher on the lift-off acoustics was investigated. The overall acoustic level taken by the far-field microphones did not show proportional increase in the number of engines, especially for the angles corresponding to the Mach wave radiation from the free jets. The observation here disagrees with the empirical prediction model. The computational fluid dynamics was also employed to analyze the acoustic mechanism of clustered engines.

10:40

2aPA7. Comparison of spatial correlation parameters between full and model scale launch vehicles. Clothilde Giacomoni (All Points Logistics, LLC, M.S. ER42, Bldg. 4203, Huntsville, AL 35812, clothilde.b.giacomoni@nasa.gov) and R. Jeremy Kenny (Marshall Flight Ctr., NASA, Huntsville, AL)

The current vibro-acoustic analysis tools require specific spatial correlation parameters as input to define the liftoff acoustic environment experienced by the launch vehicle. Until recently, these parameters have not been very well defined. A comprehensive set of spatial correlation data were obtained during a scale model acoustic test conducted in 2014. From these spatial correlation data, several parameters were calculated: the decay coefficient, the diffuse to propagating ratio, and the angle of incidence. Spatial correlation data were also collected on the EFT-1 flight of the Delta IV vehicle which launched on December 5, 2014. A comparison of the spatial correlation parameters from full scale and model scale data will be presented.

11:00

2aPA8. Intensity-based laboratory-scale jet noise source characterization using the phase and amplitude gradient estimator method. Kent L. Gee, Tracianne B. Neilsen, Eric B. Whiting, Darren K. Torrie (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu), Masahito Akamine, Koji Okamoto (Graduate School of Frontier Sci., Univ. of Tokyo, Kashiwa, Chiba, Japan), Susumu Teramoto (Graduate School of Eng., Univ. of Tokyo, Bunkyo-ku, Japan), and Seiji Tsutsumi (Japan Aerosp. Exploration Agency, Sagamihara, Kanagawa, Japan)

A new method for the calculation of vector acoustic intensity from pressure microphone measurements has been applied to the aeroacoustic source characterization of an unheated, Mach 1.8 laboratory-scale jet. Because of the ability to unwrap the phase of the transfer functions between microphone pairs in the measurement of a broadband source, physically meaningful near-field intensity vectors are calculated up to the maximum analysis frequency of 32 kHz. This result improves upon the bandwidth of the traditional cross-spectral intensity calculation method by nearly an order of magnitude. The new intensity method is used to obtain a detailed description of the sound energy flow near the jet. The resulting intensity vectors have been used in a ray-tracing technique to identify the dominant source region over a broad range of frequencies. Additional aeroacoustics analyses provide insight into the frequency-dependent characteristics of jet noise radiation, including the nature of the hydrodynamic field and the sharp transition between the Mach wave and sideline radiation.

11:15

2aPA9. Spatial variation in similarity spectra decompositions of a Mach 1.8 laboratory-scale jet. Aaron B. Vaughn, Tracianne B. Neilsen, Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Brigham Young University, C-110 ESC, Provo, UT 84602, aaron.burton.vaughn@gmail.com), Koji Okamoto, and Masahito Akamine (Adv. Energy, Univ. of Tokyo, Kashiwa, Japan)

The primary source of noise from supersonic jets is turbulent mixing noise. Tam *et al.* [AIAA Paper 96-1716 (1996)] proposed similarity spectra for a two-source model of turbulent mixing noise corresponding to noise from omnidirectional fine-scale turbulence structures and directional large-scale turbulent structures. These empirical similarity spectra have been compared with reasonable success to the spectra of both military and laboratory-scale jets. Most application have looked at the variation in angle: fine-scale similarity spectra matches sideline of the jet nozzle, large-scale in the maximum radiation lobe, and a combination needed in between. A similarity spectra decomposition of from an ideally expanded, Mach 1.8 laboratory-scale jet allows for a spatial comparison between the near and far-field spectra. The sound from the convergent-divergent nozzle was collected at the Hypersonic and High-Enthalpy Wind Tunnel at Kashiwa Campus of the University of Tokyo at a variety of near, mid, and far field locations. Comparison of similarity spectra decompositions over distance yield insights into the sound propagation from this supersonic jet. In addition, results from a preliminary wavenumber eduction technique to build a wavepacket-based equivalent source model of the large-scale turbulent mixing noise. [Work supported by Office of Naval Research grant.]

11:30

2aPA10. Far-field acoustical measurements during QM-2 solid rocket motor static firing. Brent O. Reichman, Tracianne B. Neilsen (Brigham Young Univ., 453 E 1980 N, #B, Provo, UT 84604, brent.reichman@byu.edu), Won-Suk Ohm (Yonsei Univ., Seoul, South Korea), and Blaine M. Harker (Brigham Young Univ., Provo, UT)

The five-segment Space Launch System solid rocket motor was recently tested at Orbital ATK. Far-field acoustical measurements were performed at angles between 80° and 120° relative to the rocket exhaust at a distance of roughly 2500 m from the rocket, approximately 800 nozzle diameters. The angular aperture allows for evaluating spatial variation in acoustic properties and a comparison with similar tests in the past, including the 2015 test of the same rocket motor. Although terrain variations introduce uncertainty, an approximate 10 dB change in level is seen throughout the aperture, consistent with previous studies. In addition, at low frequencies a high degree of correlation is seen. Near the peak radiation direction high levels of derivative skewness indicate significant shock content and crackle. This dataset also presents the opportunity to test a new method for processing acoustic vector intensity [Thomas *et al.*, JASA **137**, 3366-3376 (2015)]. Comparison with the traditional method shows an increase in usable bandwidth of more than an order of magnitude.

Session 2aPP

Psychological and Physiological Acoustics: Linking Human Physiology with Psychoacoustics—From Brainstem to Cortical Measurements and Beyond

G. Christopher Stecker, Cochair
Hearing and Speech Sciences, Vanderbilt University, 1215 21st Ave. South, Room 8310, Nashville, TN 37232

Adrian KC Lee, Cochair
University of Washington, Box 357988, University of Washington, Seattle, WA 98195

Chair's Introduction—8:30

Invited Papers

8:35

2aPP1. Pupillary responses signify more than just effort: Windows into processing, prediction, reflection, and uncertainty. Matthew Winn (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, mwinn83@gmail.com)

Pupil dilation has long been used to index effort or general cognitive arousal in a variety of tasks. Traditionally, the main outcome measure is peak pupil dilation or the mean pupil dilation within some target window. In this talk, we explore how the time course of the growth and decay of pupil dilation can lend further insight into auditory processing. A series of pupillometry experiments revealed elevated listening effort resulting from various influences such as poor spectral resolution, unpredictability of sentence content, selective attention, or the need to reflect on what was heard to produce a creative response. Peak and mean pupillary responses generally confirmed original hypotheses, but more interesting findings emerge upon inspection of unforeseen trends in the data. Specifically, the timing of the pupil decay response can arguably be interpreted as a signature of the completion of sentence prediction, or the completion of processing that relies on memory trace (rather than perception) of the signal. The ultimate goal of this discussion is to unpack a new approach to pupillometry experiments that is aimed at separating processing that stems from perception versus processing that stems from cognitive processing, or reflecting upon a memory trace of the stimulus.

8:55

2aPP2. Electrophysiological correlates of auditory object binding with application to autism spectrum disorders. Hari M. Bharadwaj, Sheraz Khan, Matti Hämäläinen (Athinoula A. Martinos Ctr. for Biomedical Imaging, Massachusetts General Hospital, 149 Thirteenth St., Boston, MA 02129, hari@nmr.mgh.harvard.edu), and Tal Kenet (Dept. of Neurology, Massachusetts General Hospital, Boston, MA)

A fundamental problem in auditory neuroscience is to understand how the fragmented physiological representation of sound along the auditory pathway leads to unified percepts of broadband objects and streams. Natural sounds from a single physical source often exhibit strong temporal coherence between their acoustic components. We know from previous psychoacoustic studies that this temporal coherence is a cue for perceptual binding. Using magnetoencephalography (MEG), we measured neural responses to (1) a mixture of modulated tones, and (2) sentences of four-tone sine-wave speech, as we parametrically varied the temporal coherence between their components. We found that in parallel with behavior, MEG responses from the primary auditory cortex varied in with the temporal coherence of the stimulus. Further, cortical oscillatory signatures from the same region also varied with the stimulus coherence. These results suggests that signatures of temporal-coherence-mediated binding are already present at the level of the auditory cortex and may be mediated in-part by oscillatory thalamocortical processes. Finally, when we measured MEG responses in a cohort of children with autism spectrum disorders, we found a reduction in both the evoked responses and the oscillatory activity suggesting anomalous early cortical processing of complex sounds.

9:15

2aPP3. Relating binaural psychophysics to human physiology with functional magnetic resonance imaging: Opportunities and challenges. G. C. Stecker, Nathan C. Higgins, Sandra Da Costa (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu), and Susan A. McLaughlin (Univ. of Washington, Seattle, WA)

Functional magnetic resonance imaging (fMRI) has revolutionized the study of human perception in all sensory modalities. While early studies focused on identifying the brain regions involved with various processes ("where"), recent approaches aim to quantify relationships between brain activity and the physical or perceptual features of sensory stimuli ("how"). These modern approaches thus mirror the psychophysical goal of relating sensation to its underlying physical parameters, providing a clear means to connect human psychophysics and brain physiology. This presentation will describe a series of studies investigating the sensitivity of human auditory

cortex (AC) to binaural features of sounds and their perceptual correlates. Specifically, three questions will be addressed: (1) Does AC activity respond to the physical features or to the perception of binaural sound? (2) Do competing sounds sharpen or broaden the binaural selectivity of AC responses? (3) Does binaural sensitivity in human AC change across spatial, non-spatial, and non-auditory tasks? Questions (1) and (2) assess cortical correlates of psychophysical performance, elucidating the AC processes involved in spatial hearing. Question (3), however, goes a step further by addressing the potential effects of the psychophysical task itself. [Work supported by NIH R01 DC011548.

9:35

2aPP4. Switching of auditory attention. Adrian K. C. Lee, Susan A. McLaughlin, and Eric Larson (Univ. of Washington, Box 357988, Seattle, WA 98195, akclee@uw.edu)

Active listening in an acoustically crowded environment requires both maintaining and switching attention between auditory objects. General behavioral costs associated with task-switching are well documented in the cognitive literature. Yet there are many unanswered questions about the cost associated with switching between auditory objects relative to the type of feature-based attention employed. For example, do listeners use the same strategy to switch attention between locations as would be used to switch attention between speakers with different pitches? Is it equally difficult to switch attention within the same feature (e.g., attending to the left then the right talker) compared to switching across features (e.g., attending to the left talker then the one with the higher pitch)? Are the neural substrates recruited differentially depending on whether a listener is attending to spatial or non-spatial features of the auditory stimuli? In this talk, evidence from human psychophysical and physiological studies will be presented suggesting that auditory attention switching is not a monolithic construct. The interaction between top-down attention and different degrees of auditory stream degradation will also be discussed.

Contributed Papers

9:55

2aPP5. Effects of Auditory Attention on Otoacoustic Emissions. Heivet Hernandez Perez, Catherine McMahon (Linguist, Macquarie Univ., The Australian Hearing Hub, Sydney, NSW 2109, Australia, heivet.hernandez-perez@students.mq.edu.au), Sumitrajit Dhar, Sriram Boothalingam (Commun. Sci. and Disord., Northwestern Univ., Chicago, IL), David Poeppel (Dept. of Psych., New York Univ., New York, NY), and Jessica J. Monaghan (Linguist, Macquarie Univ., Sydney, NSW, Australia)

In humans, auditory efferent control of the cochlea has been implicated in sharpening of frequency tuning, improved detection of signals in noise, and the ability to switch the focus of auditory attention. However, it remains unknown whether the modulation of efferent activity during an auditory attention task depends on the degree of task difficulty. This study aimed to compare the suppression of otoacoustic emissions (OAEs), an objective measure of auditory efferent activation, during a lexical decision task with varying degrees of difficulty, compared with passive auditory listening. Ten normal-hearing 18-35 year-olds were assessed using monosyllabic words and non-words presented in a natural or noise-vocoded (less intelligible) condition. The participants were instructed to press a button every time they heard a non-word. Simultaneously, click evoked-OAEs were obtained from the contralateral ear to the speech stimuli with a probe in the external ear canal. Preliminary results showed that OAEs amplitudes were suppressed during the lexical decision tasks relative to passive listening. In addition, an effect of task difficulty was found, whereby the less intelligible condition showed stronger suppression. These data suggest that the auditory efferent system is recruited through auditory attention, and that this may play a role in speech perception.

10:10–10:30 Break

10:30

2aPP6. Frequency characteristics of neuromagnetic auditory steady-state responses at the supra-threshold level. Asuka Otsuka (Biomedical Res. Inst., National Inst. of Adv. Industrial Sci. and Technol. (AIST), Ikeda, Japan), Masato Yumoto (Graduate School of Medicine, The Univ. of Tokyo, Tokyo, Japan), Shinya Kuriki (Res. Ctr. for Adv. Technologies, Tokyo Denki Univ., Inzai, Japan), and Seiji Nakagawa (Ctr. for Frontier Medical Eng., Chiba Univ., 1-33 Yayoicho, Inage-ku, Chiba, Chiba 263-8522, Japan, s-nakagawa@chiba-u.jp)

Auditory steady-state response (ASSR) is a neuronal electrical component recorded as a continuous sinusoidal signal phase-locked to a modulation frequency of sounds. Since ASSR amplitude shows high correlation with perceptual characteristics at hearing threshold level, the ASSR has been examined as a candidate for an objective index of the human hearing ability per frequency. However, the relationship between ASSR and perception at the supra-threshold level has not been clarified. In this study, characteristics of ASSR magnitude relative to loudness at the supra-threshold level were investigated. Neuromagnetic 40-Hz ASSR was recorded in response to sinusoidally amplitude-modulated sweep tones with carrier frequency covering the frequency range of 0.1-12.5 kHz. Sound intensity was equalized at 50-, 60-, and 70-dB SPL with an accuracy of $\pm$0.5-dB SPL at the phasic peak of the modulation frequency. Corresponding loudness characteristics were modeled by substituting the detected individual hearing thresholds into a standard formula (ISO226:2003(E)). The strength of the ASSR component was maximum at 0.5 kHz, and it decreased linearly on logarithmic scale toward lower and higher frequencies, whereas loudness model was plateaued between 0.5 and 4 kHz. The results indicated that frequency characteristics of the ASSR were not equivalent to those of SPL and loudness model.

10:45

2aPP7. Auditory selective attention in cochlear implant users. Inyong Choi, Subong Kim (Dept. of Commun. Sci. and Disord., Univ. of Iowa, 250 Hawkins Dr., Iowa City, IA 52242, inyong-choi@uiowa.edu), and Phillip E. Gander (Dept. of Neurosurgery, Univ. of Iowa, Iowa City, IA)

Selective attention is a fundamental process for our communication in noisy everyday environments. Previous electrophysiologic studies have shown that selective attention modulates the neural representation of the auditory scene, enhancing responses to a target sound while suppressing the background. Given that most cochlear implant (CI) users complain about difficulty understanding speech within background noise, we investigated whether CI users' selective attention performance and underlying neural processes are 1) degraded compared to normal-hearing (NH) listeners and 2) variable among individuals. Using speech stimuli recorded from two competing speakers (male and female), we measured 12 CI and 17 NH listeners' selective attention performance while their cortical neural processes were simultaneously recorded with high-density electroencephalography. While both groups showed above-chance level performance and significantly larger evoked responses to attended stimuli in auditory cortex, the CI group's selective attention performance and neural gain control were poorer than those of NH listeners. Furthermore, a positive correlation was found between the attentional modulation of evoked cortical responses and behavioral performance in the CI group. This result suggests that degraded selective attention processes contribute to an explanation of CI users' difficulty in real-world communication.

11:00

2aPP8. Effects of noise on a person's psychological state under task-loaded condition from the viewpoint of brain function analysis. Takeshi Akita, Naoko Sano (Dept. of Sci. and Technol. for Future Life, Tokyo Denki Univ., 5 Senju-Asahi-cho Adachi-ku, Tokyo 1208551, Japan, akita@cck.dendai.ac.jp), and Ayako Matsuo (Graduate School of Adv. Sci. and Technol., Tokyo Denki Univesity, Tokyo, Japan)

Measurement and analysis of brain function by near infrared spectroscopy (NIRS) obtained from the frontal area of scalp can reveal some aspects of psychological state of human. In the present research, investigation of effects of noise on task-loaded person is attempted using analysis of NIRS. Two experiments are conducted. In the first experiment, brain function of frontal regions is measured under resting and task-loaded condition with back ground noise to confirm that task-loaded person uses their frontal area of brain. The results show that subjects use their frontal region of brain sufficiently while performing their tasks. In the second experiment, subjects carry out their calculating tasks under several noise conditions. Analyses of the data show that activity of frontal brain area diminishes. It is suggested that noise interrupt subjects' thinking activity for carrying out their task. These results suggest that measurement and analysis of NIRS is available for clarifying the negative effects of noise on task-loaded person. Using these methods can lead to create an index of distracting aspects of noise.

11:15

2aPP9. An objective measure of temporal fine structure correlates with perception in a speech discrmination task. David McAlpine, Macarena Bowen, and Jaime Undurraga (Linguist, Macquarie Univ., Australian Hearing Hub, 16 University Ave., Sydney, NSW 2109, Australia, david.mcalpine@mq.edu.au)

Sensitivity to the temporal fine structure (TFS) of sounds is considered important for understanding speech, especially in the presence of background noise, but few studies have employed objective measures to assess the relationship between sensitivity to TFS and listening to speech. Here, we used EEG (electroencephalography) to obtain an objective measure of sensitivity to the TFS of low-frequency sounds (520-Hz amplitude-modulated tones) in listeners who also undertook a speech-discrimination task (discrimination of vowel-consonant-vowel, VCV, sounds in background noise). Two different sound levels (60 and 80 dB SPL) and three different signal-to-noise ratios (SNRs; -6, 0 and + 6 dB) were used. Three symmetric interaural phase modulations (IPMs), were measured: -60°/60°, -90°/90°, and -120°/120°. A fourth condition used an IPM switching between 0° and 180° degrees. IPM-FRs were higher for the 0/180° degrees than all other conditions. Employing the ratio of the 3 symmetric to the asymmetric (0/180°) IPM-FR as a measure of the neural dynamic range of TFS processing, we found that VCV scores increased with age at 60 dB SPL, but not 80 dB SPL, and indeed were better at 60 compared to 80 dB SPL in older listeners, and correlated with a higher ratio of IPM-FRs.

TUESDAY MORNING, 29 NOVEMBER 2016 — SEA PEARL, 8:00 A.M. TO 11:25 A.M.

Session 2aSA

Structural Acoustics and Vibration and Physical Acoustics: Acoustic Metamaterials I

Christina J. Naify, Cochair
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Invited Papers

8:00

2aSA1. Acoustic topological insulators. Romain Fleury (Langevin Inst., ESPCI Paris Tech, 1 University Station C0803, Austin, TX 78712, romain.fleury@utexas.edu)

The recent discovery of topological states in condensed matter physics has spawned a quest for similar effects in classical physics. Topological insulators are bulk insulators supporting bandgap crossing, chiral edge states that are immune to backscattering caused by defects or disorder. Here, we present the acoustic equivalent of topological insulators, obtaining the analog of the Quantum Hall Effect (QHE) and the Quantum Spin-Hall Effect (QSHE) in acoustic metamaterials with, respectively, broken or preserved time-reversal symmetry. We demonstrate how our recent results in the field of non-reciprocal acoustics [cf. Science **343**, 516 (2013); Ac. Today. **11**, 14 (2015)] can be used to induce an acoustic version of the QHE in active metamaterials with broken time-reversal symmetry. We also put forward resonant and non-resonant schemes to emulate the Kane-Mele Hamiltonian in completely passive acoustic metamaterials, obtaining the acoustic analog of the QSHE. Our work reveals that topological acoustic metamaterials enable fundamentally new ways of manipulating sound, including one-way, chiral propagation of acoustic waves over broad bandwidth and topological protection against geometry imperfections, impedance mismatches, crystal defects, and local disorder.

8:20

2aSA2. Breaking time reversal symmetry with coriolis mean flow systems. Farhad Farzbod (Dept. of Mech. Eng., Univ. of MS, 201A Carrier Hall, University, MS 38677, farzbod@olemiss.edu) and Michael J. Leamy (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

One way to break time reversal symmetry is to have a moving wave propagation medium. If the acoustic wave vector and the moving fluid velocity are collinear, such as the one created by Fluery et al. (Science, **343**, 2014) we can use the wave vector shift caused by the fluid flow to break reciprocity. An alternative approach we have taken is to use a fluid velocity field which enters the differential equation of the system as a cross product term with the wave vector. A circular field in which the fluid velocity increases radially has a Coriolis acceleration term. In this system, the acoustic wave enters from the central wall and exits from the perimeter wall. Equations of conservation of mass and momentum, after linearization can be simplified to: $(\partial/\partial t+\mathbf{V}.\nabla)P+\rho C^2\nabla.(\mathbf{u}+\mathbf{V})=0$, $\rho\partial/\partial t(\mathbf{u}+\mathbf{V})+\rho(\mathbf{V}.\nabla)\mathbf{u}+\rho(\mathbf{u}.\nabla)\mathbf{V}+\rho(\mathbf{V}.\nabla)\mathbf{V}+\nabla P=0$, in which $\mathbf{u}$ is the particle acoustic velocity and $\mathbf{V}=r\omega\mathbf{e}_\theta$is the fluid velocity due to the rotational velocity field. It can be shown that the third term in the second equation can be simplified as:$\rho\omega\mathbf{e}_k\times\mathbf{u}$. This term Coriolis acceleration induces nonreciprocity. In this work, we solved the differential equation numerically and investigated the effect of fluid velocity on the nonreciprocity factor.

Contributed Papers

8:40

2aSA3. Unidirectional wave transmission in a diatomic acoustic metamaterial. Kwek Tze Tan and Bing Li (Mech. Eng., The Univ. of Akron, Mech. Eng. 101 ASEC, Akron, OH 44325, ktan@uakron.edu)

In this study, we propose the use of dual resonators to design a linear diatomic metamaterial, consisting of several small-sized unit cells, to realize large unidirectional wave transmission in low frequency domain (below 1 kHz). The asymmetric transmission mechanism is theoretically derived and systematically investigated. Numerical results are verified using both lattice structures and continuum models. This passive system does not require external energy supply or result in any frequency conversion. Various wave transmission band gaps, including passing band, stopping band gap, and asymmetric transmission band, can be theoretically predicted and mathematically controlled, which extends the design concept of unidirectional transmission devices. The asymmetric transmission band can be easily altered by careful and deliberate design of the unit size geometrical parameters and material properties. This proposed idea offers a new design concept in wave energy control through unidirectional systems, and may be applied to practical applications such as ultrasound imaging, directional transducer, and noise control.

8:55

2aSA4. Loss-induced asymmetric transmission through gradient-index metasurfaces. Chen Shen (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., Eng. Bldg. 3, Campus Box 7910, Raleigh, NC 27695-7910, cshen5@ncsu.edu), Yong Li (Université de Lorraine, Institut Jean Lamour, Vandœuvre-lès-Nancy, France), and Yun Jing (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

Gradient-index metasurfaces have shown a great potential for wavefront modulation and transmission/reflection control. However, the effect of loss is rarely discussed in acoustic metasurfaces. Here, we study the effect of loss in gradient-index metasurfaces and suggest its application in asymmetric transmission. For a gradient-index metasurface composed of six types of unit cells, theoretical analysis reveals that, when loss is considered, the transmission coefficients for different diffraction modes will have different responses for acoustic waves with oblique incidence. The gradient-index metasurface thus can be utilized to achieve asymmetric transmission. Numerical simulations based on effective medium and real structure are performed to validate the theoretical findings using a single layer of gradient-index metasurface. The transmission contrast can be greater than 20 times in simulations within a certain range of incident angles. This design may be useful in ultrasonic and noise control applications.

9:10

2aSA5. A transparent broadband metallic lens for focusing underwater sound. Xiaoshi Su and Andrew N. Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, xiaoshi.su@rutgers.edu)

Diffractive lenses usually suffer from spherical aberration. Even though it is possible to minimize aberration by design, the quality of focused sound is still limited owing to the side lobes caused by aperture diffraction. In recent years, hyperbolic secant index profiles have been widely used in gradient index (GRIN) metamaterials design to reduce aberration and suppress side lobes. Here, we introduce a 1D transformation to modify the secant profile for further aberration reduction, and present a GRIN lens comprised of hexagonal unit cells obeying the modified profile for focusing underwater sound in the near field. The transversely isotropic unit cells are designed with water-like acoustic properties using foam mechanics and optimized using a homogenization technique based on finite element method. The in-plane shear modulus of the unit cells are minimized to suppress the shear modes in the GRIN lens. Moreover, the impedances of the unit cells are tuned close to water, such that most of the energy carried by the normally incident plane wave propagate across the interface between the lens and water. This lens can focus underwater sound with high efficiency over a broadband (10 kHz—40 kHz).

9:25

2aSA6. Experimental validation of the underwater sound focusing properties of a pentamode gradient index metamaterial lens. Colby W. Cushing, Michael R. Haberman, Preston S. Wilson (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, colby.cushing@utexas.edu), Xiaoshi Su, and Andrew N. Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

A gradient index (GRIN) acoustic lens is a material slab with spatially varying properties that refracts an incident acoustic wave to produce a desired transmitted field. Hyperbolic secant index profiles have widely been used to create focusing lenses with minimal aberration and good sidelobe suppression. This work reports on the underwater experimental characterization of a GRIN lens that employs a modified secant profile for improved aberration reduction when compared to the hyperbolic secant profile. The index profile is produced using a two-dimensional pentamode material created from air-filled aluminum hexagonal unit cells. The lens is fabricated from waterjet cut aluminum plates that are stacked and sealed together. The test apparatus is comprised of a custom plane-wave source, an automated horizontal planar scanner, a pentamode GRIN lens, a custom support structure, and a calibrated hydrophone and associated electronics. The source was tested for amplitude planarity across its aperture at a constant distance to confirm its usability for this system. Experimental results of lens performance over the frequency range of 10-40 kHz are compared with the isolated source and model predictions. [Work supported by ONR.]

9:40

2aSA7. Spatial acoustic modulator for projecting high-quality holographic image. Bin Liang, Jianchun Cheng, Yifan Zhu (Inst. of Acoust., Nanjing Univ., P. R. China, 22 Hankou Rd., Nanjing, Jiangsu 210093, China, liangbin@nju.edu.cn), Xuefeng Zhu (School of Phys., Huazhong Univ. of Sci. and Technol., Wuhan, China), Xinye Zou, and Jing Yang (Inst. of Acoust., Nanjing Univ., P. R. China, Nanjing, China)

Spatial modulation of acoustic wave has demonstrated theoretical and experimental importance in the field of acoustics, with wide applications such as in medical ultrasound and particle manipulation. However, the existing means for acoustic modulation have to rely on the acoustic metasurfaces providing only pure phase modulation or active element arrays with complicated fabrications and limited spatial resolutions. In this work, we present the design, numerical simulation, and experimental demonstration of a spatial acoustic modulator (SAM) capable of manipulating the phase and amplitude profiles separately and continuously, offering the possibility to address the above challenges by enabling arbitrarily complex acoustical patterns. The proposed structure has a simple configuration with planar profile, high efficiency of spatial modulation, and deep-subwavelength unit scale, ensuring easy and low-cost fabrication and nearly-continuous wavefront manipulation. The designed SAM is used to project high-quality acoustic holograms both theoretically and experimentally to demonstrate its wave-steering functionality. Our metastructure-based holographic projector with simplicity and extreme acoustic performance advances the capability of spatial acoustic modulation, and have various applications in places where the conventional techniques would lead to complexity and limited ability.

9:55–10:10 Break

10:10

2aSA8. Quantised acoustic meta-surfaces using metamaterial bricks. Gianluca Memoli (Dept. of Eng., Computing and Design, School of Informatics, Eng. and Design, Univ. of Sussex, Brighton BN1 9RH, United Kingdom, g.memoli@sussex.ac.uk), Mihai Caleap (Eng., Univ. of Bristol, Bristol, United Kingdom), Michihiro Asakawa, Deepak Sahoo (School of Informatics, Eng. and Design, Univ. of Sussex, Brighton, United Kingdom), Bruce W. Drinkwater (Eng., Univ. of Bristol, Bristol, United Kingdom), and Sriram Subramaniam (School of Informatics, Eng. and Design, Univ. of Sussex, Brighton, United Kingdom)

Acoustic meta-surfaces offer an extremely successful way of transforming an input acoustic wave into almost any diffraction-limited field. This is typically achieved by encoding a phase delay function on the unit cells that form the meta-surface, which is then manufactured by rapid prototyping. One of the limits of current technology is that meta-surfaces built in this way are targeted to a single, specific application. In this work, we present a method of building acoustic meta-surfaces using metamaterial bricks, designed to operate at 40 kHz and quantized both in the space and in the phase domains. First, we present finite-elements simulations and experiments to discuss key parameters of our bricks as frequency response, directivity, impedance. Second, our metamaterial bricks are used here to achieve some key beam transformations like steering and focusing, but also more complex fields like an acoustic tractor beam. We use the latter to levitate one or more polystyrene beads and diffraction theory to explain some of the findings. Our method, where the phase in a specific position may be changed by changing the specific brick, may find applications in loudspeaker design, ultrasound imaging/therapy, or acoustic particle manipulation.

10:25

2aSA9. Nonlinear wave propagation and rotational dynamics In microscale granular media. Samuel Wallen and Nicholas Boechler (Mech. Eng., Univ. of Washington, Mech. Eng., Stevens Way, Box 352600, Seattle, WA 98195, walles@uw.edu)

Granular media have been shown to exhibit complex dynamics stemming from dispersion and nonlinear particulate interactions. Studies of wave propagation in granular materials composed of ordered arrays of spherical particles interacting via Hertzian contact, known as "granular crystals," have demonstrated rich nonlinear phenomena, including amplitude-dependent and tunable sound speeds, solitary waves, and discrete breathers. However, most of the past work on granular crystals has considered macroscale spheres, with diameters on the order of centimeters or millimeters. More recent works have explored wave phenomena in microscale granular crystals, having particle diameters of a few microns, which can be fabricated using self-assembly methods, and characterized experimentally using laser ultrasonic techniques. The physics of microscale granular media have been shown to vary drastically from their macroscale counterparts; for example, interparticle adhesion, which is negligible at macroscale, creates an intrinsic static load, which linearizes the contact forces at low amplitude, and can increase the importance of shear interactions and particle rotations. In this work, we present theoretical and computational analyses of nonlinear wave propagation in microscale granular crystals, examining the interplay between nonlinearity and rotational dynamics. In particular, we focus on amplitude-dependent distribution of energy into the translational and rotational modes of a hexagonally close-packed granular crystal.

10:40

2aSA10. Metamaterials made of granular chains for nondestructive evaluation applications. Piervincenzo Rizzo and Amir Nasrollahi (Civil and Environ. Eng., Univ. of Pittsburgh, 942 Benedum Hall, 3700 O'Hara St., Pittsburgh, PA 15261, pir3@pitt.edu)

In the last two decades it has been demonstrated that highly nonlinear solitary waves (HNSWs) propagating in metamaterials made of chains of granular particles can be used in many physics and engineering applications, including acoustic lenses, impurity detectors, and nondestructive evaluation (NDE). HNSWs are compact nondispersive waves, and in this paper, we demonstrate their application for the nondestructive evaluation of civil, mechanical, and aerospace structures. In particular, this presentation delves with the non-invasive assessment of concrete strength and the evaluation of thermal stress in slender beams. For the concrete application, the NDE method exploits the dynamic interaction between the metamaterial and the concrete, and the hypothesis is that this interaction depends on the stiffness of the specimen being inspected. The results show that the time of flight of the HNSWs is inversely proportional to the modulus of elasticity of the concrete. For the buckling application, we present the coupling mechanism between a beam and the HNSWs propagating along a straight granular chain in contact with the beam. We show that the waves can be used to measure the stress of thermally loaded structures or to infer the neutral temperature, i.e., the temperature at which this stress is null.

10:55

2aSA11. Enhanced energy dissipation with nonlinear snapping acoustic metamaterial inclusions. Stephanie G. Konarski, Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, skonarski@utexas.edu)

Through the design of subwavelength structures, nonlinear acoustic metamaterials can be engineered to exhibit regimes of positive and negative incremental stiffness. Negative incremental stiffness leads to large, rapid deformation, commonly called snapping, between local stable configurations. When perturbed by a small external pressure, snapping elements can be exploited to enhance energy absorption of acoustic waves. The present research focuses on a low volume fraction of snapping inclusions, dispersed within a nearly incompressible viscoelastic matrix to create a multiscale nonlinear material. The forced dynamic response of the heterogeneous material is modeled using a generalized Rayleigh-Plesset equation, which couples the dynamic behavior at the micro- and macroscale. We investigate the nonlinear dynamic behavior of the heterogeneous medium for small harmonic perturbations about several pre-strains of the snapping inclusions to demonstrate the influence of the microscale dynamic response on the macroscale response. Of primary interest are energy dissipation capabilities and characteristic acoustic impedance, which change with inclusion pre-strain. The behavior is also compared to heterogeneous media created from conventional inclusions, such as air voids and steel inclusions, to demonstrate the wide range of material behavior obtained using snapping inclusions. [This work was supported by the Office of Naval Research.]

11:10

2aSA12. Underwater acoustic absorption using a periodically voided soft elastic medium. Gyani Shankar Sharma (School of Mech. and Manufacturing Eng., UNSW Australia, Rm. #408, Sydney, New South Wales 2052, Australia, gyanishankar.sharma@student.unsw.edu.au), Alex Skvortsov, Ian MacGillivray (Maritime Div., Defence Sci. and Technol. Group, Melbourne, VIC, Australia), and Nicole Kessissoglou (School of Mech. and Manufacturing Eng., UNSW Australia, Sydney, New South Wales, Australia)

A coating comprising periodic cavities in a soft rubber medium can significantly reduce the reflection of acoustic waves from a steel plate. In this work, analytical and numerical models are developed to study acoustic reflection and transmission through a single layer of periodic cylindrical voids embedded in a soft elastic medium with steel backing. The analytical model is based on effective medium approximation theory which allows modeling of a composite material as a homogeneous material using effective material properties. The numerical model is based on the finite element method and developed using commercial finite element code COMSOL Multiphysics. Minimal reflection and transmission and maximal absorption of sound waves in a broad frequency range is attributed to the monopole type resonance of voids. The monopole resonance frequency of voids in an array is shown to be significantly higher compared to the monopole resonance of a single void in an infinite medium due to resonance coupling. An analytical framework to predict the resonance frequency of voids in an array is presented. The effect of strong and weak coupling of void resonances on reflection and transmission characteristics is investigated. Non-dimensionalized results obtained from the analytical and numerical models are compared.

TUESDAY MORNING, 29 NOVEMBER 2016 CORAL 4, 9:00 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication: Cross-Linguistic Speech Production and Perception

Katie Drager, Cochair
Linguistics, University of Hawaii at Manoa, 1890 East-West Rd., More 569, Honolulu, HI 96822

Timothy Vance, Cochair
Dept. of Linguistic Theory and Structure, Natl. Inst. for Japanese Lang. and Ling., Tokyo, Japan

Chair's Introduction—9:00

Invited Papers

9:05

2aSC1. Articulatory settings of high- and low-proficiency second-language speakers. Ian Wilson (CLR Phonet. Lab, Univ. of Aizu, Tsuruga, Ikki-machi, Aizuwakamatsu, Fukushima 965-8580, Japan, wilson@u-aizu.ac.jp)

Articulatory setting, a language-specific underlying position of the articulators during speech, can be measured by using inter-speech posture (ISP)—the speech rest position. In an x-ray study, it was shown to behave like a target posture, in that it has as little variability as /i/, one of the least variable vowels. In an rt-MRI study, it was found to be more mechanically advantageous than absolute rest position. In an ultrasound/Optotrak study, results showed differences across normalized monolingual groups of speakers, and bilinguals who were perceived to be native in each of their languages had a different articulatory setting for each language. In an EMA study, tongue width in the coronal plane during ISP was found to be less efficient in second-language (L2) speakers of English. In recent research using ultrasound and lower-proficiency Japanese-English bilinguals, we were surprised to find differences in their articulatory settings for each of their languages, when it would have been simpler for such a speaker to use the articulatory setting of their dominant language while speaking their non-dominant one. In this talk, a state-of-the-art analysis of research on articulatory setting will be given including recent research on Japanese-English bilinguals.

9:25

2aSC2. Language contact and accent changes in Japanese. Haruo Kubozono (Theory & Typology Div., National Inst. for Japanese Lang. and Linguist, NINJAL, 10-2 Midori-cho, Tachikawa 190-8561, Japan, kubozono@ninjal.ac.jp)

This talk discusses how word accent patterns of regional Japanese dialects have changed over the years as they were heavily exposed to standard Tokyo Japanese. A particular attention is paid to the accent changes that have taken place in Kagoshima Japanese, a southern dialect whose accent system is strikingly different from that the standard variety. Original fieldwork studies show that this regional dialect has been heavily influenced by the standard variety with respect to the presence or absence of a pitch fall, while the position of the fall is still determined by its traditional system: e.g., /doo.na.TU/ → /doo.NA.tu/ "donut" (Tokyo form: /DOo.na.tu/). Interestingly, the same mechanism is at work when (Tokyo) Japanese borrow words from English as loanwords: the recipient language copies the

presence or absence of a pitch fall from the host language but it follows its own accent rule with respect to the position of the fall, e.g. /ba'.na.na/ "banana," /wa.shi'n.ton/ "Washington." These observations indicate that native Japanese speakers are highly sensitive to the presence or absence of a pitch fall when listening to a foreign language and a different variety of their native language, but are quite insensitive to the position of the pitch change.

9:45

2aSC3. Effects of prosodic focus on discourse processing by native speakers versus second language learners. Amy J. Schafer (Dept. of Linguist, Univ. of Hawaii, 1890 East-West Rd., Manoa, Honolulu, HI 96822, amy.schafer@hawaii.edu)

The acquisition of sentence-level prosody and intonation is often challenging for second language learners (L2ers). A speaker's native and second language can differ at several levels of representation involved in the use of prosody, and so multiple new mappings must be acquired for proficiency in the second language's prosodic system. This talk examines prosodic focus in English, Japanese, and Korean, and its effects on discourse processing by native speakers and L2ers. Although all three languages make use of prosodic focus, and draw on some of the same acoustic cues to realize it, there are also substantial differences across the languages in how prosodic focus interfaces with other grammatical components. Results from a series of experiments indicate context-sensitivity in how L2ers respond to prosodic focus, showing native-like responses in one discourse context but quite different responses to the same tonal realization in another context. Such findings point to the importance of testing the perception of prosody across multiple grammatical configurations, particularly when investigating language acquisition.

10:05

2aSC4. Phonological traits of the Japanese and English spoken by Nikkei speakers in Hawaii. Daniel Long (Japanese Linguist, Tokyo Metropolitan Univ., 1-1 Minami Osawa, Hachioji, Tokyo 1920364, Japan, dlong@tmu.ac.jp)

The Japanese and English language varieties spoken by Nikkei speakers in Hawaii show phonological traits which result from language contact and are uncommon in other varieties. Characteristics of the Japanese spoken by Nikkei speakers which differ from native varieties of the language include the following. Non-geminate and geminate consonants are found in variation ([kappamaki] vs. [tekkamaki], [napa]) and speakers do not hear germination as being phonemically significant. The English labiodental fricative [f] is substituted for the Japanese bilabial fricative [ϕ] ([furo], [furikake]). Vowel lengthening is in inconsistent and not phonemically significant ([obatʃan] contrasts with [oba:tʃan], but [zori] has shortened vowel). Characteristics of Hawaiian English spoken by Nikkei speakers (as well as other ethnicities) include the following. The alveolar flap /ɾ/ ([kaɾai] 'spicy') is a phoneme contrasted with /l/ and /r/. Phonotactic rules allow for the /ts/ sequence even in the word-initial position (Hawaiian English [tsunami] vs. General American English [sunami]). In this paper, I will use PRAAT acoustical analyses to illustrate these traits.

10:25–10:40 Break

10:40

2aSC5. Variation in the second formant: Its role in a creole's coarse-refined contrast. Nala H. Lee (Dept. of English Lang. and Lit., National Univ. of Singapore, Block AS5, 7 Arts Link, Singapore 117570, Singapore, elllhn@nus.edu.sg)

Baba Malay encodes an interesting coarse-refined contrast. Speakers overtly recognize words ending with -al, -aɾ, and -as (e.g., bakaɾ "burn") as *kasar* "coarse," while their counterparts ending with -ɛ (e.g., bakɛ "burn") are *halus* "refined." The contrast is neither phonetic nor is it constrained by phonology or morphology. The coarse-refined system may be based on sound symbolism. The frontness of -ɛ is associated with a smaller articulatory space in the oral cavity, and hence refinedness, while the more backwards al, -aɾ, and -as forms, are associated with a larger articulatory space and hence coarseness. A perception experiment, mainly a within-subjects matched guise task, is conducted. Coarse and refined pairs are elicited from speakers. The relevant F2 of both variants are adjusted twice upwards and twice downwards in steps of 100 Hz. Listeners rate these forms on a scale of 1 to 5, 5 being most associated with values that are "refined." Results show that the lower F2 is, the more likely listeners are to associate the coarse guise with "coarse" values, and the higher F2 is, the more likely listeners are to associate the refined guise with "refined" values. The *kasar-halus*contrast is thus mitigated by sound symbolism operationalized by F2.

11:00

2aSC6. Hailu Hakka and Taiwan Min tone changes. Ho-Hsien Pan, Shao-Ren Lyu, and Hsiao-Tung Huang (Dept. of Foreign Lang. and Literatures, National Chiao Tung Univ., 1001 University Rd., Hsinchu 300, Taiwan, hhpan@faculty.nctu.edu.tw)

A pitch contour overlap and a tone-sandhi relationship can both contribute to confusion in tone that lead to change in tone. This study investigates the duration, f0, and phonation between Hailu Hakka and Taiwan Min tone pairs that (1) were neither overlapped in pitch contours nor tone-sandhi related, such as Hakka tones 55 and 33, and Min tones 55 and 13; or (2) were overlapped in pitch but not tone-sandhi related, such as Hakka 33 and 11, and Min 13 and 31; or (3) were not overlapped in pitch but tone-sandhi related, such as Hakka and Min tones 3 and 5, or (4) were both overlapped in pitch contour and tone-sandhi related, such as Hakka 24 and 33, and Min tones 31 and 33. The results indicated that Hakka speakers under 30 years of age produced tones 24 and 33 that were overlapped in pitch and tone-sandhi related, and with similar pitch contour; whereas, Min speakers, regardless of age, produced tones 3 an 5 that were not overlapped in pitch but were sandhi-related bi-directionally (3$\leftrightarrow$5) similarly. It is proposed that sound changes occur among sandhi related tones with and without overlap in pitch contours.

11:20

2aSC7. Substrate effects in the phonetic realization of Pidgin vowels. James Grama (Modern Lang., Santa Monica College, 18933 Nashville St., Los Angeles, CA 91326, james.grama@gmail.com)

Hawai'i is home to an English-lexified creole, known locally as Pidgin. While much research has focused on the structural development of Pidgin (e.g., Bickerton & Odo 1976), little work has investigated acoustic phonetic variation in the Pidgin vowel system. This talk presents results from an analysis of 854 tokens of /a/ produced by 32 speakers of Pidgin from two corpora—one from the 1970s and one from the 2000s—and investigates factors that contribute to differences in realizations of this vowel. Results indicate that /a/ in Pidgin exhibits a raised nucleus over real time for female speakers ($p<0.001$) but not male speakers. Additionally, analysis reveals that the youngest group of Pidgin speakers in the 2000s corpus realize /a/ in words of Hawaiian origin as higher in the vowel space than words of English origin ($p<0.0001$). Possible reasons for the emergence of this fine-grained phonetic difference in the youngest speaker group are discussed, including the role played by Hawaiian language education and the apparent valuation of Hawaiian. Taken together, results suggest that a combination of factors influences the realization of /a/ in Pidgin, including age, gender, and whether a word derives from the superstrate language (=English) or a substrate language (=Hawaiian).

11:40

2aSC8. Voice onset time and closure duration of word-initial voiceless plosives in Pidgin conversation

. Katie Drager (Linguist, Univ. of Hawaii at Manoa, 1890 East-West Rd., More 569, Honolulu, HI 96822, kdrager@hawaii.edu), James Grama (Santa Monica College, Santa Monica, CA), Lorenzo Gonzalez (Linguist, Univ. of Hawaii at Manoa, Orlando, FL), and Cassidy Copeland (Linguist, Univ. of Hawaii at Manoa, Honolulu, HI)

Hawai'i is home to many languages, including an English-lexified creole, known locally as Pidgin. Despite much work on the development of Pidgin, little is known about the acoustic properties of Pidgin consonants. This talk presents results from an analysis of word-initial plosives produced by eight speakers of Pidgin across two age groups, investigating what factors influence VOT and closure duration in the plosives. The data were analyzed using linear regression models with by-speaker random intercepts. Results demonstrate that VOT in word-initial plosives tends to be shorter in Pidgin than in English, with mean duration values nearly half those reported for English in Yao (2007). The effects of social and linguistic factors, however, tend to be similar (e.g., shorter VOT for older speakers, VOT for /k/ is longer than for /t/ or /p/). Results also indicate that VOT is significantly shorter both for words of Hawaiian origin (β=-0.01, $t(289)$=-2.0, and $p<0.05$) and—in a separate model fit to a subset of the data with only non-Hawaiian words—highly frequent words (β=-0.01, $t(216)$=-4.2, and $p<0.001$). In contrast, no factors tested significantly predicted closure duration. Taken together, results demonstrate that a combination of linguistic, social, and probabilistic factors influence VOT in Pidgin.

Session 2aSPa

Signal Processing in Acoustics: Compressive Sensing in Acoustics I

Peter Gerstoft, Cochair
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Jeffrey S. Rogers, Cochair
Acoustics Division, Naval Research Lab, 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375

Yoshifumi Shiraki, Cochair
Communication Science Laboratories, Nippon Telegraph and Telephone Corp., 3-1, Morinosato-Wakamiya, Atsugi-City, Kanazawa pref. 243-0198, Japan

Chair's Introduction—7:55

Invited Papers

8:00

2aSPa1. Introduction to sparse and compressive sensing. Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gillman Dr., La Jolla, CA 92093-0238, gerstoft@ucsd.edu)

Compressed sensing or sparse sampling is a signal processing technique for efficiently acquiring and reconstructing a signal, by finding solutions to underdetermined linear systems. I will describe the basic features of compressive sampling that has many opportunities in acoustics.

8:20

2aSPa2. A brief introduction to distributed compressed sensing and its typical reconstruction limits. Yoshifumi Shiraki (Commun. Sci. Labs., Nippon Telegraph and Telephone Corp., 3-1, Morinosato-Wakamiya, Atsugi-City, Kanazawa pref. 243-0198, Japan, shiraki.yoshifumi@lab.ntt.co.jp) and Yoshiyuki Kabashima (Dept. of Computational Intelligence and Systems Sci., Interdisciplinary Graduate School of Sci. and Eng., Tokyo Inst. of Technol., Yokohama, Japan)

A brief introduction to compressed sensing and a discussion of its reconstruction limit analysis are presented. In addition, we briefly introduce the replica method, a technique in statistical mechanics, to show its effectiveness for typical performance analysis on signal processing problems. In this presentation, we especially focus on a signal model called joint sparse model 2 (JSM-2) or the multiple measurement vector problem, in which all sparse signals share their support. The signal model JSM-2 is important for dealing with practical processing problems. We investigate the typical performance of l2,1-norm regularized minimization and the Bayesian optimal reconstruction schemes in terms of mean square error for noiseless and noisy measurement JSM- 2 problems. Employing the replica method, we show that these schemes, which exploit the knowledge that the signal support is shared, can recover the signals more precisely as the number of channels increases. The main focus of our presentation is theoretical analysis; however, we will introduce applications of the signal model JSM-2 for acoustic problems. Most of our presentation is based on papers J. Stat. Mech. (2015) P05029 and J. Stat. Mech. (2016) 063304.

8:40

2aSPa3. Complex non-negative matrix factorization: Phase-aware sparse representation of audio spectrograms. Hirokazu Kameoka and Hideaki Kagami (NTT Commun. Sci. Labs., Nippon Telegraph and Telephone Corp., 3-1 Morinosato Wakamiya, Atsugi, Kanagawa 243-0198, Japan, kameoka.hirokazu@lab.ntt.co.jp)

Non-negative matrix factorization (NMF) has attracted a lot of attention after being proposed as a powerful approach for audio source separation. Factorizing the magnitude spectrogram of a mixture signal into the product of two non-negative matrices can be interpreted as approximating the observed spectra at each time frame as a linear sum of basis spectra scaled by time-varying amplitudes. One drawback of this approach is that the additivity of magnitude spectra is assumed, which holds only approximately. To address this drawback, we introduce a novel framework called the "Complex NMF," which makes it possible to realize NMF-like signal decompositions in the complex spectrogram domain. We further develop a time-domain extension of this approach, which is noteworthy in that it allows for several extensions that were not possible with the conventional NMF framework.

9:00

2aSPa4. Decomposing guided wavefields with dictionary learning. Joel B. Harley (Dept. of Elec. and Comput. Eng., Univ. of Utah, 50 S Central Campus Dr., Rm. 3104, Salt Lake Cty, UT 84112-9249, joel.harley@utah.edu) and K. Supreet Alguri (Dept. of Elec. and Comput. Eng., Univ. of Utah, Salt Lake City, UT)

Ultrasonic guided waves have become a powerful tool for the structural health monitoring of large structures due to their low attenuation and high damage sensitivity. Yet, one significant challenge of working with guided waves is their complexity. Ultrasonic guided waves demonstrate strong multi-modal and dispersive characteristics and are often used in structures with many boundaries and reflecting elements. In prior work, compressive sensing theory and sparse recovery algorithms have been used to learn the multi-modal and dispersive properties of guided waves based on relatively simple wave models. The algorithms can predict and construct wavefields based on these learned properties. Furthermore, this predictive power can be used to detect and locate damage in a structure. However, these algorithms cannot yet predict and utilize reflections as well as other complexities (such as inhomogeneities) in a structure due to the simplicity of the current model. This paper presents a dictionary learning approach to learn this model directly from simulation or experimental data. This paper focuses on studying the models (i.e., dictionaries) created by dictionary learning under various simulated conditions. We demonstrate that we can use these dictionaries to accurately predict wavefields when waves are distorted by reflections, velocity inhomogeneities, and velocity anisotropy.

9:20

2aSPa5. Dictionary learning of acoustic sound speed profiles. Michael J. Bianco and Peter Gerstoft (Marine Physical Lab., Univ. of California San Diego, Scripps Inst. of Oceanogr., 6061 La Jolla Hermosa Ave., La Jolla, CA 92037, mbianco@ucsd.edu)

Inversion for true ocean acoustic sound speed profiles (SSPs) is generally a non-linear and underdetermined problem. Traditional regularization methods model SSPs as a summation of leading-order empirical orthogonal functions (EOFs), with a minimum-energy constraint on coefficients. However, this often provides low resolution estimates of ocean SSPs. Sparse processing methods (e.g., compressive sensing) yield high resolution reconstruction of signals from few observations, provided few coefficients (of many) explain the observations. Using dictionary learning techniques, an overcomplete basis or *dictionary* of shape functions, which represent ocean SSPs using a minimum number of coefficients, is learned from a training set of SSPs. Learned dictionaries, which are not constrained to be orthogonal, optimally fit the distribution of possible ocean SSPs. Thus, each dictionary entry is informative to ocean SSP variability whereas EOFs become less informative for increasing order. It has been found that these learned dictionaries largely explain the variability of ocean SSPs using as few as one coefficient. Thus, in addition to improving the resolution of SSP estimates, learned dictionaries can improve the robustness of SSP inversion by significantly reducing the parameter search space.

9:40

2aSPa6. Acoustic single-pixel imaging using compressive sensing. Jeffrey S. Rogers (Acoust. Div., Naval Res. Lab, 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375, jeff.rogers@nrl.navy.mil), Christina J. Naify (Jet Propulsion Lab., Pasadena, CA), Charles Rhode, and Geoffrey F. Edelmann (Acoust. Div., Naval Res. Lab, Washington, DC)

The acoustic analog to the single-pixel imager found in the optical and THz imaging literature is presented. Here, the single-pixel detector is replaced with a single omnidirectional acoustic receiver and combined with a series of masks that allow for multiple orthogonal measurements. Acoustic source localization is performed based on the theory of compressive sensing which allows for the resolution of N acoustic sources using M measurements where $M << N$. A theoretical model for the acoustic single pixel imager is presented and compared to finite element models (FEM) simulated in COMSOL and experimental data taken in an air acoustic 2D waveguide. It is shown that the acoustic single pixel imager is capable of localizing multiple targets using only a single omnidirectional acoustic receiver. [Work supported by the Office of Naval Research.]

10:00–10:20 Break

10:20

2aSPa7. The utility of binary-sequence-based signals. James S. Martin (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Love Bldg., Rm. 115, Atlanta, GA 30332-0405, james.martin@me.gatech.edu)

It is frequently implied that there are advantages to the use of signals based on binary sequences for the characterization of linear time invariant (LTI) systems. However, this appears to neglect the role of the interpolation (or reconstruction) filter which is necessary in order to translate discrete sequences into analog signals and to discount the utility of analog signals that are generated by other means. Whereas a discrete sequence may offer desirable or optimal characteristics in comparison with other discrete sequences, these may easily be destroyed by the imposition of the interpolation filter. Similarly, even when these characteristics are substantially preserved in the interrogation signal, they appear consistently to be either mediocre or sub-optimal in comparison to properties of signals that are generated without consideration of binary sequences (e.g., chirps or multi-sine signals). The relative superiority of any signal with respect to the alternatives for interrogation of an LTI system depends on the goals of the characterization and relies on the assumption that some form of nonlinearity will ultimately limit signal amplitude. There does not appear to be any set of goals and limiting nonlinearity that are uniquely suited to binary-sequence-based signals.

10:40

2aSPa8. Compressive beamforming on a cylindrical array. Charles F. Gaumond (Appl. Res. in Acoust. LLC (ARiA), Washington, DC), Jonathan Botts (Appl. Res. in Acoust. LLC (ARiA), Culpeper, VA), and Jason E. Summers (Appl. Res. in Acoust. LLC (ARiA), 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

Previously, we presented subarray compressive beamforming for Doppler sensitive midfrequency active sonar on a cylindrical array, focusing on practical application for mitigating Doppler spreading caused by sidelobes and the width of the main lobe [J. Acoust. Soc. Am. **139**, 2050 (2016)]. Here, we describe ongoing work focusing on theoretical challenges of adapting compressive beamforming to a cylindrical array. Unlike linear arrays, for example, the optimal sampling for the sound field incident on a cylinder array is not evident. Formal design guidelines for compressive beamforming on a cylindrical array are discussed, guided by the statistical restricted isometry property (StRIP). Prior work has demonstrated subarray compressive beamforming by combining sparse estimates from arbitrarily chosen subarrays. Here, a formal approach is outlined in which StRIP is used to specify minimal subarrays that can individually reconstruct a sound field of a given sparsity, using a new strategy for combining individual estimates. Finally, differences between the fundamental assumptions and resulting limitations of compressive, conventional, and adaptive beamforming techniques are discussed for the cylindrical array geometry and continuous-wave (CW) waveforms of interest, as demonstrated with high-fidelity simulated data. [Work supported by a NAVSEA Phase I SBIR award.]

11:00

2aSPa9. Sparse tomographic acoustic imaging for environmental noise-source mapping. Cagdas Tuna, Douglas L. Jones, Shengkui Zhao, and Thi Ngoc Tho Nguyen (Adv. Digital Sci. Ctr., Illinois at Singapore, 1 Fusionopolis Way #08-10 Connexis North Tower, Singapore 138632, Singapore, cagdas.t@adsc.com.sg)

Environmental noise has become a major problem in large cities, increasing health risks for the urban population. Current methods are overly expensive for city-wide noise-monitoring as they generally require dense deployment of fixed microphones for noise-level measurements. We present here alternative sparse tomographic acoustic imaging techniques using arrays with relatively few number of microphones for large-region acoustic noise mapping. We first demonstrate that the locations and sound pressure levels of fixed noise sources sparsely located in a large field are recovered by collecting acoustic data at multiple locations with a portable microphone array for tomographic reconstruction. We then introduce a nonstationary tomographic imaging approach using fixed microphone arrays, which can also capture the intermittent changes in the acoustic field due to transient and/or moving noise sources. We test both the sparse static and dynamic imaging models with acoustic measurements collected with a circular microphone array mounted on an electric vehicle and two prepositioned circular arrays, respectively. The experimental results show the promise of the proposed methodology for low-cost and time-efficient large-scale noise-monitoring.

11:20

2aSPa10. Shallow water acoustic channel estimation across fluctuations in channel support. Ananya Sen Gupta (Elec. and Comput. Eng., , 4016 Seamans Ctr. for the Eng. Arts and Sci., Univ. of Iowa, Iowa City, IA 52242, ananya-sengupta@uiowa.edu)

Estimating the shallow water acoustic channel across rapidly varying multipath fluctuations have been an open challenge for decades. Recently, sparse sensing techniques have been employed to take advantage of the heavy-tailed distribution of channel components in the Delay-Doppler domain. Despite noticeable improvement over channel estimator performance by exploiting channel sparsity compared to traditional least-squared based techniques, a key challenge remains. Most sparse recovery techniques are designed to estimate high-amplitude components across static or slowly varying support. Depending on wind and sea conditions, the underlying distribution of the channel components may change unpredictably, leading to fluctuations in the channel support that are difficult to model or track efficiently with high precision. Furthermore, not all channel components are high-energy and therefore, may be unintentionally suppressed by a sparse optimization technique. We will present the current state-of-the-art on applying sparse sensing, particularly mixed norm techniques, to shallow water channel tracking. In particular, we will present extensions to prior work on channel estimation that exploit a non-convex metric to converge faster to the same solution offered by the traditional (and convex) Lasso metric. We will present how non-convex optimization can navigate varying channel sparsity and present results based on experimental field data.

11:40

2aSPa11. Grid-free channel estimation of acoustic measurement data from SAVEX15. Youngmin Choo, Jungyong Park, Yongsung Park, Minseuk Park (Res. Inst. of Marine Systems Eng., Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul, Seoul 151 - 744, South Korea, sks655@snu.ac.kr), and Woojae Seong (Naval Architecture and Ocean Eng. and Res. Inst. of Marine Systems Eng., Seoul National Univ., Seoul, South Korea)

Acoustic channel impulse responses (CIRs) consisting of multiple arrivals are extracted from the measured data during SAVEX15 experiment (conducted in the northern East China Sea during May 2015) using compressive sensing, providing high-resolution results due to sparsity promoting objective function. Performance of grid-based compressive channel estimation degrades when basis mismatch occurs because delays of the arrivals do not correspond to delays on the grid. To overcome this, grid-free compressive channel estimation is derived from the formulation of grid-free compressive beamforming [J. Acoust. Soc. Am. **137**, 1923-1935 (2015)]. The grid-free compressive channel estimation is applied to signals from SAVEX15; the signals were transmitted from a towed source and received at two separate vertical line arrays (16 elements each); they are used to measure the signals traveling through waveguides. Acoustic structures are observed clearly due to high-resolution CIRs along depth, which are varied by the moving source and internal wave activities.

Session 2aSPb

Signal Processing in Acoustics: General Topics in Signal Processing II (Poster Session)

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

Brian E. Anderson, Cochair
N145 Esc, Brigham Young Univ., MS D446, Provo, UT 84602

All posters will be on display from 8:30 a.m. to 11:30 a.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 8:30 a.m. to 10:00 a.m. and authors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m.

Contributed Papers

2aSPb1. Audio spotlight system based on formation of multiple focused sound sources via loudspeakers. Shuji Akamatsu and Yosuke Tatekura (Shizuoka Univ., 3-5-1 Johoku, Hamamatsu, Shizuoka 432-8561, Japan, s-akama@spalab.eng.shizuoka.ac.jp)

We propose a new audio spotlight system which uses focused sound sources arranged in an arc shape, and we evaluate its suppression performance and frequency responses. Audio spotlight systems use ultrasonic waves to produce sharp directional sounds without any external disturbances. However, such systems contain two problems: (1) low efficiency of the electroacoustic conversion and (2) distortion of the reproduced sound. Therefore, we attempted to form an audio spotlight system by using focused sound sources based on wave field synthesis. A single focused sound source is designed to harmonize the phase of the driving signal at an arbitrary position. Thus, the driving signals that correspond to the number of the optional focused sources can be superimposed and reproduced from a loudspeaker array in order to form multiple focused sound sources. By arranging the focused sources in arc shape, we produced the audio spotlight. A numerical simulation revealed that the sound volume difference between the center of an arc and at 1 m away from the center was approximately 10 dB. Moreover, because a flat frequency response was maintained at the center of the arc, the proposal is expected to be able to control optional sounds with minimal distortion.

2aSPb2. Study on the prevention of Sinkhole in cities through sound analysis. Ik-Soo Ann and Myung-Jin Bae (TeleCommun. & Information, soolsil Univ., 369 sangdo-ro, Dongjak-gu, Seoul 156-743, South Korea, aisbestman@naver.com)

Sinkhole is a symptom where the surface area collapses due to lack of supporting power and this is caused by erosion of rocks under the ground or washing away of soil. It is very difficult to say that this sinkhole happening in the downtown is natural but rather resulted from development of underground water, water leakage from the old water pipes buried, and subway construction. As this symptom cannot be predicted visually by looking at the surface it happens unexpectedly and thus has a high possibility of causing a lot of damage. However, active research on developing prevention system by using a special laser video or sound has been carried out. In regards to this, this study suggests methods of preventing downtown sinkhole by analyzing sound. The special feature that classifies sound made from tapping the normal surface and surface of the potential sinkhole is the resonance. If we can have a focused control over these surfaces with high potential of sinkhole we will be able to prevent any accidents taking from this.

2aSPb3. Surf infrasound recorded with smartphones during the 2016 Eddie Aikau. Karina A. Asmar (HIGP, Univ. of Hawaii, Manoa, Honolulu, HI 96822, kasmar@hawaii.edu), Milton Garces (HIGP, Univ. of Hawaii, Manoa, Kona, HI), Julie Schnurr (HIGP, Univ. of Hawaii, Manoa, Honolulu, HI), and Anthony Christe (ICS, Univ. of Hawaii, Manoa, Honolulu, HI)

Standardization of signal characteristics is a recurring problem in infrasound detection. We are developing a suite of scalable multiresolution algorithms for infrasound data processing that may include unevenly sampled smartphone barometer data. The algorithms include fundamental methods for Lomb-Scargle periodograms and coherent energy estimators. The implementation of these algorithms will facilitate reproducible and transportable characterization of diverse signatures captured by traditional and ubiquitous global infrasound networks. We discuss the application of these algorithms to surf infrasound signals recorded with smartphones during the 25 February 2016 Eddie Aikau Surf Contest held at Waimea Bay, Oahu.

2aSPb4. Directivity control for regular polyhedron loudspeaker array based on weighted least-squares method using adjusted weight in spherical harmonics domain. Makoto Daikohara and Yoichi Haneda (Dept. of informatics, The Univ. of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan, d1530034@edu.cc.uec.ac.jp)

We propose an adjusted weight least-squares method for directivity control of spherical loudspeaker arrays in the spherical harmonics domain. In previous work, we demonstrated that the sidelobes of the directivity pattern can be suppressed using the weighted least-squares method in the frequency domain. However, the gains of the calculated filter coefficients were very large at low frequencies. Consequently, to overcome this problem, we designed filter coefficients in the spherical harmonic domain. First, we set an ideal directivity pattern using a window function. Next, we obtain the filter based on the weighted least-squares method in the spherical harmonic domain with an initial weighting factor. Then, we update the weighting factor only on the side opposite that being faced based on the difference in sound pressure level between the ideal and the simulated radiation patterns. We then repeat this process until the difference level is satisfied by the condition of the update. The results of comparison of our proposed method with our previous method and the conventional least-squares method indicate that although the beam width in the look direction is wider than our previous method, our proposed method effectively prevents increasing filter coefficients by limiting order at low frequencies.

2aSPb5. A basic study of acoustic communications between an autonomous surface vehicle and an autonomous underwater vehicle. Mitsuyasu Deguchi, Yukihiro Kida, Koji Meguro, Yoshitaka Watanabe, Takuya Shimura, Hiroshi Ochi (Underwater Acoust. Technol. Group, Japan Agency for Marine-Earth Sci. and Technol., 2-15 Natsushima-cho, Yokosuka, Kanagawa 237-0061, Japan, deguchim@jamstec.go.jp), and Yukinori Akamine (Information Electronics Res. Dept., Res. & Development Group, Hitachi Ltd., Kokubunji-shi, Japan)

Recently, there has been an increasing interest in submarine resources and demand to survey them. For that purpose, Japan Agency for Marine-Earth Science and Technology (JAMSTEC) has developed autonomous underwater vehicles (AUVs). Currently, only one AUV is operated by a research vessel because a mother ship must always follow an AUV. However, to make survey more efficient, the project of multiple AUVs operation, paired with autonomous surface vehicles (ASVs), is being progressed. In this plan, one ASV always monitor status and position of its paired AUV with acoustic communication and localization, and follow it autonomously. The size of an ASV is desired to be as small as possible in terms of operation. Thus, it causes two difficulties. First, such a small ASV rolls and pitches up to a large angle easily. Second, receivers are close to the noise sources, such as thrusters or sea surface. As a basic study, at-sea experiments were demonstrated with various types of modulation using a prototype of an ASV. The results show that OFDM signals are more sensitive to rolling and pitching movement than single carrier modulation (SCM) and that it is possible to achieve demodulation with SCM in spite of noise as mentioned above.

2aSPb6. Separation of individual instrumental sounds in monaural music signals applying a modified Wiener filter on the time-frequency plane. Yukio Fukayama and Masafumi Watanabe (Elec. Systems, Hiroshima Inst. of Technol., 2-1-1 Miyake Saeki-ku, Hiroshima Inst. of Technol., Hiroshima 731-5193, Japan, fukayama@ieee.org)

This research seeks a signal processor for random processes which is suitable for distilling melodies played by a particular instrument from music ensemble. The proposed processor sequentially evaluates statistics of the mixture assuming a model of music sound and employs a modified Wiener filter so as to separate the mixture of time-variant and correlated random processes such as amplitude changing tones and same pitch tones by different instruments. The model of music sounds is assumed to be a weighted mixture of each standard tone, which is a typical tone of every pitch played by a particular instrument with invariant power, where the weight is called the amplitude factor corresponding to the standard tone. Thus, the proposed processor, at first, evaluates the current amplitude factors which are the the coordinates of the mixture where the standard tones form oblique basis. Then, according to evaluated coordinates of the mixture, its current autocorrelation and cross-correlation to each standard tone are calculated for obtaining the modified Wiener filter. Finally, the filter separates the mixture into individual tone, which is a component of a particular pitch played by an instrument with variant power, and combination of which provides a melody.

2aSPb7. Shouted speech detection using hidden markov model with rahmonic and mel-frequency cepstrum coefficients. Takahiro Fukumori, Masato Nakayama, Takanobu Nishiura (Ritsumeikan Univ., 1-1-1, Nojihigashi, Kusatsu 525-8577, Japan, fukumori@fc.ritsumei.ac.jp), and Hiroaki Nanjo (Kyoto Univ., Kyoto, Japan)

In recent years, crime prevention systems have been developed to detect various hazardous situations. In general, the systems utilize the image information recorded by a camera to monitor the situations. It is however difficult to detect them in the blind area. To address the problem, it is required to utilize not only image information but also acoustic information occurred in such situations. Our previous study showed that two acoustic features including rahmonic and mel-frequency cepstrum coefficients (MFCCs) are effective for detecting the shouted speech. Rahmonic shows a subharmonic of fundamental frequency in the cepstrum domain, and MFCCs represent coefficients that collectively make up mel-frequency cepstrum. In this method, a shouted speech model is constructed from these features by using a gaussian mixture model (GMM). However, the previous method with GMM has difficulty in representing temporal changes of the speech features. In this study, we further expand the previous method using hidden Markov model (HMM) which has state transition to represent the temporal changes. Through objective experiments, the proposed method using HMM could achieve higher detection performance of the shouted speech than the conventional method using GMM.

2aSPb8. Vandermonde transform for complex residual based on complex speech analysis. Taishi Shimabukuro (IIJ Global, Nishihara, Japan) and Keiichi Funaki (C&N Ctr., Univ. of the Ryukyus, Senbaru 1, Nishihara, Okinawa 903-0213, Japan, funaki@cc.u-ryukyu.ac.jp)

Vandermonde transform using the orthogonal basis is a time-frequency transform method that can describe a speech and audio signal to be uncorrelated and sparse, and can be considered to be also effective in speech processing. In particular, when applied to the residual considered to be effective in a pitch estimation of speech and speaker recognition. On the other hand, we have already proposed Time-Varying Complex AR (TV-CAR) speech analysis for an analytic signal. The TV-CAR analysis can improve spectral estimation accuracy in the low frequencies and it can estimate the complex AR coefficients for each sample. As a result, in the estimated complex residual signal, formant components are more subtracted from speech signal. It can perform better on pitch estimation, or so on. In this paper, we evaluated the time-frequency analysis using the Vandermonde transform based on an Orthogonal Matching Pursuit (OMP) method of the literature using the complex AR residual. The performance is evaluated for several orthogonal basis such as Fourier basis, Cos basis, K-L transform, and Gabor basis. Furthermore, the Ramanujan-sum is also introduced as the basis function to realize time-frequency transform.

2aSPb9. Individual sound image generation by null-space based sound field control under few loudspeakers in real environment. Shinpei Horii and Yosuke Tatekura (Shizuoka Univ., 3-5-1 Johoku, Hamamatsu, Shizuoka 432-8561, Japan, s-horii@spalab.eng.shizuoka.ac.jp)

To achieve an individual sound image generation with fewer loudspeakers based on NBSFC (Null-space Based Sound Field Control) filtering, we attempt to expand NBSFC under underdetermined condition, which means the number of loudspeakers is smaller than the number of control points. NBSFC filtering can suppress the sound pressure only on optional control points. In the conventional NBSFC design, the number of loudspeakers is required to be greater than that of control points. On the other hand, many control points are indispensable to achieve fine individual sound image generation. Thus, NBSFC filters are designed with the maximum control points limited by the conventional method from the transfer functions between the control points and the loudspeakers in various combinations. Also, the expanded NBSFC filters can be designed corresponding to the underdetermined condition by superimposing all conventional NBSFC filters. As the result of numerical simulation with a real environmental data, the sound pressure difference between inside and outside the control area was averaged by 7 dB. This reveals that the control area where sound pressure is suppressed can be realized by locating many control points closely.

2aSPb10. Personal audio reproduction using two wearable end-fire loudspeaker arrays. Kenta Imaizumi and Yoichi Haneda (Dept. of Informatics, The Univ. of Electro-Communications, 1-5-1, Chofugaoka, Chofu-shi, Tokyo 182-8585, Japan, i1630013@edu.cc.uec.ac.jp)

We propose a wearable personal audio reproduction system that uses two end-fire loudspeaker arrays. The two loudspeaker arrays are attached in front of the listener's breast. Consequently, the system is not close to the ears. Further, because the main beams of the two arrays are directed at each ear, crosstalk between the left and right signals is low. In addition, sound leakage is also low as a result of directivity control. We constructed a prototype of the system using four loudspeaker elements for each array, with intervals of 4 cm and array length 12 cm. The attenuation level of the area at approximately 50 cm away from the end-fire arrays was found to be -20 dB sound pressure level (SPL) relative to the main beam pointed at the ear. In addition, the crosstalk SPL difference was approximately 12 dB. We can control the sound image position by convolving the head-related transfer function with the sound source. Further, no crosstalk canceller is needed because the proposed method secures the interaural intensity difference. Improvement or application of the proposed system can facilitate realization of a personal sound field.

2aSPb11. Sound source separation using image signal processing based on sparsity of sound field. Masaaki Ito, Kenji Ozawa, Masanori Morise, Genya Shimizu (Graduate School of Medicine and Eng. Sci. Dept. of Education, Univ. of Yamanashi, 4-3-11 Takeda, Kofu, Yamanashi 400-8511, Japan, g15mk002@yamanashi.ac.jp), and Shuichi Sakamoto (Tohoku Univ., Sendai, Japan)

Microphone arrays have been used to separate sound sources to improve speech recognition in a noisy environment. We propose a method using image signal processing to achieve highly accurate sound source separation. The microphone array is 1-m long and consists of eight microphones. Temporal sequences of the sound pressures obtained from the eight microphones are converted into sequences of luminance. These are arranged in parallel in an image, which is referred to as *aspatio-temporal sound pressure distribution image*. Sparse modeling using L1 regularization (Lasso) is applied to the image for restoring a high-resolution image. The spatial spectrum of the restored image is obtained using a two-dimensional fast Fourier transform (FFT) algorithm. In this spectrum, the angle of a line through the center denotes the arrival direction of sound, and the distance from the center indicates its frequency. By extracting a line from the spectrum, the sound source can be separated. A computational experiment revealed that high-resolution sound field obtained by a 512-microphone array could be restored using the proposed method. Moreover, SNR was improved by 32.1 dB when two sounds arrived 45° apart, indicating sufficient performance to extract a desired sound. [Work supported by JSPS KAKENHI Grant (16K06384).]

2aSPb12. Modified 2nd-order nonlinear infinite impulse response (IIR) filter for compensating sharpness and nonlinear distortions of electrodynamic loudspeakers. Kenta Iwai (Organization for Res. and Development of Innovative Sci. and Technol., Kansai Univ., 3-3-35, Yamate-cho, Suita 564-8680, Japan, kenta1986@gmail.com) and Yoshinobu Kajikawa (Faculty of Eng. Sci., Kansai Univ., Osaka, Japan)

Electrodynamic loudspeakers generate the nonlinear distortions and degrade the sound quality. This is because the nonlinearity of electrodynamic loudspeaker is caused by the large displacement of the diaphragm, in particular, around the lowest resonance frequency. One of the compensation methods for the nonlinear distortions is using mirror filter. This filter is derived from the linear and nonlinear differential equations and designed by second-order nonlinear IIR filter or third-order nonlinear IIR filter. The coefficients of these filters are determined by the physical parameters of the loudspeaker. However, it is difficult to compensate the nonlinear distortions when the sharpness of the loudspeaker is high at the lowest resonance frequency because of large displacement of the diaphragm. Although it is required to compensate the sharpness of the loudspeaker, second-order nonlinear IIR filter cannot compensate the sharpness because of the compensation principle, i.e., this filter adds the nonlinear signal into the input signal and does not include the linear filtering. This paper proposes a modified second-order nonlinear IIR filter that can compensate not only the sharpness but also the lowest resonance frequency. The difference between the conventional and proposed filter structures is only coefficients. Experimental results show the effectiveness of the proposed filter.

2aSPb13. An interleave division multiple access receiver for multi-user underwater acoustic communication. Ning Jia, JianChun Huang, and Dong Xiao (Key Lab. of Underwater Acoust. Environment, Inst. of Acoust. Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, 100190 Beijing, China, jianing@mail.ioa.ac.cn)

A downlink interleave division multiple access (IDMA) receiver, employing sparse channel estimation and interference canceller (IC), is proposed for multi-user underwater communications. To eliminate inter-symbol interference (ISI) and multiple access interference (MAI), an equalizer, a decoder, and an interference canceller are implemented in the receiver. The equalizer is used to eliminate ISI. The channel parameters used in the equalizer are estimated by a modified OMP sparse channel estimation algorithm. The performance of channel estimation is further improve ed using the decoder soft output information during the receiver iteration. MAI, reconstructed by the decoder output and channel parameters, can be eliminated in IC section of other users. The receiver's performance is verified in shallow water acoustic channels during a sea-trail in the South China Sea. Three different ranges between the transmitter and the receiver, i.e., 4.5 km,6 km, and 7 km, were investigated with a maximum of four users at the data rate of 500 bps per user. The results demonstrated that the proposed IDMA receiver provides good performance with low bit error rate (BER). Furthermore, the receiver has been applied in a real-time underwater communication network.

2aSPb14. Relationship between speech recognition performance and related spoken document retrieval performance. Noboru Kanedera (Dept. of Electronics and Information Eng., National Inst. of Technol., Ishikawa College, Kitachujou, Tsubata-Machi, Ishikawa 929-0392, Japan, kane@ishikawa-nct.ac.jp)

Along with multimedia information including the speech information having increased, spoken document retrieval has attracted attention. Keywords or query sentences are used for spoken document retrieval. The keywords or query sentences cannot be used when we want to watch the video scenes which are related to a watching video scene. In this report, we propose a method of evaluating spoken document retrieval between video scenes, and report the results of the investigation about the relationship between the speech recognition performance and spoken document retrieval performance by using watching video scene. As a result of simulation experiment, it was confirmed to affect the search performance in order of substituted error, deletion error, insertion error, in the search both by question sentence and by the scene. The search by the scene is easy to be affected by the speech recognition error than the search by the question sentence. It was found that the influence of the insertion error in the search by the scene was bigger than the search by the question sentence.

2aSPb15. Automatic selection of the number of poles for different gender and age groups in steady-state isolated vowels. Thayabaran Kathiresan (Phonet. Lab., Univ. of Zurich, Plattenstrasse 54, Zurich, Zurich 8032, Switzerland, thayabaran.kathiresan@uzh.ch), Dieter Maurer (Inst. for the Performing Arts and Film, Zurich Univ. of the Art, Zurich, Zurich, Switzerland), and Volker Dellwo (Phonet. Lab., Univ. of Zurich, Zurich, Zurich, Switzerland)

Formant frequency estimation using a linear prediction (LPC) algorithm is based on the assumption of age- and gender-specific number of poles. However, when visually crosschecking the calculated formant frequencies along with a spectrogram, investigators often change the parameter because of a lack of correspondence. The misprediction is mainly due to a high variation within the calculated formant tracks, or tracks not matching with spectral peaks, possibly combined with an unexpected low or high number of occurring formants (e.g., formant merging, spurious formants). To solve the problem of changing the number of filter poles, we propose a new method which addresses the first aspect of the constancy of formant tracks. For a given vowel sound, the first three formant frequencies for three different settings (number of poles = 10, 12, and 14 for a frequency range 0-5.5 kHz) are calculated. The standard deviation of the formant tracks is used to find a Euclidean distance for three settings separately. The algorithm chooses the setting that produces least variability (minimum Euclidean distance) in steady-state vowel nuclei. We tested the method on vowel sounds of standard German /i, y, e, ø, ε, a, o, u/ produced by 14 men, 14 women, and 8 children.

2aSPb16. Performance analysis of passive time reversal communication technique for multi-path interference with synthetic data. Yukihiro Kida, Mitsuyasu Deguchi, Yoshitaka Watanabe, Takuya Shimura, Hiroshi Ochi, and Koji Meguro (Underwater Acoust. Technol. Group, JAMSTEC, 2-15 Natsushima-cho, Yokosuka 2370061, Japan, kiday@jamstec.go.jp)

The interference by time-varying and intense underwater acoustic multipath is one of the most obstructive phenomena in underwater acoustic communication. Time-Reversal (TR) communication techniques utilize the reciprocity of acoustic channels to converge the spread signals to the original one, and are considered one of the most effective methods to overcome multi-path interference as shown in many past studies. It is recognized empirically that the large number of multi-path arrival could generally raise the demodulation result of TR. However, its relationship is rarely evaluated quantitatively. In this paper, we investigate the efficiency of the TR acoustic communication techniques for the multi-path interference cancelation by applying our passive TR method to the synthetic datasets of horizontal acoustic communication in shallow water environment. Mainly, we focused on the relation between the Signal to Interference Noise Ratio (SINR) and the output SNR of demodulation result. The result shows: 1) SINR and output SNR has a nearly linear relation up to a limitation, and 2) above the limitation, output SNR reaches its ceiling or slightly decreases.

2aSPb17. Development of an underwater sound monitoring system using two hydrophones for a remotely operated underwater vehicle. Sea-Moon Kim, Sung-Hoon Byun, Hyun-Taek Choi, and Chong-Moo Lee (Korea Res. Inst. of Ships and Ocean Eng., 32 Yuseong-daero 1312beon-gil, Yuseong-gu, Daejeon 34103, South Korea, smkim@kriso.re.kr)

Recently, a sound monitoring system for hearing fishery sound using three hydrophones installed on an ROV (Rountree and Juanes, 2010) has been developed. However, the operation of the system was not successful because of the high level of the self-noise generated by the ROV. Its thrusters had to be turned off to reduce the self-noise and listen to fishery sound clearly. Currently, Korea Research Institute of Ships and Ocean Engineering (KRISO) is developing a light-work class ROV and a similar sound monitoring system was developed for the ROV. The design of the system was focused on the following two strategies: (1) minimization of self-noise effect; (2) reproducing sound for operators to identify source direction. To minimize the self-noise effect, a digital filter with seven filter banks with octave bandwidth was implemented. A time-domain beamforming technique has been applied to analyze the source location. Reproducing sound signals for the two speakers in the control room were generated according to the estimated source location. The performance of the algorithm was verified by using a receiver array signal simulator. Its detail characteristics will be presented in the talk. [This work was financially supported by the research project # 20130197 funded by KIMST.]

2aSPb18. Study on intelligibility improvement method based on subband waveform processing focusing on dynamic feature of speech. Hiroki Kohara, Hideki Banno, and Kensaku Asahi (Graduate School of Sci. and Technol., Meijo Univ., 1-501 Shiogamaguchi, Tempaku-ku, Nagoya-shi, Aichi-ken 468-8502, Japan, 163430008@ccalumni.meijo-u.ac.jp)

This paper describes intelligibility improvement method for speech signal based on subband waveform processing. Our approach is based on the observation that clear speech has higher delta-cepstrum value in transient parts between phonemes, and emphasizes delta-cepstrum of input speech by a filter in the cepstral domain which amplifies a particular modulation frequency. However, since this approach generates synthetic sound by using an analysis/synthesis system, quality of the generated sound is sometimes degraded. To prevent this degradation, a subband waveform-based method is introduced. This method divides an input signal into several subband signals by a quadrature mirror filter (QMF) which approximately enables perfect reconstruction of input signal from the subband signals, converts an amplification gain sequence in the cepstral domain into that in the subband-waveform domain, and then multiplies the converted amplification gain sequence to the subband signal on a sample-by-sample basis. Synthetic sounds were generated by the method in the cases where the number of subbands is set to two, four, and eight. We found that the sound that the number of subbands is two includes artificial power fluctuation, and increasing the number of subbands decreases the artificial power fluctuation and makes quality of the generated sound better.

2aSPb19. Stereophonic reproduction based on multi-beam-forming with curved-type parametric array loudspeaker. Shinya Komori, Nana Hasegawa, Takahiro Fukumori, Masato Nakayama, and Takanobu Nishiura (Ritsumeikan Univ., 1-1-1, Noji-higashi, Kusatsu, Shiga 525-8577, Japan, is0116fv@ed.ritsumei.ac.jp)

A parametric array loudspeaker (PAL) has a flat surface arrangement of ultrasonic transducers and achieves a sharper directivity by utilizing an ultrasound wave. It can form a narrow audible space called an "audio spot" that can provide a target sound to a particular listener. In addition, the PAL can achieve a stereophonic reproduction by designing the audio spots for the positions of ears on the basis of delay-and-sum beamforming. However, designing wider audio spots for several listeners is difficult for the conventional PAL because it has overly sharp directivity. To solve this problem, we propose a new stereophonic reproduction with a curved-type PAL that can expand the audio spots. The curved-type PAL consists of ultrasonic transducers arranged on a concave surface. It can form a wider directivity compared with the conventional flat-type PAL. The proposed method steers the acoustic beam by designing the focal point to the positions of ears, and the radius of the curved-type PAL varies the width of the acoustic-beams. In this study, we carried out an objective evaluation experiment to confirm the effectiveness of the proposed method. As a result, we confirmed that the proposed method can design two wider acoustic beams for the stereophonic reproduction.

2aSPb20. High accuracy resampling for acoustic signals using hermite interpolation with pseudo spectral method. Hotaka Maruyama, Kan Okubo, and Norio Tagawa (System Design, Tokyo Metropolitan Univ., 6-6 Asahigaoka, Hino, Tokyo 191-0065, Japan, maruyama-hotaka@ed.tmu.ac.jp)

Interpolation is an essential technique to obtain continuous data from discrete data within a known range of coordinates in many research areas. Especially, resampling acoustic signal is an important application. It is necessary to obtain the value between digitally recorded data using a signal interpolation technology. It is a fundamental method of digital processing operations. In recent years, a 192 kHz high-resolution system is often employed, so that the accurate interpolation method is required to obtain the high resolution signals from the signals measured by the conventional lower sampling rate. In signal resampling using the Hermite interpolation, an interpolated value can be expressed as the convolution form by signal values and their derivatives. By using highly accurate derivatives, accuracy of resampling can be also improved. In this study, we propose a new highly accurate acoustic signal interpolation method combining the Hermite interpolation with the pseudo-spectral method. Moreover, we also propose their efficient computational algorithm employing separation technique. As a result, our evaluation clarified that the proposed method achieved more accurate signal resampling compared to the conventional methods.

2aSPb21. Visualization of one source in a two sources sound field by estimating source positions. Maureen and Yoichi Haneda (Dept. of Informatics, The Univ. of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan, maureen@uec.ac.jp)

Sound field visualization is important in many applications. Acoustic holography (AH) uses a linear microphone array to visualize sound fields. However, in general, all existing sound sources will appear when AH is used. In this paper, we propose a method that visualizes a single source from two sources in a sound field by estimating source positions and separating them. In the proposed method, sound source positions are estimated based on the microphone input signals' phase components for each frequency. The source positions of all frequencies are placed in a histogram that expresses each position's quantity, which facilitates acquisition of accurate source positions. We determine one source and select frequency bins that indicate the obtained source position. We can also virtually extrapolate the microphone array length by solving the wave propagation model using the least-squares method. The results of sound field visualization experiments conducted considering the depth direction and using the one-dimensional near-field acoustic holography inverse propagation method show that the proposed method can separate a single source by selecting the histogram peak even when the wavefronts of two sources appear simultaneously. Future work will attempt to restore only one source or both two sources to time signal.

2aSPb22. Utilization of optional instrument sounds for simultaneous measurement of room impulse responses by ensemble music. Yuki Miyauchi and Yosuke Tatekura (Shizuoka Univ., 3-5-1 Johoku, Hamamatsu, Shizuoka 432-8002, Japan, y-miya@spalab.eng.shizuoka.ac.jp)

We describe the influence of the separation accuracy via semi-blind source separation (semi-BSS) upon the measurement accuracy of the room impulse responses. Tatekura *et al*. had proposed a simultaneous measurement method of the room impulse responses with ensemble music. In this method, the room impulse responses were measured by separating ensemble music into each instrument sound reproduced from each loudspeaker. However, the influence of the separation accuracy via semi-BSS upon the measurement accuracy of the room impulse responses has not been clarified. Therefore, we evaluated the measurement accuracy of the room impulse responses and the separation accuracy by using the different instrument sounds. The room impulse responses were measured between two loudspeakers and a control point with 11 kinds of musical instrument sounds. From the result of the measurement accuracy, the difference between the maximum value and the minimum value was 10 dB. However, from the result of the separation accuracy, most of value held equivalent measurement accuracy. Therefore, under these measurement conditions, this method is expected to be able to measure the room impulse responses by optional instrument sounds.

2aSPb23. An evaluation of speech reconstruction for degraded speech of optical microphone with deep neural network. Tomoyuki Mizuno, Takahiro Fukumori, Masato Nakayama, and Takanobu Nishiura (Ritsumeikan Univ., 1-1-1, Noji-higashi, Kusatsu 525-8577, Japan, is0140kk@ed.ritsumei.ac.jp)

Measuring distant-talking speech with high accuracy is important in detecting criminal activity. Various microphones such as the parabolic and shotgun microphones have been developed for measuring it. However, most of them have difficulty in extracting distant-talking speech at a target position if they are surrounded by noisy sound sources. Therefore, this study focuses on a microphone system to extract the distant-talking speech by vibrating a papery object using laser light. This system is referred to as an optical microphone in this study. The sound quality of the optical microphone is especially degraded at higher frequencies because it utilizes an external diaphragm consisting of various materials as the vibrating papery object. In this study, we therefore propose a reconstruction method for degraded distant-talking speech observed with the optical microphone. The method is realized by using a deep neural network (DNN) that is trained as the system between clean and observed speech signals. In the proposed method, the log-power spectra of 11 frames are used for the input of the DNN. Finally, we confirmed the effectiveness of the proposed system through an evaluation experiment.

2aSPb24. Voice livness detection based on pop-noise detector with phoneme information for speaker verification. Shihono Mochizuki, Sayaka Shiota, and Hitoshi Kiya (Information and Commun. Systems, Tokyo Metropolitan Univ., 6-6, Asahigaoka, Hino-shi, Tokyo 191-0065, Japan, mochizuki-shihono@ed.tmu.ac.jp)

This paper proposes a pop-noise detector using phoneme information for a voice liveness detection (VLD) framework. In recent years, spoofing attacks (e.g., reply, speech synthesis, and voice conversion) have become a serious problem against speaker verification systems. Some techniques have been proposed to protect the speaker verification systems from these spoofing attacks. The VLD framework has been proposed as one of fundamental solutions. The VLD framework identifies that an input sample is uttered by an actual human or played by a loudspeaker. To realize the VLD framework, pop-noise detection methods have been proposed and these methods perform well as the VLD module. However, since pop-noise is a common distortion in speech that occurs when a speaker's breath reaches a microphone, the phenomenon of pop-noise is able to be occurred by winds or attackers arbitrary. It is one problem of the pop-noise detection methods. In order to improve the robustness of the pop-noise detection methods, this paper proposes a pop-noise detector using phoneme information as an evidence of an actual human. From the experimental results, the proposed method increases the robustness of the VLD against spoofing attacks.

2aSPb25. Simultaneous recognition of phone and speaker using three-way restricted Boltzmann machine. Toru Nakashika and Yasuhiro Minami (The Univ. of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan, nakashika@uec.ac.jp)

In this study, we attempt a simultaneous recognition method of phone and speaker using a single energy-based model, a three-way restricted Boltzmann machine (3WRBM). The proposed model is a probabilistic model that includes three variables: acoustic features, latent phonetic features, and speaker-identity features. The model is trained so that it automatically captures the intensity of relationships among the three variables. Once the training is done, we can apply the model to many speech signal processing tasks because it has an ability to separate phoneme and speaker-related information from the observed speech, and generate a speech signal from the phoneme and speaker-related information on the contrary. Simultaneous phone and speaker recognition is achieved by estimating the latent phonetic features and the speaker-identity features given the input signal. In our experiments, we discuss the effectiveness of the mode lin a speaker recognition and a speech (continuous phone) recognition tasks.

2aSPb26. Segmental conditional random fields audio-to-score alignment distinguishing percussion sounds from other instruments. Ayako Noguchi, Shinji Sako, and Tadashi Kitamura (Nagoya Inst. of Technol., Gokiso-cho, Showa-ku, Nagoya, Aichi 4668555, Japan, noguchi@mmsp.nitech.ac.jp)

Audio-to-score alignment is useful technique because it can be widely applied to many practical applications for musical performance. However, it is still open problem due to the complexity of audio signal especially in the polyphonic music. Additionally, performing in real-time is also important in practical situation. In this study, we propose a new alignment method based on segmental conditional random fields (SCRFs). The attractive feature of this method is utilizing to distinguish percussion sounds from the other instruments. In general, percussion sounds have a role in managing whole music. Moreover, performers can pick up the percussion sounds from the others by hearing whole sound thanks to their unique features of the sound. In the field of score alignment, hidden Markov models (HMMs) or CRFs was used in previous studies including our previous one. In summary, these methods were formulated as a matching problem of the state sequence of mixed notes with the audio feature sequence. In this study, we extend our previous method by combining an additional state which represents percussion sounds. Furthermore, we introduce the balancing factor to control the importance of classifying feature functions. We confirmed the effectiveness of our method by conducting experiments using RWC music database.

2aSPb27. AI framework to arrange audio objects according to listener preferences. Kento Ohtani (Nagoya Univ., Furo-cho,Chikusa-ku, Nagoya, Aichi 464-8601, Japan, ohtani.kento@g.sp.m.is.nagoya-u.ac.jp), Kenta Niwa (NTT Media Intelligence Labs., Musashino, Tokyo, Japan), and Kazuya Takeda (Nagoya Univ., Nagoya, Japan)

We propose a framework for arranging audio objects in recorded music using artificial intelligence (AI) to anticipate the preferences of individual listeners. The signals of audio objects, such as the tracks of guitars and drums in a piece of music, are re-synthesizes in order to provide the preferred spatial arrangements of each listener. Deep learning-based noise suppression ratio estimation is utilized as a technique for enhancing audio objects from mixed signals. Neural networks are tuned for each audio object in advance, and noise suppression ratios are estimated for each frequency band and time frame. After enhancing each audio object, the objects are re-synthesized as stereo sound using the positions of the audio objects and the listener as synthesizing parameters. Each listener supplies simple feedback regarding his/her preferred audio object arrangement using graphical user interface (GUI). Using this listener feedback, the synthesizing parameters are then stochastically optimized in accordance with each listener's preferences. The system recommends dozens of customized synthesizing parameters, and these parameters can then be adjusted by the listener using a single-dimensional slider bar. Several tests were conducted and the proposed framework scored high marks in subjective evaluations as to whether or not the recommended arrangements were indeed preferable.

2aSPb28. An improved filter employing measured characteristics of diaphragm displacement for compensating loudspeaker nonlinearity. Manabu Omura (Faculty of Eng. Sci., Kansai Univ., 3-3-35, Yamate-cho, Suita, Osaka 564-8680, Japan, k675924@kansai-u.ac.jp), Kenta Iwai (Organization for Res. and Development of Innovative Sci. and Technol., Kansai Univ., Suita, Osaka, Japan), and Yoshinobu Kajikawa (Faculty of Eng. Sci., Kansai Univ., Suita, Osaka, Japan)

In this paper, we propose an improved filter employing measured characteristics of diaphragm displacement for compensating nonlinear distortions in loudspeaker systems. Loudspeaker systems generate nonlinear distortions that degrade the sound quality because of complicated structure of loudspeaker systems. Therefore, we set out to improve the sound quality by employing Mirror filter which compensates nonlinear distortions in electro-dynamic transducer. Mirror filter calculates linear instantaneous displacement of diaphragm from the linear differential equation of the electro-dynamic transducer and generates compensation signals based on the nonlinear differential equation of the electro-dynamic transducer and the calculated displacement of diaphragm. Hence, the compensation performance with Mirror filter is determined by the accuracy of the calculated linear and nonlinear motion based on the differential equations. However the calculated motion of the diaphragm does not agree with the actual motion perfectly because of the errors which come from parameters in the differential equations and modeled error. Therefore, an improved Mirror filter is proposed to exclude these errors. In detail, the improved Mirror filter employs the impulse response, which represents measured characteristics of diaphragm displacement, to calculate instantaneous displacement of diaphragm. In this paper, we demonstrate the validity of the proposed filter through experimental results.

2aSPb29. Amplitude limiters based on phase optimization. Akira Kakitani, Daisuke Saito (The Univ. of Tokyo, Hongo 7-3-1, Bunkyo-ku, Tokyo, Tokyo 1130033, Japan, dsk_saito@gavo.t.u-tokyo.ac.jp), Yasuhiro Kosugi (none, Tokyo, Japan), and Nobuaki Minematsu (The Univ. of Tokyo, Tokyo, Japan)

In order to reduce the peak value of source waveforms without quality degradation, a novel method is proposed. In this method, by exploiting the characteristics of human auditory system, which is often said to be insensitive to phase changes, peak value reduction is made possible. The proposed method is an improved version of our previous study, which manipulated the waveforms mainly based on phase control. In the conventional algorithm, however, amplitude characteristics have to be altered to some degree. The proposed method attempts to minimize the change of amplitude characteristics. Evaluation experiments show that the proposed method outperforms the conventional one in a Perceptual Evaluation of Audio Quality (PEAQ) test but the superiority is not found in a MUltiple Stimuli with Hidden Reference and Anchor (MUSHRA) test.

2aSPb30. Fast multiple moving sound sources localization utilizing sparseness of speech signals. Eiji Sato and Yosuke Tatekura (Shizuoka Univ., 3-5-1 Johoku, Hamamatsu, Shizuoka 432-8561, Japan, e-satoh@spa-lab.eng.shizuoka.ac.jp)

We propose multiple moving sound sources localization with reduced operation time utilizing the sparseness of speech signals, and we demonstrate the efficacy of proposed method in actual environment experiments. Sound source localization is an essential function for tasks such as robot audition, and particularly for sound sources that often move in the actual environments. Sound source localization requires high resolution and real-time processing because they are utilized for post-processes such as sound source separation. The proposed method is based on MUSIC (MUltiple SIgnal Classification) which is known as a sound source localization method with high resolution. However, MUSIC has a large computational complexity. When the signal has large energy over a wideband in the frequency domain, the operation time increases because MUSIC estimates the direction of the sound source at every frequency. Therefore, we attempted to reduce the operation time by using the sparseness of speech signals in which the sound energies exist sparsely in the time-frequency domain. We chose frequencies with large power from frequency characteristics of the observed signals. Sound source localization was performed in the chosen frequencies. The operation time of the proposed method was about 8.1 times faster than the case of utilizing all frequencies.

2aSPb31. Directivity control of a compact circular loudspeaker array based on selected orders of circular harmonic expansion. Koya Sato and Yoichi Haneda (Dept. of Informatics, The Univ. of Electro-Communications, 1-5-1 Chofugaoka, Chofu-shi, Tokyo 182-8585, Japan, Japan, s1630060@edu.cc.uec.ac.jp)

We investigated directivity control of a compact circular loudspeaker array that can be attached to the arm. The loudspeaker array can be used for various wearable systems, such as personal audio reproduction systems. A personal audio system often requires that sound leakage outside the listening area in three-dimensional space be minimized. Digital filters that control directivity are usually designed using the multi-point control method. However, the multi-point method causes large filter gains at low frequencies, owing to unstable inverse matrix calculation, which results in sound distortion during practical use. Many methods that use a regularization parameter to overcome the unstable inverse matrix have been proposed; however, they are difficult to apply directly. Thus, after studying the relationship between orders of circular harmonic expansion and filter gain, we propose a stable digital filter design based on appropriately selected orders of circular harmonic expansion for each frequency band. Directivity simulations conducted using the measured circular loudspeaker array response based on the proposed method indicate that the proposed method achieves a more stable filter gain than the multi-point control method. Further, it controls three-dimensional directivity by sacrificing a narrow directivity at low frequencies.

2aSPb32. Correct phoneme sequence estimation using recurrent neural network for spoken term detection. Naoki Sawada and Hiromitsu Nishizaki (Univ. of Yamanashi, 4-3-11 Takeda, Kofu 400-8511, Japan, sawada@alps-lab.org)

This paper proposes a correct phoneme sequence estimation method that uses a recurrent neural network (RNN)-based framework for spoken term detection (STD). It is important to reduce automatic speech recognition (ASR) errors to obtain good STD results. Therefore, we use a long short-term memory (LSTM), which is one of an RNN architecture, for estimating a correct phoneme sequence of an utterance from phoneme-based transcriptions produced by ASR systems in post-processing of ASR. We prepare two types of LSTM-based phoneme estimators: one is trained with a single ASR system's N-best output and the other is trained with multiple ASR systems' 1-best outputs. For an experiment on a correct phoneme estimation task, these LSTM-based estimators could generate better phoneme-based N-best transcriptions rather than the best ASR system's ones. In particular, the estimator trained with multiple ASR systems' outputs worked well on the estimation task. Besides, the STD system with the LSTM estimator drastically improved STD performance compared to our previously proposed STD system with a conditional random field-based phoneme estimator.

2aSPb33. An integration method of multiple search results for spoken term detection. Yoshino Shimizu (Graduate School of Software and Information Sci., Iwate Prefectural Univ., 2-10-37 Honchodori, Morioka-shi, Iwate-ken 020-0015, Japan, g231o015@s.iwate-pu.ac.jp), Eitaro Iwasaki (Iwate Prefectural Univ., Takizawa-shi, Japan), Shi-Wook Lee (National Inst. of Adv. Industrial Sci. and Technol., Tsukuba-shi, Japan), Kazuyo Tanaka (Univ. of Tsukuba, Tsukuba-shi, Japan), Kazunori Kojima, and Yoshiaki Itoh (Iwate Prefectural Univ., Takizawa-shi, Japan)

We propose a new integration method of multiple search results for improving search accuracy of Spoken Term Detection (STD). A usual STD system prepares two types of recognition results of spoken documents. If a query consists of in-vocabulary (IV) terms, the results using word-based recognizer are used, and if a query includes out-of-vocabulary (OOV) terms, the results using subword-based recognizer are used. The paper proposes an integration method of these two search results. Each utterance has a similarity score included in the search results. The scores of two results for an utterance has been integrated linearly using a constant weighting factor so far. Our preliminary experiments showed the search accuracy using the subword-based results was higher for some IV queries. In the same way, that using the word-based results was higher for some OOV queries. In the proposed method, the similarity scores of the two search results are compared for the same utterance and a higher weighing factor is given to the results

that showed a higher similarity score. The proposed method is evaluated using open test sets, and experimental results demonstrated the search accuracy improved for all test sets.

2aSPb34. The largest Cramer-Rao lower bound for variance estimation of acoustic signals. Hiroki Takahashi (Elec. Eng. Program, Meiji Univ., 1-1-1 Higashimita, Tama-ku, Kawasaki-shi, Kanagawa 214-8571, Japan, ce51038@meiji.ac.jp) and Takahiro Murakami (Dept. of Electronics and-Bioinfomatics, Meiji Univ., Kanagawa, Japan)

The Cramer-Rao Lower Bounds (CRLBs) for variance estimation of acoustic signals is discussed. The CRLB expresses the lower limit of the variance of estimation errors of an unbiased estimator. It is empirically known that the accuracy of variance estimation is improved as the signal length is long because the CRLB for variance estimation is inverse proportion to the length of a signal. Therefore, when the variance of estimation errors is desired to be lower than a certain value, the signal length must be longer than the length that is determined by the CRLB. If the distribution of a signal is known, the CRLB can be easily formulated. In practice, however, acoustic signals follow various distributions and they are unknown in advance. In our research, we focus on the distribution that gives the largest CRLB for variance estimation. In order to find such a distribution, we compare the CRLBs for variance estimation using various signal models. Our results conclude that the Gaussian distribution gives the largest CRLB for variance estimation.

2aSPb35. Novel sound logger system for sound source localization using global navigation satellite system. Chikao Takasugi (Elec. Eng. and Comput. Sci., Shibaura Inst. of Technol., Koike Lab., 3-7-5 Toyosu, Kotoh-Ku, Tokyo 135-8548, Japan, ma16065@shibaura-it.ac.jp), Keisuke Takeyama, Takuya Fukaishi, Yoshikazu Koike (Electron. Eng., Shibaura Inst. of Technol., Tokyo, Japan), Harimasa Hojo (SensorComm Co.,Ltd., Tokyo, Japan), and Tatsuo Matsuno (Matsuno Kikaku, Tokyo, Japan)

Accurate sound localization is a very important technology to solve noise problem in the association with air plane, railroad, or wind-power generation and so on. In order to localize the sound source, direction of arrival (DOA) estimations in a couple of positions at a distance of several hundred meters are required. Generally, the measurement start timing is synchronized in each estimation position using 1PPS time signal from the global navigation satellite systems (GNSSs). However, the slight deviation in each measuring instrument occurs at the start time. Because the sampling clock of ADC (analog digital converter) is not synchronized. Therefore, the authors propose the application of time synchronization signal obtained from GNSS module to a sampling clock on an acoustic measurement in a wide range. Current some GNSS modules provide a pulse multiplied by 1PPS, which we call a time-synchronization signal. For the purpose, the authors have developed a logger apparatus which consists of a microphone amplifier, a GNSS, and an ADC module. The developed logger apparatuses are employed to perform sound source localization experiment in the field. As a result, the precision of sound localization within 2 meters is achieved successfully when the microphone sets at a distance of 10 m.

2aSPb36. Evolutionary optimization of long short-term memory neural network language model. Tomohiro Tanaka (Dept. of Information Processing, Tokyo Inst. of Technol., 4259G2-2, Nagatsuta, Midori-ku, Yokohama, Kanagawa 226-8502, Japan, tanaka.t.bh@m.titech.ac.jp), Takafumi Moriya (NTT Media Intelligence Labs., NTT Corp., Yokosuka-Shi, Japan), Takahiro Shinozaki (Dept. of Information and Commun. Eng., School of Eng., Tokyo Inst. of Technol., Yokohama-shi, Japan), Shinji Watanabe, Takaaki Hori (Mitsubishi Electric Res. Laboratries, Cambridge, MA), and Kevin Duh (Johns Hopkins Univ., Baltimore, MD)

Recurrent neural network language models (RNN-LMs) are recently proven to produce better performance than conventional N-gram based language models in various speech recognition tasks. Especially, long short-term memory recurrent neural network language models (LSTM-LMs) give superior performance for its ability to better modeling word history information. However, LSTM-LMs have complex network structure and training configurations, which are meta-parameters that need to be well tuned to achieve the state-of-the-art performance. The tuning is usually performed manually by humans, but it is not easy because it requires expert knowledge and intensive effort with many trials. In this study, we apply covariance matrix adaptation evolution strategy (CMA-ES) and automate the tuning process. CMA-ES is one of the most efficient global optimization techniques that has demonstrated superior performance in various benchmark tasks. In the experiments, the meta-parameters subject to the tuning included unit types at each network layer (LSTM, RNN, and feed-forward), the number of hidden layers, mini-batch size, and so on. We show that CMA-ES successfully optimizes the meta-parameters of LSTM-LM improving the recognition performance. We also report what are the common characteristics of the well-performing models by analyzing the results of the evolutions.

2aSPb37. Development of a communication system using audio signal using audible frequency bands. Chihiro Terayama, Masahiro Niitsuma, and Yoichi Yamashita (Graduate School of Information Sci. and Eng., Ritsumeikan Univ., Nojihigashi1-1-1, Kusatsu-shi, Shia-ken 525-8577, Japan, terayama-ASJ@slp.is.ritsumei.ac.jp)

This paper proposes a technique of embedding information into audio signal to communicate using loudspeakers and microphones mounted on tablet-type devices. This technique is based on amplitude modulation for high audible frequency bands. The embedded information is represented by existence of sine waves with 11 different frequencies. In consideration of the frequency characteristics of tablet-type devices, two frequency bands to embed information, were chosen: (1) frequencies up to 12 kHz; (2) frequencies up to 16 kHz. The experimental result showed 85.9% of correct decoding in the case of (1), and 76.8% in the case of (2). Moreover, less audible noise was perceived in the case of (2).

2aSPb38. Iterative spectral subtraction based on variable frame processing for musical noise reduction. Yukoh Wakabayashi, Yuki Nagano, Takahiro Fukumori, Masato Nakayama, and Takanobu Nishiura (Ritsumeikan Univ., 1-1-1 Noji-higashi, Kusatsu, Shiga 525-8577, Japan, gr0221ss@ed.ritsumei.ac.jp)

In recent years, speech communication and speech applications like speech recognition are widely used with the development of information equipment. But a speech corrupted by noises makes fluent communication difficult and causes the poor performance of speech applications. Thus, an accurate noise reduction method is necessary to solve these problems. Spectral subtraction (SS) and iterative spectral subtraction (I-SS) are well-known noise reduction methods that subtract estimated noise from noisy speech in the power spectral domain. However, SS and I-SS generate an artificial noise called musical noise. The noise is non-stationary and has an offensive sound. Miyazaki *et al.* have proposed a musical-noise-free SS that determines subtraction parameters such that musical noise is not generated. The method effectively reduces the musical noise but requires estimation of the statistical characteristics of the observed noise. Meanwhile, we tackle the musical noise problem by proposing a new I-SS based on variable frame processing. In this processing, variable frame analysis is carried out in each iteration to reduce the musical noise. Evaluation experiments show that the proposed I-SS achieved a high SDR and resulted in less musical noise compared with constant frame processing.

2aSPb39. Effect of individualization of interaural time/level differences with non-individualized head-related transfer functions on sound localization. Kanji Watanabe, Masayuki Nishiguchi, Shouichi Takane, and Koji Abe (Faculty of Systems Sci. and Technol., Akita Prefectural Univ., 84-4 Ebinokuchi, Tsuchiya, Yuri-Honjo, Akita 015-0055, Japan, kwatanabe@akita-pu.ac.jp)

Head-related transfer functions (HRTFs) are known to include comprehensive auditory cues for sound source positions and show large inter-subject variation. Therefore, individualization of HRTFs is important for highly accurate sound localization systems such as virtual auditory displays. In the present study, we assume that the interaural time deffierence (ITD) and the interaural level difference (ILD) can be estimated from the listener's anthropometric parameters. The individualization of the HRTFs is achieved by replacing the ITDs and ILDs of the non-individualized HRTFs with the

listener's ones. In this report, the effect of the individualization was evaluated by listening experiment. The non-individual HRTFs were obtained by interpolating magnitude response of the listener's own HRTF and the flat ones. The ratios of the interpolation corresponded to the degrees of individuality. Then, the ITDs and ILDs were added to individualize HRTFs. From the results, the effect of the degree of individuality was insignificant, but the front-back confusions were significantly different between HRTFs on ipsilateral side and contralateral side. The results suggested that the individualization of HRTFs with ITDs/ILDs might be influenced by the spectrum of non-individual HRTFs.

2aSPb40. Projection of a virtual speaker into a vehicle using sound field control. Toshiki Yamamura, Yoshio Ishiguro (Nagoya Univ., Furo-cho, Chikusa-ku, Nagoya, Aichi 464-8601, Japan, yamamura.toshiki@g.sp.m.is.nagoya-u.ac.jp), Takanori Nishino (Mie Univ., Tsu, Japan), and Kazuya Takeda (Nagoya Univ., Nagoya, Japan)

In this study, we propose a system which projects a virtual speaker into a vehicle by controlling the sound field of the interior of the vehicle. The system uses a sound field reproduction technique to reproduce a point source in the vehicle representing a virtual speaker, enabling drivers to communicate with virtual speakers as if they were riding in the same vehicle. The system is based on a sound field reproduction technique known as Wave Field Synthesis (WFS), which can represent the depth of a sound source by reproducing its wavefront precisely. However, since WFS is generally implemented in an ideal environment, the influences of reflection and reverberation are not generally taken into consideration. Therefore, in order to apply WFS to an in-vehicle system, it is necessary to investigate the effects of reflection and reverberation on WFS within a vehicle. To do so, we compare the accuracy of the perception of locations of sound point sources reproduced using WFS with a 32 channel loudspeaker array in both a vehicle and a low reverberation room. We also compare perception accuracy of the proposed system with perception accuracy when a loudspeaker is placed in the same position as the virtual speaker.

2aSPb41. The mathematical paradigm of "copula" to model a faded signal received by multiple sensors. Shih-Hao Huang, Mong-Na L. Huang (Dept. of Appl. Mathematics, National Sun Yat-sen Univ., Kaohsiung, Taiwan), and Kainam T. Wong (Dept. of Electron. & Information Eng., Hong Kong Polytechnic Univ., Hong Kong Polytechnic University, Hung Hom KLN, Hong Kong, ktwong@ieee.org)

In sonar, radar, or wireless communication that deploys multiple sensors in the receiver for a fading channel, the received signal (at each time instant) may be modeled as a stochastic vector whose elements correspond to the various sensors of the receiver. This random signal's spatial statistics may be described (at each time instant) by a multivariate probability distribution. For this multivariate distribution to have ARBITRARILY different univariate marginal distributions and/or an ARBITRARY spatial correlation function—the mathematical paradigm of "copula" has recently been advanced in Huang, Huang, Wong, and Tseng, "Copula—to model multi-channel fading by correlated but arbitrary Weibull marginal, giving a closed-form outage-probability of selection-combining reception," IET Microwaves, Antennas & Propagation, December 2015. That seminal work shows how the mathematical paradigm of copula can allow unprecedented flexibility in modeling and can offer unparalleled mathematical simplicity in expressing the receiver's outage probability in a closed and simple mathematical form explicitly in terms of the model parameters. That reference focuses on a particular copula called the survival-Gumbel copula. This paper will explore other copulas.

2aSPb42. Multiple-input multiple-output underwater acoustic communications by using a quasi-orthogonal group space-time architecture. Chen Zhang, FengXiang Ge (Beijing Normal Univ., No. 19, XinJieKouWai St., HaiDian District., Beijing 100875, China, fox@mail.bnu.edu.cn), Lin Dai (Univ. of Hong Kong, Hong Kong, China), and Huiling Zhu (Univ. of Kent, Kent, United Kingdom)

A Quasi-orthogonal Group Space-Time (QoGST) architecture is presented in this paper for better diversity and multiplexing tradeoff gains in Multiple-Input Multiple-Output (MIMO) underwater acoustic (UWA) communications. Both QoGST and Group Layered Space-Time (GLST) are the combination of space-time block coding (STBC) and layered space-time coding. For GLST, the STBC is taken just in each group. For QoGST, however, all groups are encoded together via inter-group STBC, which can effectively suppress the interference between groups with the orthogonal STBC. At the receivers, the STBC decoding is performed and thus we have the higher diversity gains for the group detection. A statistical model of UWA channel which is presented by Milica Stojanovic and contains variations caused by small- and large-scale effects is considered. Simulation results show that the frame error rate (FER) of QoGST is much lower than that of GLST, which means the QoGST has the higher diversity gains. Furthermore, the lower bound diversity and multiplexing tradeoff of QoGST is coincides with that of GLST and the upper bound diversity and multiplexing tradeoff of QoGST is closer to the optimal values.

2aSPb43. Comparing frequency-difference beamforming and delay-and-sum beamforming as an estimator of direction-of-arrival. Eldridge Alcantara, Les Atlas (Elec. Eng., Univ. of Washington, Seattle, 3040 17th Ave. West #404, Seattle, WA 98119, eealcant@uw.edu), and Shima Abadi (Mech. Eng., Univ. of Washington, Bothell, Bothell, WA)

Direction-of-arrival estimation (DOA) is an important and long-studied problem in array signal processing. Existing algorithms such as conventional delay-and-sum beamforming are limited in performance and become inadequate for a high-frequency broadband source and when the receiving linear array is sparse. Abadi, Song, and Dowling (2012), however, showed that DOAS estimation is possible in this case with an algorithm they introduced called frequency-difference beamforming. This was demonstrated empirically in their previous work using a real dataset. In addition, the resemblance between frequency-difference beamforming and delay-and-sum beamforming was briefly discussed. The purpose of this study is to analyze this relationship further and determine how well frequency-difference beamforming compares to delay-and-sum beamforming as an estimator of DOA. The approach taken in this study is to explicitly formulate the estimator equation for frequency-difference beamforming and compare it mathematically to the estimator equation of delay-and-sum beamforming. The equations are compared for the case of one DOA and two DOAs. For both cases, we show the effect of both source frequency and source bandwidth on frequency-difference beamforming. Using these results, we probe the following question: what are the conditions on the source frequency or source bandwidth for which frequency-difference beamforming will outperform delay-and-sum beamforming?

Session 2aUW

Underwater Acoustics and Acoustical Oceanography: Underwater Acoustic Studies in Asian Seas II

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Chair's Introduction—7:55

Contributed Papers

8:00

2aUW1. Effect of internal waves on acoustic energy structure in the northern East China Sea. Jungyong Park, Sangkyum An (Naval Architecture and Ocean Eng., Seoul National Univ., Bldg. 36, Rm. 212, 1, Gwanak-ro, Gwanak-gu 151 - 744, Seoul KS013, South Korea, ioflizard@snu.ac.kr), Youngmin Choo, and Woojae Seong (Naval Architecture and Ocean Eng. and Res. Inst. of Marine Systems Eng., Seoul National Univ., Seoul, South Korea)

Relation between depth direction acoustic energy and sound speed fluctuation caused by internal wave activity observed during SAVEX15 experiment is investigated. In the SAVEX15 region (northern East China Sea with water depths around 100 m), underwater sound channel was observed during May 2015 along with extensive internal wave activities. When the source is near the sound channel axis, most of the acoustic energy is trapped in the channel. However, internal waves change the sound speed profile, which results in spreading the acoustic energy over depth. Measured data during the experiment is used to investigate the relation between acoustic energy structure and internal wave activity. Parameters reflecting their variabilities are defined. Two main parameters are standard deviation of the acoustic energy integrated over depth and the sound speed gradient at the channel axis, which reflects the internal wave activity. The correlation of two parameters is used to show the effect of internal waves on the changes of acoustic energy distribution.

8:15

2aUW2. Measurements of mid-frequency bottom interacting signals in Jinhae bay located on the southern coast of Korea. Jee Woong Choi, Hyuckjong Kwon, Su-Uk Son (Dept. of Marine Sci. and Convergence Eng., Hanyang Univ., 55 Hanyangdaehak-ro, Ansan 426-791, South Korea, choijw@hanyang.ac.kr), Sungho Cho (Korea Inst. of Ocean Sci. & Technol., Ansan, South Korea), Joung-Soo Park, and Jooyoung Hahn (Agency for Defense Development, Changwon, South Korea)

Acoustic bottom interacting measurements were conducted in shallow water (nominal water depth of 60 m) in Jinhae bay, southern coast of Korea, using 4 to 12 kHz CW signals in May 2015 and 2016. The surficial sediment at a site is mainly composed of silty clay with a mean grain size of 8 phi. Since the seafloor is relatively soft, the bottom-bounced path was very weak compared to direct and sea-surface-bounced paths. On the other hand, a strong arrival reflected from the sub-sediment layer was received after the bottom-bounced arrival especially at lower frequencies. The arrival time difference between the arrival reflected from water-sediment interface and that reflected from the second interface is used to estimate the sound speed in the surficial sediment layer. In addition, the bottom loss as a function of grazing angle are estimated using the bottom-bounded path. Finally, the results are compared to the sedimentary structure imaged by chirp sonar survey. [Work supported by Agency for Defense Development, Korea.]

8:30

2aUW3. Acoustical characteristics of snapping shrimp noise measured during SAVEX 15 and its use as a natural source to estimate sea surface properties. Myungkwon Ko, Dae-Hyuk Lee, Jee Woong Choi (Marine Sci. and Convergence Eng., Hanyang Univ., Hanyangdaehakro 55, Ansan, Gyeonggi do 15588, South Korea, ko.myungkwon@gmail.com), and Sungho Cho (Korea Inst. of Ocean Sci. & Technol., Ansan, South Korea)

The Shallow-water Acoustic Variability Experiment (SAVEX15) was conducted at a site in the Northern East China Sea (ECS), about 100 km off the southwest of Jeju Island, Korea in May 2015. The purpose of SAVEX15 was to obtain acoustic and environmental data for studying on oceanography, underwater communication, and acoustic propagation characteristics in the ECS. The experiment utilized a vertical line array (VLA) of 16 elements spanning 56.25 m of a 100 m water column. During the experiment, a large number of unintended snapping shrimp noises were recorded by the receiving system. In this talk, the acoustic properties of the snapping shrimp noise are presented. In addition, these noises are used as natural acoustic sources to estimate the sea surface properties from the sea-surface bounced paths. Finally, the results are compared with the sea surface conditions collected by a wave buoy operated during the experimental period. [This work was jointly supported by KRISO, KIOST, MPL, and ONR, and partially supported by ADD, Korea.]

8:45

2aUW4. Comparison of underwater communication performances in two shallow-water sites with different sediment types located off the coast of Korea. Sunhyo Kim, Kanghoon Choi, Jee Woong Choi (Ocean Sci. and Convergence Eng., Hanyang Univ., Hanyang daehakro 55, Ansan 15588, South Korea, sunhyo4485@hanyang.ac.kr), Joung-Soo Park (Agency for Defense Development, Changwon, South Korea), and Sea-Moon Kim (Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea)

Shallow water waveguide characterized by multipath channel produces a significant delay spreading of transmitted signals, which is referred to as inter symbol interference (ISI). Since the ISI results in distortion of communication signals, many studies to reduce the effect of ISI have been conducted. For successful underwater communication, it is important to understand the correlation between the spatial and temporal properties of ISI and communication performance. Underwater acoustic communication experiments were conducted in two different seafloor environments with relatively fine-grained and coarse sediments. The experimental geometries were the same; water depths of both sites were about 40 m and source

depths were about 32 m. Communication signals were measured by a four-channel receiving array, covering waters 5 to 35 m in depth. Sound speed profiles were measured by CTD casts and the surficial sediment samples were taken by a grab sampler. The received communication signals were demodulated using the time reversal, phase lock loop and decision feedback equalizer techniques. In this talk, the communication performances at two different sites are presented and compared with the channel impulse responses at two sites. [Work supported by ADD(UD140001DD) and KRISO(PMS3310).]

9:00

2aUW5. Measurements of high frequency acoustic transmission loss during SAVEX15. Su-Uk Son, Jee Woong Choi (Dept. of Marine Sci. and Convergence Eng., Hanyang Univ., 55 Hanyangdaehak-ro, Sangnok-gu Ansan, Gyeonggi-do 15588, South Korea, suuk2@hanyang.ac.kr), Seung Woo Lee, SungHyun Nam (Res. Inst. of Oceanography/School of Earth and Environ. Sci., Seoul National Univ., Seoul, South Korea), and Sungho Cho (Korea Inst. of Ocean Sci. & Technol., Ansan, South Korea)

Transmission loss measurements in high frequency range along with the ocean environmental measurements were made during the Shallow-water Acoustic Variability EXperiment 2015 (SAVEX15) in the south of Jeju Island, Korea, where the water depth is about 100 m. Two vertical line arrays (VLAs) were moored with a distance of 5.5 km, covering the water column between 24 and 80 m. Continuous waves in a frequency band of 10-25 kHz were transmitted from a vertical source array at depths of 30-45 m. Sound speed and temperature profiles were measured using a standard/stationary conductivity-temperature-depth (CTD), underway CTD, and temperature loggers attached to the two VLAs. The measured sound speed profiles consistently reveal a sound speed minimum layer (SSML) at depths between 30 to 50 m. During the SAVEX15 period, the perturbations in the SSML derived by various kinds of internal waves were observed. In this talk, the temporal fluctuations of the measured transmission loss are presented, and the results are discussed in comparison with the internal wave field in the region. [Work supported by ONR, KIOST, SNU, and ADD.]

9:15

2aUW6. Sound focusing in shallow water using a vertical phased array. Dayong Peng, Juan Zeng, Haijun Liu, Haifeng Li, and Wenyao Zhao (Inst. of Acoust., Chinese Acad. of Sci., No. 21, North 4th Ring Rd., Haidian District, Beijing 100190, China, pengdayong@mail.ioa.ac.cn)

Vertical phased array is an efficient and complex equipment in underwater acoustics. Single-mode excitation and time-reversal sound focusing are two typical techniques based on vertical phased array, but both of them have some shortcomings. In order to realize sound focusing at variable distance and depth using vertical phased array, two techniques are proposed in this paper. One of them is the Green's function construction of target sound field based on normal mode eigen-function and attenuation coefficient extraction, and the other is controllable sound focusing based on optimal target sound field determination. Simulation and experiment results are presented. This technique indicates a method for objective depth detecting in shallow water.

9:30

2aUW7. Geoacoustic inversions in the East China Sea for a 3D sediment model. Gopu R. Potty, James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), Stan E. Dosso (Univ. of Victoria, Victoria, BC, Canada), Jan Dettmer (Dept. of GeoSci., Univ. of Calgary, Calgary, AB, Canada), and Julien Bonnel (Ensta Bretagne, Brest, France)

Geoacoustic inversions using wide-band acoustic sources (WBS) deployed during the Asian Seas International Acoustic Experiment (ASIAEX) along a circular path of radius 30 km centered on a vertical hydrophone array will be used to construct a pseudo 3D model of the seabed sediments. The acoustic data used in the inversions consist of modal dispersion as estimated using a warping transform technique. Existing information about the bottom includes historic data, seismic records, and gravity and piston cores which indicate significant spatial variability in the thickness of sediment layers and types of sediment. The geoacoustic inversion approach is based on trans-dimensional Bayesian methodology in which the number of sediment layers is included as unknown in addition to the layer parameters. A sediment-acoustic model, such as viscous grain shearing, will be used to estimate the wave parameters corresponding to the sediment parameters. One-dimensional (depth-dependent) inversions will be applied along the various acoustic propagation paths to construct a pseudo 3D sediment model using interpolation. [Work supported by the Office of Naval research.]

9:45–10:00 Break

10:00

2aUW8. Acoustic property of sound production by fat greenling (*Hexagrammos otakii*) in spawning season. Naoto Matsubara, Hiroyuki Munehara (Hokkaido Univ., 3-1-1, Minato-cho, Hakodate, Hokkaido 041-8611, Japan, naotomatsubaraf91@gmail.com), Ryuzo Takahashi (Japan Fisheries Res. and Education Agency, National Res. Inst. of Fisheries Eng., Kamisu, Japan), Tomonari Akamatsu (Japan Fisheries Res. and Education Agency, National Res. Inst. of Fisheries Sci., Yokohama, Japan), Kazuki Yamato, Ikuo Matsuo (Tohoku Gakuin Univ., Sendai, Japan), Nobuo Kimura, Kazuyoshi Maekawa, and Hiroki Yasuma (Hokkaido Univ., Hakodate, Japan)

Fat greenling (*Hexagrammos otakii*) is one of the important demersal fishes in Japanese local fisheries. Male fishes come over to neritic zone in spawning season, and make the territory as the spawning bed. It was suggested that fat greenling produces specific sounds during the protection of territory.The objective of this study was to examine acoustic properties of male sound production in spawning season as the basic data for estimating habitat density and activity by the passive acoustic methods. Field recordings were conducted during spawning season between Sept. and Nov. in 2014 and 2015. In each experiments, a specialized recorder was set by a territory throughout the breeding process, including courtship, spawning, and egg care. We extracted the acoustic properties such as the frequency of sound production, pulse duration, pulse number, call duration, and the frequency of sound production from recording data. There were different patterns of sound properties and their occurrence frequencies between before and after spawning. It was suggested that fat greenling changes the frequency and property of sound production depending on the situation like courtship, spawning, and protecting eggs.

10:15

2aUW9. Acoustic tracking in shallow coastal areas on spawning migrations of pacific herring (*Clupea pallasii*) using ultrasonic telemetry. Makoto Tomiyasu (Graduate school of Environ. Sci., Hokkaido Univ., Hakodate-city, Benten-cho, 20-5, Hokkaido 040-0051, Japan, lion-heart@eis.hokudai.ac.jp), Hokuto Shirakawa (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate-city, Hokkaido, 20-5 Benten-cho, Japan), Yuki Iino (School of Fisheries Sci., Hokkaido Univesity, Hakodate-city, Hokkaido, Japan), and Kazushi Miyashita (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate-city, Hokkaido, Japan)

The behavioral ecology of small pelagic fish has been studied by acoustic method for a long time. In particular, on pacific herring (*Clupea pallasii*), the behavioral ecology in shallow coastal areas which is observed spawning migration should be understood for the resource management and reproduction protection. In this study, supersonic telemetry was used for behavioral tracking of herring at shallow coastal areas. For the tracking, herring of regional group (n = 23) at Akkeshi bay (Depth ave. 5.78 $\pm$ 2.15 m) in east Hokkaido were used. On the horizontal aspect, herring showed two migration patterns (pattern1: staying at the bay, pattern2: migration to Akkeshi lake (ave. 0.84 $\pm$ 0.13 m) which is brackish water lake adjacent the bay). The lake is the main spawning grounds for herring, therefore the pattern 2 is indicated to be for spawning, and the pattern1 is before or after spawning. On the vertical aspect, diurnal distribution patterns were shown at the bay (day: ave. 5.41 $\pm$ 3.78 m, night: ave. 3.83 $\pm$ 2.60 m). Besides, the rapid horizontal migration was mainly observed at night. From these, the distribution patterns may influence on efficient migration at night without predation risk. Migration span at shallow coastal areas is estimated to be comparatively short, during 4-5 days.

10:30

2aUW10. Regional variations of *Saccharina japonica* forests distribution around the Tsugaru Strait by acoustic method. Huamei Shao (Graduate school of Environ. Sci., Hokkaido Univ., Bentencho. Hakodate City, Hokkaido, Hakodate, Hokkaido 0400043, Japan, huamshao@ees.hokudai.ac.jp), Kenji Minami (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Hokkaido, Japan), Yoshikazu Fujikawa (Div. of Fisheries Infrastructure Aomori Prefecture, Fisheries Bureau, Aomori, Japan), Takashi Yusa (Aomori Prefectural Industrail Technol. Res., Fisheries Res. Inst., Aomori, Japan), Norishige Yotsukura (Field Sci. Ctr. for Northern Biosphere, Hokakido Univ., Sapporo, Japan), Masahiro Nakaoka (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Akkeshi, Japan), and Kazushi Miyashita (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Japan)

As the global warming, commercially important species *Saccharina japonica* disappeared, what is called sea desertification occurred in some coastal waters around the Tsugaru Strait recently. For the sustainability of coastal ecosystem and resources use, understanding the distribution and variation of *S. japonica* forests is necessary. Surveys for the distribution of *S. japonica* forests by acoustic method were conducted in four coastal waters: ① Osatsube, Hakodate, the Pacific Ocean side of Hokkaido; ② Ishizaki, Hakodate in Tsugaru Strait of Hokkaido; ③ Okoppe, Oma, in Tsugaru Strait of Aomori Prefecture; ④ Shiriyazaki, Higashidori, the Pacific Ocean side of Aomori Prefecture. Thickness and spatial distributions of seaweed were obtained by acoustic method and density of sea urchins by visual observations. In Osatsube coastal water, main species was *S. japonica*and which forests were most among survey regions. In Ishizaki and Okkope coastal waters, *Sargassum spp.* was also luxuriant except *S. japonica*. In Shiriyazaki coastal water, *S. japonica* dominated and a greater number of sea urchins were observed than other regions. In regions where little benefit from Oyashio Current, *Sargassum* spp. grew thickly when sea urchins were rare whereas sea urchin barren occurred when sea urchins with high density.

10:45

2aUW11. Annual change in the distribution of *Sargassum horneri* using a quantitative echo sounder in Yamada Bay, Iwate, Japan. Kenji Minami (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., 20-5 Benten-cho, Hakodate, Hokkaido 0400051, Japan, kminami@fish.hokudai.ac.jp), Chihomi Kita (Graduate school of Environ. Sci., Hokkaido Univ., Hakodate, Hokkaido, Japan), Masayuki Sawai (Sanriku Fisheries Res. Ctr., Iwate Univ., Kamaishi, Iwate, Japan), Hokuto Shirakawa (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Hokkaido, Japan), Huamei Shao, Makoto Tomiyasu (Graduate school of Environ. Sci., Hokkaido Univ., Hakodate, Hokkaido, Japan), Motoki Kobayashi, and Kazushi Miyashita (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Hokkaido, Japan)

Sargassum horneri, a type of seaweed, has attracted attention as a new fisheries resource in Japan because it contains large amounts of functional ingredients. Yamada Bay (16 km^2), in Iwate, Tohoku, Japan, was seriously damaged by the 2011 tsunami, and a sustainable new fisheries resources would aid in the reconstruction of local industries. In this study, we estimated annual change in the distribution of *S. horneri* using a quantitative echo sounder for sustainability of *S. horneri* in Yamada Bay. Surveys were conducted in spring (during the growth season) 2013, 2014, and 2015. The acoustic data were obtained using a 120 kHz quantitative echo sounder (KCE300, Sonic Co.). *S. horneri* were detected on the rope of aquaculture facilities and the rock reef in the coastal sea area. Growth conditions were better in 2013 than those in 2014 and 2015. In 2013, water temperature in Yamada Bay was lower than in 2014 and 2015 because of the influence of cold water from the strong Oyashio Current. It was concluded that primary production of *S. horneri* in Yamada Bay is influenced by the Oyashio Current, causing their abundance to fluctuate annually.

11:00

2aUW12. Accuracy verification of tilt-angle estimation by split-beam echosounder: Comparison with acceleration data-logger data. Hokuto Shirakawa, Kenji Minami (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., 20-5 Benten-cho, Hakodate 040-0051, Japan, dipper@fish.hokudai.ac.jp), Yohei Kawauchi (Seikai National Fisheries Res. Inst., Fisheries Res. Agency, Nagasaki, Japan), Makoto Tomiyasu (Graduate School of Environ. Sci., Hokkaido Univ., Hakodate, Japan), Yuichi Tsuda (Bluefin Tuna Resources Div., National Res. Inst. of Far Seas Fisheries, Fisheries Res. Agency, Shizuoka, Japan), Takeru Umetsu, Huamei Shao (Graduate School of Environ. Sci., Hokkaido Univ., Hakodate, Japan), Motoki Kobayashi (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Japan), Yuki Iino (School of Fisheries Sci., Hokkaido Univ., Hakodate, Japan), Masahiro Ogawa (Graduate School of Environ. Sci., Hokkaido Univ., Hakodate, Japan), and Kazushi Miyashita (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Japan)

Fish tilt-angle is one of the most important factors for fish target strength (TS) estimation, because the little change varies TS value dramatically. Split-beam echosounder collects a fish position with 3-D coordinates. Hence, if fish swim within acoustic beam continuously, the tilt-angle can be estimated from the gap between two positions easily. However, the accuracy of the estimation depends on performance of the echosounder. We verified accuracy of the estimated tilt-angle by comparing high-resolution angle from an acceleration data logger. Experiment was conducted at a large experimental tank (W:10m x D:5m x H:6m) in HRCFO. A 2-axis acceleration data logger (φ15 mm, L: 53 mm, M190L-D2GT, Little Leonardo, Inc.) was tethered into the tank, and towed at respective angle from -50 to 50 degree (1 degree pitch). Positions of the logger echos (from KCE300, Sonic Co., 120kHz) were exported by algorithms of Echoview7.1. After this process, the tilt-angles were calculated. 28 logger echos were identified from the acoustic data, and the estimated tilt-angles was from -45.59° to 32.87 °. Significant correlation of tilt-angles (logger vs. acoustic data) was found ($p >$ 0.01, r^2 = 0.90). This result shows high precision of the estimated tile angle from the split-beam echosounder data.

11:15

2aUW13. Distributional factors of juvenile Walleye Pollock in the Musashi Bank area, northern Japan Sea. Takeru Umetsu (Graduate School of Environ. Sci., Hokkaido Univ., Bententown 20-5, Hakodate, Hokkaido 040-0051, Japan, ume.take33@gmail.com), Yuichi Tsuda (Bluefin Tuna Resources Div., National Res. Inst. of Far Seas Fisheries, Shizuoka, Japan), Yasushi Ito (The Japanese Inst. of Fisheries Infrastructure and Communities, Chiyoda, Japan), Hokuto Sirakawa, Kenji Minami, and Kazushi Miyashita (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Japan)

Most of juvenile walleye pollock (*Gadus chalcogrammus*) in the northern Japan Sea migrate to the bottom layer with 150-200 m depth of Musashi bank area in summer. Clarification of distribution factors of nursery area after the migration is important for resource management. We examined the factors of juvenile pollock after migration to bottom in summer 2014 and 2015. The distribution of density of juvenile pollock was obtained by quantitative echo sounder. The distribution of bottom sediment was discriminated by the side scan sonar. The bottom backscattering strength and sediment data were jointed to analyze bottom sediment of quantitative echo sounder survey transect line based on the Roxann method. In the result, bottom sediment was separated into clay, sand, and boulders. Distribution density of juvenile pollock was high in boulders area. Additionally, the density of juvenile pollock was higher at northward upslope than gentle landform and else way upslope in all boulders area. It is considered that northward ocean flow collided with the northward upslope of boulders make upward flowing and stagnate water, and the water forms the retention space of prey plankton that is favorable for juvenile pollock.

11:30

2aUW14. Tracking whales in the Indian Ocean using the comprehensive test ban treaty hydrophone data. David A. Lechner (Mech. Eng., The Catholic Univ. of America, 520 Michigan Ave., NE, Washington, DC 20064, 66Lechner@CUA.edu), Shane Guan (Mech. Eng., The Catholic Univ. of America, Silver Spring, MD), and Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This paper will present preliminary results and status in development of an automated tracking algorithm for whale signals using the Comprehensive Test Ban Treaty (CTBT) sensor network. The data from the CTBT stations is being made available for academic work, and the stations are also detecting bioacoustics sounds as well as seismic activity. We present an approach used to process several channels of the acoustic data collected off Cape Leeuwin, Australia, and automatically search for biologic activity using cross-correlation and Time-Difference-of-Arrival methods. After developing a graph of locations for peak correlation amplitude, the locations are mapped and grouped using location and proximity, in order to develop motion tracks and compare coherence of the signals within a grouping and between groupings, thus identify individual whales based on the differences in location and coherence levels.

11:45

2aUW15. Geoacoustic inversion base on modeling ocean-bottom reflection wave. Licheng Lu, Qunyan Ren, and Li Ma (Key Lab. of Underwater Acoust. Environment, Inst. of Acoust., Chinese Acad. of Sci., No. 21 West Rd.,North 4th Ring Rd., Haidian District, Beijing 100190, China, luce_1983@sina.com)

Ocean-bottom parameters surveying experiment was conducted at two sites in Chinese South Sea. A vertical array of four elements was used to measure reflection waves from ocean-bottom with the explosive charges. The ocean-bottom reflection waves are processed by analyzing the arriving time to identify the sediment layers, and then, a geoacoustic model with several numbers of homogeneous fluid sediment layers overlying a homogenous substrate half space is constructed. Using the direct arriving wave, an approach that matches the measured and modeled ocean-bottom reflection wave is to invert the bottom velocity and density profile. A low velocity layer is obviously observed at one site and a complex seabed profile at another site is established. The modeled wave form is compared with the measured one and shows a relative agreement.

TUESDAY MORNING, 29 NOVEMBER 2016 CORAL FOYER, 9:00 P.M. TO 5:00 P.M.

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/

Session 2pAA

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

Kenneth W. Good, Cochair
Armstrong, 2500 Columbia Av, Lancaster, PA 17601

Takashi Yamakawa, Cochair
Yamaha Corporation, 10-1 Nakazawa-cho, Naka-ku, Hamamatsu 430-8650, Japan

Catherine L. Rogers, Cochair
Dept. of Communication Sci. and Disorders, Univ. of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. The impact of architecture on both the intelligibility and privacy of speech. Kenneth P. Roy (Bldg. Products Technol. Lab, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrong.com)

Architects design the "visual space" that we can all see, and this we call architecture. Acousticians upon entering this visual space invariably clap their hands, and as such are assessing the resulting "aural space." Acoustics is the result of architecture, and so, speech clarity is determined by architecture—meaning the size, shape, and surface treatments chosen by the architect to deliver the mission for which the space is intended. We can look at (model) or listen to (field evaluate) the impulse response of a room, and from this determine the degree of speech clarity that is projected to any listening location. Designing for good speech intelligibility within a space means providing adequate speech clarity and then protecting this signal with a high signal-to-noise ratio (S/N) to ensure that the signal can be heard above the ambient noise. If on the other hand we seek to limit the speech intelligibility (privacy) between spaces, we need to design the architecture to limit the intrusion of speech between spaces taking into consideration the architectural choices as barriers to sound transmission via walls, ceilings, and floors. Examples of design approaches in both cases of intelligibility and privacy will be shown.

1:25

2pAA2. Fundamentals of speech privacy in the built-environment. Kenneth W. Good (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

Since this session intends to bring together experts in speech, architecture, and hearing, this paper will present the fundamentals of Speech Privacy from the architectural perspective. Considering the room as a transfer-function from the talker to the listener, I will cover the importance of various architectural components, measurements, and design considerations for the built-environment. The intent is sharing practices and perspective across disciplines not necessarily new discoveries in the field.

1:45

2pAA3. Effects of dominance of the first-arriving sound on speech intelligibility in reverberant sound fields. 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 Yasushi Hoshino (Nippon Sheet Glass Environment Amenity Co., Ltd., Tokyo, Japan)

Generally speaking, the energy of the direct sound exceeds that of reflections in rooms for listening to speech. In this case, early-reflections enhance loudness of the direct sound. It is well-known that the useful-to-detrimental sound ratio, in which useful is the sum of the direct sound and early-reflections and detrimental is late-reflections, can predict speech intelligibility. On the other hand, considering speech leakage from the adjacent room, the first-arriving sound is attenuated by the boundary wall, and it is sometimes too weak to trigger the temporal integration of loudness. This would change the time structure of the useful energy. In the present study, prediction of speech intelligibility in such particular situations was investigated. Word intelligibility tests were performed in reverberant sound fields varying reverberation time, direct-to-reverberation ratio, and rise time at the onset of reverberation sound. The results demonstrated that prediction accuracy of the useful-to-detrimental ratio was still in a practical range though the parameters systematically affect the prediction error.

2:05

2pAA4. An approach to the design of drama theaters for speech clarity. Gregory A. Miller, John Strong, Carl Giegold (Threshold Acoust., LLC, 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, gmiller@thresholdacoustics.com), and Laura C. Brill (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE)

In drama theaters, the unamplified human voice is used to convey both information and nuanced emotional cues to large groups of listeners. Supportive sound reflective surfaces are relatively distant from the actors, and the inherent (and desirable) reverberance of a larger space run counter to many accepted practices for achieving speech clarity in smaller spaces. Some theater configurations, such as thrust and in-the-round, are even arranged so that actor faces away from large segments of the audience at any given time, offering further challenges to the connection between actor and audience. This paper will present an understanding of the way that people listen to speech in large rooms, and approaches to room design to support the clarity of unamplified speech. A case study will be presented describing the development of theater design from initial concept through computer modeling and finally with three dimensional measurements and visualizations of the acoustic environment in the completed room.

2:25

2pAA5. The highly exaggerated attributes of the pop vocal in sound recordings. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

The voice in sound recordings must generally reign supreme over the rest of the multitrack arrangement. The loudspeaker mediates its interaction with the architecture. The entire upstream production chain anticipates the challenge. Recording space, microphone choice, and signal processing are carefully coordinated to maximize the audibility of multiple attributes the voice — intelligibility, emotion, musicality, and tonal quality. The highly manipulated approach to speech for sound recording can shed light on approaches taken in the built environment for the talker/listener interface.

2p TUE. PM

2:45

2pAA6. Basic study based on diffuse sound field model on relationships among architectural conditions, intelligibility, and speech privacy. Takashi Yamakawa (Yamaha Corp., 10-1 Nakazawa-cho, Naka-ku, Hamamatsu, Shizuoka 430-8650, Japan, takashi.yamakawa@music.yamaha.com)

Rates of both speech transmission and speech privacy are usually estimated on the basis of intelligibility score. There are many physical indexes proposed to stand for intelligibility, like STI, AI, U50, SNRuni32, and so on. It will be reported that the results of calculations of intelligibility indexes with various room conditions based on classical diffuse sound field model, which shows relationships between degrees of absorption, sound blocking, background noise, distance between a target source and listeners, and intelligibility indexes. Some graphs of intelligibility versus each architectural condition will be provided. For example, architects could know how much each countermeasure, which includes adjusting absorption, set a barrier, adding masking noise, acquiring space, works for speech privacy to see the graphs. It is expected that the results will give us the basic guideline to develop architecturally oral communication and speech privacy.

3:05–3:20 Break

3:20

2pAA7. Prediction of speech intelligibility through the use of simulation tools. Bruce C. Olson and Ana M. Jaramillo (Olson Sound Design, LLC, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu)

In a previous paper, we presented the design of a new sound system for the House of Representatives in the State of Minnesota, with the help of simulations. After the installation was completed, we performed new measurements to compare the predicted speech intelligibility with the resulting one. We will discuss differences between the two sets and how to better use simulation tools for the prediction of speech intelligibility and related metrics in this key project as well as others with similar requirements.

3:40

2pAA8. Controlled minimum background sound, the missing link in acoustic design. Ric Doedens (K.R. Moeller Assoc., Ltd., 1050 Pachino Court, Burlington, ON L7L 6B9, Canada, rdoedens@logison.com)

Current industry standards related to controlling background sound in occupied spaces are all based on “not to exceed” maximum levels. At the other end of the scale, it is assumed that we cannot predict with any degree of certainty how low the background sound level may descend, what its spectral composition is, and when or how it may vary. Given that the “signal to noise ratio” is fundamental to all communication based acoustic design, the absence of this knowledge presents significant challenges. As a consequence, we adapt our acoustic designs to compensate for our inability to control minimum noise levels by over-specifying STC, CAC, and NRC values in order to obtain the desired results. And yet, precise generation of controlled minimum background sound levels is readily available. This session explores the negative impact of traditional compensatory acoustic design practises, the significant value of incorporating predictable/consistent minimum background sound levels into acoustic design planning, and makes a case for establishing minimum background sound level industry standards.

4:00

2pAA9. Experimental studies on speech level and room acoustic condition focusing on speech privacy. Hyojin Lee and Shinichi Sakamoto (Inst. of Industrial Sci., Univ. of Tokyo, Meguro-Ku, Komaba 4-6-1, #Ce401, Tokyo 1538505, Japan, leehj@iis.u-tokyo.ac.jp)

Recently, concerns about speech privacy to avoid oral information leakage at medical facilities are increasing continuously in Japan. In order to achieve the requirements of the speech privacy, standardization has also been discussed. On discussing the speech privacy, speech level is one of the most important factors. This study investigated the speech levels of Japanese talkers and the relationships between the speech levels and the room acoustic conditions (reverberation time, ratio of direct and reflected sound, and loudness of background noise). Subjective experiments were conducted under three conversation conditions; normal voice level with a partner sitting 1 m away, raised voice level with the partner sitting 3 m away, and voice level considering overhearing about the contents by a third party. Acoustic environments were reproduced in an anechoic room with a binaural sound field simulation system. As a result, the speech levels of the three conversation conditions were varied. In the simulated test conditions, it was revealed that the room acoustic conditions affected on the speech levels, but the changes of the levels were small.

4:20

2pAA10. Intelligibility and acoustic analysis of speech production in synthetic and natural reverberant and noisy environments: A case study of Parkinson's disease speech. Mark D. Skowronski (Commun. Sci. and Disord., Univ. of South Florida, 4202 East Fowler Ave., PCD 1017, Tampa, FL 33620, skowronski@usf.edu)

Our understanding of the effects of noise and reverberation on speech production and intelligibility is enhanced by the study of speech from talkers with Parkinson's disease (PD) due to the fact that PD speech spans a wide range of intelligibility in quiet and that talkers with PD also span a wide range of speech motor control for compensation. A previous study on the intelligibility of Parkinson's disease (PD) speech in a synthetic reverberant and noisy environment showed that speech intelligibility suffered primarily due to noise although the effects of reverberation were also significant [Shrivastav, ASHA 2011]. A follow-up study in natural reverberant and noisy environments (allowing for talker compensation) found that the decrease in speech intelligibility as speaking conditions worsened was proportional to dysarthria severity [Kopf *et al*., ASHA 2013], and an acoustic analysis quantified the range of compensations made by talkers with PD [Shrivastav *et al*., ASA May 2014]. The current study modeled the intelligibility data from the synthetic environment study and applied the model to the data from the natural environment study to quantify the effects of talker compensation on intelligibility and to relate acoustic measures of speech to talker compensation.

4:40

2pAA11. Self-fitting hearing aids reveal strong individual preferences in noisy rooms. Peggy B. Nelson and Dianne VanTasell (Univ. of Minnesota, 164 Pillsbury Dr. Se, Minneapolis, MN 55455, peggynelson@umn.edu)

Noisy restaurants are among the most challenging listening situations for listeners with hearing loss, but little data are available to optimize hearing aids for noisy situations. In addition, evidence suggests that listeners are less satisfied with hearing devices in noisy listening environments. New technology from EarMachine(c) allows for individuals with hearing loss to self-adjust hearing aid amplification parameters. Self-adjustment technology can provide data to empirically answer questions about listener preferences in noise. Results from 30 listeners demonstrated self-selected gain profiles that reveal very large individual preferences that are not well predicted by age, gender, or hearing loss. These results indicate that there are strong preferences for listening in noisy rooms that are not yet well understood. [Funding provided by NIDCD R01 DC013267.]

Session 2pABa

Animal Bioacoustics: Tropical Bioacoustics II

Thomas F. Norris, Cochair
Bio-Waves, Inc., 517 Cornish Dr., Encinitas, CA 92024

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

Contributed Papers

1:00

2pABa1. The snapping shrimp conundrum: Spatial and temporal complexity of snapping sounds on coral reefs. Ashlee Lillis and T A. Mooney (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MRF 239, MS#50, Woods Hole, MA 02543, ashlee@whoi.edu)

Snapping shrimp are abundant crevice-dwelling crustaceans worldwide. The short-duration broadband "snap" generated by the collapse of a cavitation bubble upon the rapid closure of their specialized claw is among the loudest bioacoustic sound in the sea. Because these shrimp form large high density aggregations, their colonies create a pervasive crackling in many coastal environments, and variation in shrimp acoustic activity substantially alters ambient sound levels at a given location or time. Despite their fundamental contribution to the soundscape of coral reefs, and probable influence on sound-receptive organisms, relatively little is known about snapping shrimp sound production patterns, and the underlying behavioural ecology or environmental factors. Recent advances in recording capacity and efforts to sample habitat soundscapes at high spatiotemporal resolution have provided datasets that reveal complex dynamics in snapping shrimp sound production. Our analyses of soundscape data from coral reefs show that snap rates generally exhibit diurnal and crepuscular rhythms, but that these rhythms can vary over short spatial scales (e.g., opposite diurnal patterns between nearby reefs) and shift substantially over time (e.g., daytime versus nighttime dominance during different seasons). These patterns relate to abiotic variables such as temperature, light, and DO, as well as life history processes, but the nature of these relationships and underlying causal mechanisms are only beginning to be explored.

1:15

2pABa2. Long-term passive acoustic monitoring of Western Australian cockatoos. Shyam Madhusudhana and Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Phys. Bldg. 301 Rm. 136A, Kent St., Bentley, WA 6102, Australia, s.madhusudhana@postgrad.curtin.edu.au)

Carnaby's black cockatoos (*Calyptorhynchus latirostris*) and forest red-tailed black cockatoos (*Calyptorhynchus banksii naso*) are threatened avian fauna endemic to Southwestern Australia. Both species vocalize during various activities including flight, foraging, feeding, and breeding, and their vocalizations are known to differ by age and gender. In this study, passive acoustic monitoring is employed to assist with ongoing monitoring and recovery efforts. Autonomous acoustic recorders are installed at two locations—an urban setting in the Swan coastal plains and a bushland setting in the Perth hills district. The urban setting is a well-known roosting site for Carnaby's cockatoos. Noise from frequent aircraft activity is a dominant feature at the bushland setting. Call repertoires, established using visual sighting information, are used in developing automatic recognition algorithms for efficient processing of long-term recordings. In addition to providing information on population estimates, distribution, and demographics, the project aims to study the differences in call characteristics between the urban and bushland settings and establish a framework for observing potential seasonal changes in calling behavior.

1:30

2pABa3. Comparison of whistle classification results for three overlapping false whale populations in the Hawaiian Islands. Yvonne M. Barkley (Hawaii Inst. of Marine Biology, Univ. of Hawaii at Manoa, 1845 Wasp Blvd, Bldg. 176, Honolulu, HI 96818, ybarkley@hawaii.edu)

The false killer whale (*Pseudorca crassidens*) is a globally distributed species found in temperate and tropical waters. The Hawaiian Archipelago is home to three genetically distinct populations of *P. crassidens* with overlapping distributions: two insular populations, one within the main Hawaiian Islands (MHI) and the other within the Northwestern Hawaiian Islands (NWHI), and a broadly distributed pelagic population. The mechanisms that created and maintain the separation between these populations are unknown, but previous studies have shown that acoustic diversity may reflect the genetic differences. For this study, whistles from 14 MHI, 11 NWHI, and 7 pelagic groups were extracted and measured to assess whether differences exist between the vocal repertoires of each population. Whistle measurements were tested for statistical differences and then used in two different classification methods: a random forest algorithm and Kullback-Leibler (KL) divergence calculations. Preliminary results suggest that the MHI and pelagic populations are vocally distinct, with up to 100% correct classification of whistles for some groups. A comparison of the classification results from the random forest and KL divergence studies will be discussed. Understanding the acoustic differences between these populations may open up new acoustic monitoring approaches for a difficult-to-study species.

1:45

2pABa4. Time frequency analysis of the coconut rhinoceros beetle chirps. John S. Allen, Hans Ramm (Mech. Eng., Univ. of Hawaii-Manoa, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu), and Daniel Jenkins (Dept. of Molecular BioSci. and BioEng., Univ. of Hawaii-Manoa, Honolulu, Hawaii)

The Coconut Rhinoceros Beetle attacks the coconut palm trees in the Pacific Region and its foraging has led to the devastion of the palm trees in Guam. The beetle is an invasive species on the island of Oahu, Hawaii, since its latent arrival in 2013. Stridulation sounds produced by the beetles have been reported with respect to mating and aggressive male behavior. Previous studies have proposed the mechanism for the stridulation and reported on some preliminary acoustic recording of the chirp for groups of beetles. However, the chrip characteristics wth respect to different behavior have not been examined extensively and also the time frequency analysis of the chirp sounds specifically or male and female beetles has been limited. In laboratory setting, the chirp characteristics are investigated with respect individual beetles in terms of pulse length and chrip duration. Time frequency analysis is used quantifiy the frequency modulation. An Empirical

2p TUE. PM

Mode Decompositon (EMD) provides a means of determing the instantaneous frequency and phase. Sounds of wing beats during the beetle's take off and flight are also investigated.

2:00

2pABa5. Preliminary analysis of social calls used by tagged humpback whales in the Los Cabos, México, breeding ground. Kerri Seger (CCOM/JHC, Univ. of New Hampshire, 9331 Discovery Way, Apt. C, La Jolla, CA 92037, kseger@ucsd.edu), Ann M. Zoidis (Cetos Res. Organization, Bar Harbor, ME), Brooke L. Hawkins (Scripps Inst. of Oceanogr., San Diego, CA), M. Esther Jimenez-Lopez (Universidad Autónoma de Baja California Sur, La Paz, Mexico), Aaron M. Thode (Scripps Inst. of Oceanogr., La Jolla, CA), and Jorge Urban R. (Universidad Autónoma de Baja California Sur, La Paz, Mexico)

Humpback whales produce a large variety of diverse sounds beyond their well-known songs. Mothers, calves, and non-breeding whales may use these "social sounds" to maintain group cohesion, facilitate feeding, and/or increase a calf's safety. To date, social sounds have been studied off Hawaii, Alaska, and Australia. During the 2014-2015 breeding seasons, acoustic tag data were collected from humpback whales off Los Cabos, México. Twenty-one tags successfully recorded data (nine in 2014; 13 in 2015) from three mother/calf pairs, ten mother/calf/escort groups, and eight competitive pods varying in size from four to fourteen individuals. A subsequent manual analysis found 1587 social sounds in 2258 total minutes of data. Currently, there are 42 identified distinct social sounds used by humpback whales in the Los Cabos breeding ground, six of which seem unique to the Cabo region as compared to those published from Australia, Hawaii, and Alaska. Call type usage, call rates, and repertoire diversity (measured using information entropy) vary between mother/calf, mother/calf/escort, and competitive groups. Results suggest high variability in the types of social sounds used by different humpback whale groups within the same geographic area, and some social sounds overlap with repertoires known from other regions.

TUESDAY AFTERNOON, 29 NOVEMBER 2016 — CORAL 2, 3:00 P.M. TO 4:55 P.M.

Session 2pABb

Animal Bioacoustics and Signal Processing in Acoustics: Anthropogenic Transient Noise Sound Field and Its Effects on Animal Communication II

Shane Guan, Cochair
Office of Protected Resources, National Marine Fisheries Service, 1315 East-West Highway, SSMC-3, Suite 13700, Silver Spring, MD 20902

Satoko Kimura, Cochair
Field Science Education and Research Center, Kyoto Univ., Kyoto, Japan

Invited Papers

3:00

2pABb1. Time-domain modelling of underwater transient noise for environmental risk assessment. Adrian Farcas (Cefas, Cefas, Pakefield Rd., Lowestoft NR33 0HT, United Kingdom, adrian.farcas@cefas.co.uk), Claire F. Powell, Rebecca C. Faulkner (Cefas, Lowestoft, Suffolk, United Kingdom), Gordon D. Hastie (Univ. of St. Andrews, St. Andrews, United Kingdom), Paul M. Thompson (Univ. of Aberdeen, Cromarty, United Kingdom), and Nathan D. Merchant (Cefas, Lowestoft, Suffolk, United Kingdom)

Underwater transient noise has the potential to cause auditory damage in marine mammals, fish, and invertebrates. The risk of auditory damage is closely linked to the temporal structure of transient signals (e.g., rise time, peak pressure), which is modulated through propagation in the environment. Current models used in environmental risk assessment of underwater noise are based on pulse energy and do not predict the change in temporal structure as the transient propagates away from the source. To address this, the current work presents a time-domain model of transient propagation, based on Fourier synthesis of frequency domain solutions computed using a parabolic equation method. The model outputs are benchmarked first against standard analytical models, then validated against measurements of seismic survey and pile driving noise at various ranges from the source. The results will be discussed in the context of recent regulatory guidelines on the extent of transient acoustic fields and the associated risk of auditory damage to marine mammals.

3:20

2pABb2. Lost listening area assessment of anthropogenic sounds in the Chukchi Sea. David E. Hannay, Marie-Noel Matthews, Angela Schlesinger (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC V8Z 7X8, Canada, David.Hannay@jasco.com), Leila Hatch (Stellwagen Bank Natl. Marine Sanctuary, NOAA, Scituate, MA), and Jolie Harrison (Office of Protected Resources, Natl. Marine Fisheries Service, Silver Spring, MD)

The term listening area, refers to the region of ocean over which sources of sound can be detected by an animal at the center of the space. The lost listening area assessment method has been applied to in-air sounds for a noise effects assessment on birds but not, in our knowledge, previously to the assessment of underwater noise effects on marine mammals. The lost listening area method calculates a

fractional reduction in listening area due to the addition of anthropogenic noise to ambient noise. It does not provide absolute areas or volumes of space, as does the communication space method; however, a benefit of the lost listening area method is that it does not rely on source levels of the sounds of interest. Instead, the method depends only on the rate of sound transmission loss. We present a preliminary application of the method from an assessment of "cumulative and chronic effects" of noise produced by oil and gas exploration activities used in the National Marine Fisheries Service's Effects of Oil and Gas Exploration in the Arctic Ocean Final Programmatic Environmental Impact Statement.

Contributed Papers

3:40

2pABb3. Characterizing bounded transient sound fields in terms of potential effects on communication in aquatic animals. Mardi C. Hastings (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

Characterization of underwater transient sound fields in laboratory test chambers and shallow water field locations often used for bioacoustic studies is challenging. For example, no closed form solution exists for transient excitation of a water-filled waveguide with elastic walls. Key issues affecting interpretation of effects on animal communication in these situations are spatially dependent initial conditions that create subsequent sound propagation immediately following termination of the source signal; dissipative effects created by interaction of sound with boundaries; and possible excitation of modal resonances within the confined fluid. Therefore, evaluation and interpretation of data require time domain analysis of sound wave propagation and accurate knowledge of the locations of animal receivers within the sound field. Discrepancies in observed effects reported in the literature can often be explained by proper spatial evaluation of the transient sound field. Several examples will be presented along with results of an experimental evaluation of a water-filled steel waveguide excited by pulsed tones.

3:55

2pABb4. Long range sound propagation modeling for assessing seismic survey noise impacts on marine mammals. Yukino Hirai (Tokyo Univ. of Marine Sci. and Technol., 2-1-6, Etchujima, Koto-ku, Tokyo 135-8533, Japan, hirai.yukino@gmail.com), Toshio Tsuchiya (Japan Agency for Marine-Earth Sci. and Technology/Tokyo Univ. of Marine Sci. and Technol., Yokosuka, Kanagawa, Japan), Shinpei Gotoh, Etsuro Shimizu (Tokyo Univ. of Marine Sci. and Technology/Japan Agency for Marine-Earth Sci. and Technol., Koto-ku, Tokyo, Japan), and Koji Futa (Mitsubishi Precision Co., Ltd., Kamakura, Kanagawa, Japan)

Despite of small land area, Japan has vast EEZ, which is expected to have potential mineral resources. Extensive seismic surveys have been conducted for those targets. However, airguns used for the survey generate intense and low-frequency impulse sound, which could disturb or harm marine mammal behavior and auditory system. In June 2016, NOAA released "Ocean Noise Strategy Roadmap" that recommends modeling of sound propagation in the context of realistic environmental parameters to assess noise impacts on marine creatures. Japan has remarkable seasonal climate changes and complex ocean-bottom topography with trenches and seamounts; therefore, such model is worthwhile. Here, we propose a modeling method utilizing ARGO data includes seasonal climate changes and LEVITUS data, and ocean-bottom topographic configuration data from Google Earth. We calculated long range propagation of airgun sounds in different sound profiles and ocean-bottom topographic configurations around Ogasawara Islands, which is known as a cetacean habitat. Results showed that seasonal climate change and the precision of the ocean-bottom topographic configuration caused significant differences in sound propagation. These findings suggest that the realistic environmental parameters is essentially needed for the modeling of sound propagation and proposed method is a useful tool to assess noise impacts on marine mammals.

4:10

2pABb5. Long-term spatially distributed observations of deep diving marine mammals in the Northern Gulf of Mexico using passive acoustic monitoring. Natalia Sidorovskaia, Kun Li (Phys., UL Lafayette, UL BOX 44210, Lafayette, LA 70504-4210, nas@louisiana.edu), Christopher Tiemann (R2Sonic, Austin, TX), Azmy Ackleh, and Tingting Tang (Mathematics, UL Lafayette, Lafayette, LA)

This paper will present the results of processing long-term passive acoustic monitoring (PAM) data collected July through October 2015 in the Northern Gulf of Mexico in the vicinity of the Deep Water Horizon oil spill site to aid in understanding factors driving the distribution of sperm and beaked whales in the Gulf of Mexico. The Littoral Acoustic Demonstration Center -Gulf Ecological Monitoring and Modeling Consortium (LADC-GEMM) deployed five bottom-anchored acoustic moorings (LADC EARS buoys) at 10, 25, and 50 nmi distance from the 2010 oil spill location. Autonomous surface vehicles and a glider were simultaneously operated in the area for the additional collection of PAM data. The daily and monthly activity of three species of beaked whales exhibits spatial and seasonal variability, which appear to be correlated with levels of anthropogenic noise at the monitoring sites. Acoustic detection data are used to estimate abundances at three sites and compare them to the estimates obtained from baseline data collected before (2007) and right after (2010) the oil spill. Long-term abundance trends for both beaked and sperm whales are discussed. [Research supported by GOMRI.]

4:25

2pABb6. Evaluating underwater ambient noise levels across a changing Arctic environment. Melania Guerra, Kathleen M. Stafford, and Rex K. Andrew (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105, melania@apl.washington.edu)

As a result of climate change, the Arctic underwater acoustic environment is undergoing an unprecedented transformation of its ambient noise sources, including an expanded wind fetch over open water and an intensification of maritime transportation lanes across Northern routes. This comparative study synthesizes multi-annual passive acoustic monitoring (PAM) data collected at various Alaskan Arctic environments (Bering Strait, Chukchi Plateau, Chukchi, and Beaufort Seas) in order to evaluate the statistics of ambient noise across the changing region. Mean levels at significant frequency ranges are presented, as well as percentile PSD levels in 1-Hz frequency bins. Remote sensing sea ice data is utilized to separate each annual dataset into 3 different regimes: open water, covered, and transition (when ice is forming or breaking up). During each, the dominant large-scale noise sources are characterized and the relationship between noise and local wind speed is evaluated. Through manual inspection, ship transits and seismic surveys have been extracted, allowing ambient level associations on the scale of hours or days. Consistently across all years, the Chukchi Plateau site, which is the most distant from shore, presents the lowest overall ambient noise levels (20-40 dB lower than other locations), potentially indicating a new Arctic ambient noise baseline.

4:40

2pABb7. The effect of aircraft noise on bird song. Selvino R. de Kort and Andrew Wolfenden (Conservation, Evolution and Behaviour Res. Group, Div. of Biology and Conservation Ecology, Manchester Metropolitan Univ., All Saints Bldg., All Saints, Manchester M15 6BH, United Kingdom, S. Dekort@mmu.ac.uk)

In general, it has been found that birds respond to exposure of anthropogenic noise by singing at a higher frequency range. This is presumably the result of an attempt by the birds to reduce the masking effect of the low rumble of urban noise. In sharp contrast, we report that chiffchaffs reduce song frequency at airports where noise levels are intermittent rather than constant, but at a much higher amplitude level. In addition, we show experimentally that airport birds respond more aggressively to playback of airport songs than the control songs, while a nearby control population does not discriminate between airport and control songs. Potential explanations for these results will be discussed.

TUESDAY AFTERNOON, 29 NOVEMBER 2016 CORAL 4, 1:00 P.M. TO 4:00 P.M.

Session 2pAOa

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

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

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

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

Invited Papers

1:00

2pAOa1. Passive travel time tomography using ships as acoustic sources of opportunity. Bruce Cornuelle, William A. Kuperman, William S. Hodgkiss, Jeff Tippmann, Jit Sarkar, Chris Verlinden (UCSD-Scripps Inst. of Oceanogr., 9500 Gilman Dr., Dept 0230, La Jolla, CA 92093-0230, bcornuelle@ucsd.edu), and Karim Sabra (College of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Active ocean acoustic tomography transmits known source signals and often includes precise source and receiver positioning. Passive estimation of channel Greens functions has been widely demonstrated using both diffuse and concentrated noise sources, including the estimation of ocean sound speeds and currents. The uncertainty of surface ship positions determined by the Automatic Identification System (AIS) translates to travel time variability that is larger than expected from ocean structure at ranges of a few kilometers, but listening to the same source from spatially-distributed receivers adds more data while maintaining the same source position unknowns and may allow useful estimates of bottom depths from bottom-interacting rays and perhaps ocean structure. A fall 2016 experiment will test these ideas in the Santa Barbara Basin using 4 vertical line receiver arrays recording ship-radiated sound for 10 days. Estimates of the expected performance of the estimates of bottom depth and ocean sound speed have been a part of the design process. A general circulation model will be used to increase the efficiency of information use by parameterizing the four-dimensional ocean sound speed structure in terms of initial conditions, boundary conditions, and surface forcing. We present the design process and some proof-of-concept calculations from experimental observations.

1:20

2pAOa2. Recent progress in Taiwan on integrating moving vehicles for acoustic tomography applications. Chen-Fen Huang and Naokazu Taniguchi (Inst. of Oceanogr., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, chenfen@ntu.edu.tw)

Coastal seas around Taiwan exhibit a wide variety of oceanographic processes associated with the complex bottom topography, tides, winds (monsoons and typhoons), the Kuroshio, and the others. Conventional oceanographic measurements (e.g., acoustic Doppler current profiler) do not provide a synoptic image of the dynamic processes unless a large number of instruments are deployed. Ocean Acoustic Tomography (OAT) is an effective method for mapping the time-evolving spatial distribution of ocean current and temperature. As illustrated by Cornuell *et al.* (1989), the small spatial scale of ocean features is resolved by incorporating a ship-towed transceiver to the moored transceiver array, referred to as moving ship tomography (MST). The MST technique is applied to the ocean current reconstruction in coastal seas around Taiwan and is extended by using an autonomous underwater vehicle (AUV) as an alternative carrier for the transceiver. This talk presents a brief summary of the recent experiments and our upcoming project. Included are 1) a series of current-mapping experiments integrating moving vehicles in shallow-water environments and 2) simulation study of MST for experiment design on mapping the Kuroshio-induced current wakes of Green Island southeast of Taiwan.

1:40

2pAOa3. Ocean gliders as tools for acoustic tomography. Lora J. Van Uffelen (Ocean Eng., Univ. of Rhode Island, 15 Receiving Rd., Sheets Bldg., Narragansett, RI 02882, loravu@hawaii.edu), Sarah E. Webster, Craig M. Lee, Jason I. Gobat (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Ocean gliders are capable and increasingly ubiquitous platforms for collecting oceanographic data such as temperature and salinity measurements. These data alone are useful for ocean acoustic tomography applications, but gliders equipped with acoustic receivers can also record transmissions from acoustic sources, potentially providing additional paths for tomographic inversions. There are challenges associated with the use of gliders as tomography receivers, however, notably the uncertainty in underwater glider position, which can lead to ambiguity between sound-speed and glider position. Glider-based acoustic receptions from moored tomography sources can provide range measurements to aid in subsea glider localization. These data can be used in post-processing for precise glider positioning and in real-time for glider navigation. The current state of the art will be reviewed, and preliminary results will be presented from an experiment in the Arctic Ocean, which seeks to use moored acoustic tomography source signals for both subsea glider positioning and real-time navigation, including navigation under ice. [Work sponsored by ONR.]

Contributed Papers

2:00

2pAOa4. Moving ship tomography experiments in a coastal sea. Naokazu Taniguchi, Yun-Wen Li, and Chen-Fen Huang (Inst. of Oceanogr., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, naokazutaniguchi@gmail.com)

This study considers measuring differential travel times (DTT) between moored and ship-towed transceivers in shallow waters as a feasibility study to extend the concept of the moving ship tomography (Cornuelle *et al.*, 1989) to current field reconstructions. The delay-Doppler ambiguity function (AF) method is used to estimate and compensate the waveform distortions induced by the relative motion between transceivers. The Doppler shift (the relative speed between transceivers) is estimated by the peak of the AF, and the delay time series associated with the peak is the Doppler compensated arrival pattern. The DTTs are determined using the compensated arrival patterns in reciprocal directions. For the current estimation the estimated relative speed is used for removing the effect of the transceiver motion from the DTTs. A reciprocal sound transmission experiment was conducted using a ship-towed transceiver and two moored transceivers in the Sizihwan coastal near Kaohsiung, Taiwan. Due to the complicated arrival patterns observed, the DTTs were determined by tracking the peaks of the cross-correlation functions between the compensated reciprocal arrival patterns over the transmissions. The path-averaged currents derived from DTT agreed well with the direct measurement using a bottom-mounted acoustic Doppler current profiler.

2:15

2pAOa5. Autonomous underwater vehicle navigation using a single tomographic node. Sheng-Wei Huang (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 106, Taiwan, d99525008@ntu.edu.tw), Naokazu Taniguchi, Chen-Fen Huang (Inst. of Oceanogr., National Taiwan Univ., Taipei, Taiwan), and Jenhwa Guo (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan)

This study investigates AUV navigation using the one-way travel-time from an ocean tomographic sensor node. For the study of ocean acoustic tomography, several tomographic nodes are moored in the study area and broadcast *m*-sequence signals periodically. Each moored node consists of an acoustic transceiver, a GPS module, and a processing unit. The pulse-per-second signal from the GPS module is used to ensure the time synchronization of all the moored nodes. A tomographic sensor is installed on an AUV, serving as a moving node. To obtain highly accurate one-way travel-time between moored nodes and AUV, a chip scale atomic clock is operated on the AUV. Based on dead-reckoning and travel-time measurement, Extended Kalman Filter (EKF) is employed to improve the AUV localization. An experiment was conducted using an AUV and one moored node deployed in a shallow water environment southwest of Taiwan. The measurements from a Doppler velocity logger and a compass are used to calculate the current position of vehicle. The AUV was operated near the surface to obtain the GPS position as the ground truth for the performance evaluation. Preliminary results show the RMS position error between the EKF prediction and ground truth is about 80 m.

2:30

2pAOa6. Simulation study of moving ship tomography for mapping the current wakes of Green Island southeast of Taiwan. Kai-Fu Chang and Chen-Fen Huang (Inst. of Oceanogr., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, r04241107@ntu.edu.tw)

This study examines the design of acoustic arrays using moving ship tomography (MST) applied to the horizontal mapping of the Kuroshio-induced current wakes in the lee of Green Island southeast of Taiwan. The acoustic tomographic array consists of six transceivers; each transceiver can be either fixed at one position or towed by a ship. Numerical tomographic experiments using the synthetic ocean currents from the 8 km X 8 km study area are conducted for two configurations: five moored transceivers and one shipboard transceiver or four moored transceivers and two shipboard transceivers. Different spatial configurations of moored and shipboard transceivers are considered. For all configurations the ship moved around the periphery of the study area. Compared with the traditional tomographic arrays with only moored transceivers, the current reconstructions using MST show the reduction in residual error from 21% to 16%. Among all the MST arrays, the smallest residual error variance is obtained for the configuration of the five moored transceivers placed in a pentagon shape and one shipboard transceiver. This study will provide an optimum array design for the coming MST experiment near the Green Island.

2:45–3:00 Break

3:00

2pAOa7. Ocean remote sensing with acoustic daylight: Lessons from experiments in the Florida Straits. Michael G. Brown (RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, mbrown@rsmas.miami.edu), Oleg A. Godin (Dept of Phys., Naval Postgrad. School, Monterey, CA), Xiaoqin Zang (RSMAS, Univ. of Miami, Miami, FL), Nikolay A. Zabotin, and Liudmila Y. Zabotina (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado, Boulder, CO)

Ambient and shipping noise in the ocean provides acoustic illumination, which can be used, similarly to daylight in the atmosphere, to characterize the environment. Phase information, which is particularly sensitive to sound speed variations and current velocity, can be retrieved from noise observations by the process known as noise interferometry. Approximations to acoustic Green's functions, which describe sound propagation between two locations, are estimated by cross-correlating time series of diffuse noise measured at those locations. Noise-interferometry-based approximations to Green's functions can be used as the basis for a variety of inversion algorithms, thereby providing a purely passive alternative to active-source ocean acoustic remote sensing. This paper gives an overview of results from noise interferometry experiments conducted in the Florida Straits at 100 m depth in December 2012, and at 600 m depth in September/October 2013. Under good conditions for noise interferometry, estimates of cross-correlation functions are shown to allow one to perform advanced phase-coherent signal processing techniques to: perform waveform inversions; estimate currents by exploiting nonreciprocity; perform time-reversal/back-propagation calculations; and investigate modal dispersion using time-warping techniques. Conditions which are favorable for noise interferometry are identified and discussed. [Work supported by NSF and ONR.]

3:20

2pAOa8. Acoustic measurements of vertical velocity and temperature fluctuations of a hydrothermal plume with comparisons to large eddy simulations. Daniela Di Iorio (Dept. of Marine Sci., Univ. of Georgia, 250 Marine Sci. Bldg., Athens, GA 30602, daniela@uga.edu), J W. Lavelle (none, Seattle, WA), and Ian Adams (Dept. of Marine Sci., Univ. of Georgia, Athens, GA)

The acoustic scintillation method is used to study the hydrothermal plume of Dante within the Main Endeavour vent field (MEF) at the Endeavour segment of the Juan de Fuca Ridge. Forty days of vertical velocity and temperature fluctuations were obtained from the rising plume above the Dante edifice in an environment where the flow is dominated by strong (5cm/s) semi-diurnal tidal currents and a northerly mean residual current (3cm/s). These acoustic measurements provide a window on deep-sea hydrothermal plume dynamics in strong oscillatory cross flows. A large eddy simulation, parameterized with anisotropic mixing coefficients, taking into account ambient stratification and time-dependent background flows and calibrated by the acoustic measurements, yields insight into turbulent processes, entrainment, plume bending, rise height, and, inferentially, mound heat flux. The turbulent dissipation rates for kinetic energy (ε) and thermal variance (ε_θ) is approximated by computing the Reynolds averaged sub-grid scale turbulent production from shear and buoyancy ($\varepsilon = P-B$) and from temperature fluxes ($\varepsilon_\theta = P_\theta$), respectively, which are needed to compare to the acoustic derived turbulence levels. A new cabled observatory reciprocal acoustic scintillation system on the NEPTUNE observatory that will allow real time measurements of mean and turbulent flow of a hydrothermal plume will also be introduced.

3:40

2pAOa9. Applicability and feasibility studies of coastal acoustic tomography for long-term monitoring of the Indonesian throughflow transport variability. Fadli Syamsudin, Yudi Adityawarman, Reni Sulistyowati (Technol. for Regional Resource Development, Agency for the Assessment and Application of Technol. (BPPT), Jl. Salemba Bluntas C216, Jakarta, DKI Jakarta 10430, Indonesia, fadlihiro@yahoo.com), Bayu Sutedjo (Technologyfor Ocean Survey, Agency for the Assessment and Application of Technol. (BPPT), Jakarta, Indonesia), Arata Kaneko, and Noriaki Goda (Graduate School of Eng., Hiroshima Univ., Hiroshima, Japan)

We have been trying to monitor the Indonesian Throughflow (ITF), especially to understand how prominent tides and tidal mixing along the ITF main pathways in the Makassar and Lombok straits could transform the Pacific waters into Indian Oceans in the shorter scale and how the ITF variability in longer time scales affect the climate regimes over the region, using a modern and innovative technology of Coastal Acoustic Tomography (CAT). The first trial will be done in the Lombok sill where two 5-kHz CATs and three tide gauge sensors are put in the sill of depth range 200-300 m, lying between Lombok and Nusa Penida islands in the southern Lombok strait by the end of July 2016. The experimental design is a real-time land-cable CAT system, equipped with a 3G mobile antenna for telemetering CAT data to our Maritime Continent Center of Excellence (MCCOE) monitoring center in Jakarta every 10 minutes interval. Results are expected to verify CAT aplicability and feasibility as a cost effective oceanographic instrument to monitor the ITF transport variability along major straits in the Indonesian seas. This talk will also present the possibility of CAT to study mixing processes as important factors to improve model parametrizations.

Session 2pAOb

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

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

John A. Colosi, Cochair
Department of Oceanography, Naval Postgraduate School, 833 Dyer Road, Monterey, CA 93943

Chair's Introduction—4:15

Invited Paper

4:20

2pAOb1. Acoustic observations and characterization of oceanic methane gas bubbles rising from the seabed. Thomas C. Weber (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, tom.weber@unh.edu)

Methane, a potent greenhouse gas, can be found escaping the ocean seabed as gas bubbles in a wide range of geographic locations and water depths. As they rise toward the sea surface, methane gas bubbles undergo a complicated evolution. During ascent, methane gas is transferred into aqueous solution and other dissolved gases are transferred into the bubble. The gas transfer rate—a key factor in determining the ultimate fate of the methane—may be inhibited by hydrate formation in deep, cold water, or potentially by surfactants and adsorbed particulates at any depth. The presence of methane gas bubbles from both natural and anthropogenic sources is often identified using acoustic echo sounders. Beyond simple detection, acoustic techniques can be used to characterize methane bubbles in several ways. For example, narrow-band observations of seep target strength can be used with knowledge of bubble size distributions to estimate methane flux. Similar acoustic observations can be coupled with bubble-evolution models to constrain the fate of bubbles as they rise. Broadband techniques offer the potential to directly observe bubble size and rise speed, and consequently depth-dependent gas flux. The application of these and other techniques for detecting and characterizing methane gas-bubble seeps will be discussed.

Session 2pBAa

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

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

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

Chair's Introduction—1:00

Invited Papers

1:05

2pBAa1. Efficient dispersion analysis of guided waves in cortical bone. Pascal Laugier, Nicolas Bochud, Quentin Vallet, Xiran Cai, Quentin Grimal, and Jean-Gabriel Minonzio (Laboratoire d'Imagerie Biomedicale, Sorbonne Universités, UPMC Univ. Paris 06, CNRS UMR 7371, INSERM UMRS1146, 15 rue de l'ecole de medecine, Paris 75017, France, laugierp@gmail.fr)

Guided wave propagation is at the heart of the axial transmission techniques designed to assess bone health status. The method involves matching observed and predicted dispersion characteristics of guided waves. The strength of the technique is that it can infer more than one bone property from the measured ultrasonic data, such as cortical thickness, stiffness, or porosity. The suitability of the model chosen for the inversion has recently been debated and the question has been raised whether the physical model must take the soft tissue coating influence into account as well as perhaps other factors such as bone curvature. We present in this talk a series of experiments conducted on bone-mimicking phantoms (plates or tubes) with or without soft tissue-mimicking coating showing evidence 1/ that the experimental guided wave branches in the range of 0.4-1.6 MHz mainly exhibit sensitivity to the influence of the solid subsystem (bone) and 2/ that a simple non absorbing transverse isotropic free plate model provides an appropriate inverse model in all investigated cases, i.e., coated or non-coated plates and tubes. Finally, we demonstrate effectiveness of the inversion procedure in characterizing cortical bone using ex vivo and *in vivo* data.

1:25

2pBAa2. The role of acoustic microscopy in bone research. Kay Raum, Susanne Schrof, Johannes Schneider, Gianluca Iori, Vantte Kilappa, Juan Du, Matthias Pumberger, Michael Putzier (Charité Universitätsmedizin Berlin, Augustenburger Platz 1, Berlin 13353, Germany, kay.raum@charite.de), Jinming Zhou, and Hanna Isaksson (Lund Univ., Lund, Sweden)

Scanning acoustic microscopy (SAM) has been introduced 3 decades ago with the hope to open a new dimension in the microscopic analysis of biological tissues. However, only during the last decade this technology has emerged from a qualitative imaging modality to a quantitative measurement tool that provides fast and nondestructively elastic maps of acoustic and elastic properties with microscale resolution. Particularly, the spatial registration of parameter maps obtained by SAM with those obtained by complementary modalities, e.g., synchrotron microcomputed tomography, Raman spectroscopy, and inelastic micromechanical testing provided unprecedented inside into structure-composition-function relations, tissue changes with respect to adaptation, ageing, pathologies, and healing. Moreover, elastic maps generated by acoustic microscopy can serve as direct input for numerical simulations. This talk will review the key theoretical principles, experimental clues but also the limitations for the reconstruction of calibrated maps of elastic tissue properties and will present recent findings obtained by correlative imaging and SAM-based numerical sound propagation models. In particular, the role of matrix stiffness heterogeneity as a potential indicator for bone brittleness will be discussed.

1:45

2pBAa3. Recent advances in resonant ultrasound spectroscopy to measure bone stiffness tensor. Quentin Grimal, Xiran Cai, Laura Peralta, Kailiang Xu, Guillaume Marrelec (Biomedical Imaging Lab., Sorbonne Universités - Université Pierre et Marie Curie - CNRS - INSERM, 15 rue de l'école de médecine, Paris 75006, France, quentin.grimal@upmc.fr), Hassiba Daoui (Laboratoire de physique des matériaux, Université des Sci. et Technologie Houari Boumediéne d'Alger, Paris, France), and Pascal Laugier (Biomedical Imaging Lab., Sorbonne Universités - Université Pierre et Marie Curie - CNRS - INSERM, Paris, France)

Resonant Ultrasound Spectroscopy (RUS) is a method to measure the elasticity tensor of a material. RUS is particularly advantageous to measure small samples of anisotropic materials. In RUS, resonant frequencies of a sample are measured and computed frequencies of a numerical model of the sample are fitted, yielding the stiffness tensor. RUS was developed in the 1990s, but until recently, it was in practice limited to measure materials with a high quality factor. We have adapted the method to measure bone whose quality

factor is about 25. Our strategy combines Bayesian methods to retrieve overlapped resonant peaks in the RUS spectra and to solve the inverse problem using a limited number of resonant frequencies. The method allows a quasi-automated processing of RUS spectra where it is not necessary to know a priori the pairing between measured and computed frequencies. In the last years we have extensively used RUS to document the anisotropic elastic properties of human cortical bone and to investigate the determinants of elastic properties. We present in this talk 1) recent advances in signal processing for RUS; 2) a discussion on the precision of bone elasticity measurement; and 3) a new perspective regarding cortical bone elasticity determinants.

2:05

2pBAa4. Passive twin-layer spatial-temporal phase-interference compensator for improved transcranial ultrasound propagation. Christian M. Langton (Inst. of Health & Biomedical Innovation, Queensland Univ. of Technol., 60 Musk Ave., Brisbane, QLD 4049, Australia, christian.langton@qut.edu.au)

Transcranial ultrasound wave degradation created by variations in both thickness and tissue composition is a significant impediment to diagnostic and therapeutic interventions of the brain. The current "active" solution is to vary the transmission delay of ultrasound pulses, inherently necessitating electronic control of each individual transducer element. By applying the sonic-ray concept of ultrasound wave propagation, it was hypothesised that wave degradation can be significantly reduced if both the transit-time and propagation path-length for all sonic-rays are made constant. A computer simulation study was performed to investigate the implications of zero, partial and full spatial-temporal matching. Considering ultrasound propagation through a 20-step acrylic wedge test sample, only full spatial-temporal matching removed phase-interference. A "passive" physical ultrasound phase-interference compensator (UPIC), consisting of two layered materials of variable thickness, was developed and experimentally evaluated using acrylic step-wedge test samples exhibiting varying degrees of transit-time heterogeneity induced phase-interference. Time- and frequency-domain analysis demonstrated that incorporation of the UPIC successfully removed phase-interference in all cases. It is further hypothesised that the UPIC concept may be applied to both pulse-echo mode diagnostic imaging and transmission mode therapeutic interventions, incorporating either a single-element or a multi-element array transducer, of any practicable dimension.

2:25–2:40 Break

2:40

2pBAa5. Is there a role for approximations to the Kramers-Kronig relations in understanding the propagation of ultrasound in bone? James G. Miller (Phys., Washington U Saint Louis, Box 1105, 1 Brookings Dr., Saint Louis, MO 63130, james.g.miller@wustl.edu)

Clinical measurements of the speed of sound (SOS) usually exhibit moderately large site-to-site variations within a given individual, leading to rather wide ranges for osteoporosis, osteopenia, and normal. Consequently, the rather modest variation of phase velocity over 200 kHz to 800 kHz predicted by the approximate form of the Kramers-Kronig relations developed in our laboratory would appear to be of limited interest. However, whereas the Kramers-Kronig relations predict a systematic increase in phase velocity with frequency, laboratories in Kyoto, Paris, and Washington, D.C., each reported a systematic decrease in phase velocity with increasing frequency. We suggested that this "anomalous" (negative) decrease in velocity with frequency was actually the consequence of interference between two waves, each of which satisfied the Kramers-Kronig-predicted positive dispersion, with one wave sufficiently stronger than the other that the combination appeared to be only a single wave. Although separation of the fast and slow waves by Bayesian, Prony's, and other methods has provided substantial insight, significant unexplained results remain and are a focus of this presentation. The author wishes to acknowledge significant contributions by former colleagues: C. Anderson, A. Bauer, G. Brandenburger, L. Bretthorst, A. Groopman, J. Hoffman, S. Handley, M. Holland, M. Hughes, E. Jaynes, J. Katz, K. Marutyan, J. Mobley, R. Norberg, M. O'Donnell, R. Trousil, and K. Waters.

3:00

2pBAa6. Signal processing methods for through-transmission measurements of fast and slow waves in bovine and equine cancellous bone. Keith A. Wear (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg. 62, Rm. 2104, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov), Amber Groopman, Jonathan Katz (Washington Univ., St. Louis, MO), Mark Holland (Indiana Univ., Indianapolis, IN), Yoshiki Nagatani (Kobe City College of Technol., Kobe, Japan), Katsunori Mizuno (Univ. of Tokyo, Tokyo, Japan), Mami Matsukawa (Doshisha Univ., Kyoto, Japan), and James G. Miller (Washington Univ., St. Louis, MO)

Through-transmission measurements in cancellous bone *in vitro* often reveal two longitudinal waves. The velocities and amplitudes of fast and slow waves are related to bone microstructure and composition and may provide useful diagnostic information. Phase velocity and attenuation for fast and slow waves were measured as a function of propagation depth in bovine and equine cancellous bone. Ultrasound measurements were performed as the thicknesses of bone samples were systematically reduced from 15 to 6 mm (bovine) or 12 to 4 mm (equine) (Nagatani *et al.*, *Ultrasonics*, **48**, 607-612, 2008). Fast and slow wave properties were estimated using a Bayesian method (Marutyan *et al.*, *J. Acoust. Soc. Am.*, **121**, EL8-EL15, 2007) and the modified least-squares Prony's method with curve fitting (Wear, *J. Acoust. Soc. Am.*,**133**, 2490-2501, 2013). The two methods, although based on different sets of assumptions, showed excellent agreement. This work provides strong support for both algorithms and for the two-wave model for cancellous bone upon which both are based (Marutyan *et al.*, *J. Acoust. Soc. Am.*, **120**, EL55-EL61, 2006).

3:20

2pBAa7. Phase cancellation effect on broadband ultrasonic attenuation analysis for human trabecular bone assessment using a 2-D synthetic array. Yi-Xian Qin and Jiqi Cheng (Biomedical Eng., Stony Brook Univ., BioEng. Bldg., Rm. 215, Stony Brook, NY 11794, yi-xian.qin@stonybrook.edu)

Quantitative ultrasound has been developed to evaluate trabecular BMD and structural integrity. The ultrasound parameters, i.e., normalized broadband attenuation (nBUA), have been widely used for bone health status. However, the reproducibility and accuracy of nBUA are influenced by the phenomena of phase cancellation. The objectives of this study have two folds, 1) to accurately investigate the effects of phase cancellation on BUA calculation using an ultra-small receiver (aperture size: 0.2 mm) in a newly developed 2-D synthetic array-system, and 2) to evaluate the effects of phase cancellation on human trabecular bones. Most energy was located within a circle with a diameter of 12.70 mm corresponding to the transmitter aperture size. The tests of ultrasound BUA and micro-CT on the trabecular bone cylinders with a diameter of 25 mm. Both phase sensitive (PS) detection and phase insensitive (PI) detections were performed. The data indicated that the average nBUA is 24.8±9.5 dB/MHz/cm and 19.2±5.5 dB/MHz/cm for PS and PI respectively. PS nBUA is 28.5% higher than PI BUA, which PS-nBUA can explain 81.2% of the variability in PI nBUA. Both PI-nBUA and PS-nBUA are highly correlated with BV/TV ($R=0.911$ and $R=0.898$, $p<0.0001$), and the Young's modulus ($R=0.9061$ and $R=0.822$ $p<0.0001$), respectively.

3:40

2pBAa8. Structure-function relationship in bone: Anisotropic mechanical properties of trabecular bone determined using poroelastic ultrasound. Luis Cardoso (Biomedical Eng., City College of New York, 160 Convent Ave., Steinman Hall, ST-401, New York, NY 10031, cardoso@ccny.cuny.edu)

In the ageing skeleton, trabecular bone adapts its porosity, microarchitecture, and tissue composition in response to its mechanical environment through a complex and well-orchestrated bone remodeling process. Such adaptation process allows bone to be mechanically competent to resist low-trauma fractures. However, the relationship between structure and function in bone is not yet fully known. To take into account the effect of anisotropic architecture on the mechanical properties of porous media, the fabric tensor **F** was introduced in bone Biomechanics by Cowin. The fabric tensor **F** is a quantitative stereological measure of the degree of structural anisotropy and the mechanical principal orientations of a porous medium. We recently incorporated the fabric tensor into the theory of wave propagation in a fluid saturated porous media, and showed that the fabric tensor is a good predictor of directional-dependent measurements of ultrasound in cancellous bone. In our current study, we demonstrate that the spatial distribution of mass and pore space in trabecular bone (i.e., fabric), when combined with its volume fraction and tissue mineral density, is able to well describe the directional-dependent variability of the anisotropic elastic and yield behavior of bone, which cannot predicted by bone mineral density alone.

4:00

2pBAa9. Attenuation and dispersion of ultrasound in cancellous bone. Theory and experiment. 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)

The paper presents theoretical and experimental issues related to modeling of elastic wave propagation in cancellous bone. The Biot's theory, commonly used for modeling of ultrasonic wave propagation in cancellous bone, is discussed in context of its potential applicability for theoretical prediction of wave parameters. Particular attention is focused on the analysis of physical mechanisms of ultrasonic wave propagation in cancellous bone that govern phase velocity and attenuation coefficient as function of frequency and porosity. The analysis of the model is focused on the absorption and scattering mechanisms responsible for attenuation of ultrasonic waves in cancellous bone, which based on the ultrasonic experiments presumably play a predominant role in the total attenuation. The suitability of the model is discussed and verified by comparison of results of sensitivity analysis of the model with experimental ultrasonic data for bone mimicking phantoms and human cancellous bones.

4:20

2pBAa10. Ultrasound multiple scattering for the assessment of bone micro-architecture. Mason Gardner (MAE, NCSU, Raleigh, NC), Sylvain Haupert, Guillaume Renaud (Laboratoire d'Imagerie Biomedicale, Paris, France), and Marie M. Muller (MAE, NCSU, 911 oval Dr., Raleigh, NC 27695, mmuller2@ncsu.edu)

Retrieving bone micro-architectural parameters non-invasively using ultrasound would enable the monitoring of bone loss, and the follow up of the efficacy of treatments. When ultrasound propagate in bone in the MHz range, multiple scattering is non-negligible, and we demonstrate that it is possible to take advantage of it. We propose methods based on ultrasound multiple scattering for the quantification of micro-architectural parameters in cortical and trabecular bone. Using linear transducer arrays at 5 and 8 MHz, we present both finite differences simulations and experimental results in 3D printed phantoms of trabecular bone, in equine trabecular bone and in human cortical and trabecular bone. We show that the framework of the Independent Scattering Approximation can be used to model the attenuation in both cortical and trabecular bone. We demonstrate that measuring the diffusion constant is feasible in cortical and trabecular bone, and that it is correlated to micro-architectural parameters. Preliminary results indicate a significant correlation between the trabecular anisotropy and the anisotropy of the diffusion constant ($r=0.87$, $p=7.6e\text{-}8$). In cortical bone, a significant quadratic relationship was observed between the mean pore size and the diffusion constant. The cortical pore density was highly and significantly correlated to the diffusion constant ($r=0.98$, $p=0.001$).

4:40

2pBAa11. Ultrasonic backscatter difference measurements of cancellous bone: Relationships with microstructure and bone mineral density. Brent K. Hoffmeister, P. L. Spinolo, Matthew T. Huber, Joshua T. Moore, Ann M. Viano (Dept. of Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, hoffmeister@rhodes.edu), and Jinsong Huang (College of Medicine, The Univ. of Tennessee, Memphis, TN)

Background: Backscatter difference techniques are being developed to detect changes in bone caused by osteoporosis. Backscatter difference techniques compare the power in one portion of an ultrasonic backscatter signal to the power in a different portion of the same signal. Goal: Investigate how backscatter difference measurements depend on the density and microstructural characteristics of cancellous bone. Procedure: Ultrasonic backscatter signals were acquired from 30 specimens of bone using a 1 and 5 MHz broadband transducer. The normalized mean backscatter difference (nMBD) was determined by computing the power difference (in dB) between two gated portions of the backscatter signal and dividing by the center to center time separation between gates. Microstructural characteristics of the specimens and bone mineral density (BMD) were determined using high resolution x-ray micro-computed tomography. Results: nMBD demonstrated moderate to strong linear correlations with microstructure and BMD ($0.50 \leq |R| \leq 0.83$). The measured correlations did not depend strongly on transducer frequency. Conclusions: The backscatter difference parameter nMBD may be sensitive to changes in microstructure and density caused by osteoporosis. [Work supported by NIH/NIAMS R15AR066900.]

5:00

2pBAa12. Fabrication of bone phantoms using 3D printers. Masahiro Ohno (Faculty of Eng., Chiba Inst. of Technol., Tsudanuma 2-17-1, Narashino, Chiba 275-0016, Japan, ohno.masahiro@p.chibakoudai.jp)

Requests for making physical models of human body parts, or phantoms, are increasing in accordance with the improvement of medical instruments. Advance in 3D technologies such as 3DCAD and 3D printers has facilitated the fabrication of more precise phantoms. In this paper, we report on our trial to make phantoms of bones that can be used in developing ultrasonic methods of osteoporosis diagnosis. We did not aim to make a complete model that mimics some certain diseased bones, but rather aimed to develop models the parameters of which, such as trabecular thickness, pore size, and its orientation, can be set as freely as possible. Multi-sliced X-ray CT images of bovine bones, those of artificial polymer foam, and purely numerically generated images were used as raw data. These images were converted to 3D data using commercial volume data generation software together with some 3DCAD and intermediate software packages, and finally printed by 3D printers. The bone-volume/total-volume ratio (BV/TV) was controlled by altering the thresholding value in 3D data construction, as well as by air-blast-polishing printed models. Results of ultrasonic transmission experiments are also shown.

TUESDAY AFTERNOON, 29 NOVEMBER 2016 CORAL 1, 1:00 P.M. TO 5:25 P.M.

Session 2pBAb

Biomedical Acoustics and Physical Acoustics: Cavitation in Therapeutic Ultrasound IV: Tissue Fractionation

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

Contributed Papers

1:00

2pBAb1. Erosion of soft tissue by focused ultrasound-induced streaming. Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, 1013 NE 40th St., Seattle, WA 98105, amax38@u.washington.edu), Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Michael R. Bailey, Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Tanya D. Khokhlova (Dept. of Gastroenterology, Univ. of Washington Medical Ctr., Seattle, WA), George R. Schade (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), and Vera A. Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Mechanical erosion of soft tissues into subcellular debris has been demonstrated with pulsed high-intensity focused ultrasound, facilitated by either boiling or cavitation bubbles. In this work, we propose acoustic streaming as a primary cause of tissue erosion at a tissue-fluid interface. Bovine liver tissue and polyacrylamide gels were sonicated in a degassed water bath, with the focus positioned at the tissue/fluid interface. Pulses with duration between 1 to 10^4 μs and constant duty cycle of 0.5% were applied from a 1 MHz transducer generating focal pressures $|p_-| \leq 17$ MPa and $p_+ \leq 90$ MPa. Results showed a strongly nonlinear change in erosion with pulse duration, being greatest for pulse lengths between 50-500 μs. For longer pulses (>1 ms), high-speed videos showed streaming velocities >10 m/s. Moreover, lesions >1cm in depth were produced in tissue phantoms even with a single pulse and tissue disruption was evident where no bubbles were observed at the tissue-fluid interface. Enhancement of erosion was observed with the presence of bubbles not directly adjacent to tissue but along the beam path, possibly due to bubble-enhanced streaming. [Work supported by NIH R01

EB007643, T32 DK007779, K01 DK104854, P01 DK043881, and NSBRI through NASA NCC 9-58.]

1:15

2pBAb2. A multimodal evaluation of boiling histotripsy lesion properties in *ex vivo* bovine liver. Yak-Nam Wang (APL, CIMU, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ynwang@u.washington.edu), Tanya D. Khokhlova (Gastroenterology, Univ. of Washington, Seattle, WA), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Wayne Kreider (APL, CIMU, Univ. of Washington, Seattle, WA), Ari Partanen (Philips Healthcare, Washington DC, MD), Navid Farr (BioEng., Univ. of Washington, Seattle, WA), George R. Schade (Urology, Univ. of Washington, Seattle, WA), Valeriy P. Chernikov (Res. Instotue of Human Morphology, Moscow, Russian Federation), Sergey V. Buravkov (Ecological and Extreme Medicine, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Michael R. Bailey (APL, CIMU, Univ. of Washington, Seattle, WA), and Vera A. Khokhlova (Dept. of Acoust., M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation)

New types of high intensity focused ultrasound (HIFU) therapy aiming at mechanical homogenization of tissue has shown great promise, namely, cavitation-cloud histotripsy and boiling histotripsy (BH). BH uses millisecond-long bursts of HIFU waves containing shocks to repeatedly induce boiling at the focus; the interaction of incident HIFU waves with vapor bubbles homogenizes tissue. In this study, degassed *ex vivo* bovine liver samples were sonicated using a 256-element 1.2 MHz array of a clinical MR-HIFU system. The BH lesions were produced using 10-ms long pulses with 80 MPa shocks in situ and pulse repetition frequencies (PRFs) of 1-10 Hz to cover a range of effects from pure mechanical homogenization to thermal ablation. Individual lesions were generated for the multimodal analysis of the lesion including ultrastructure (electron microscopy), molecular (biochemistry), and microstructure (histological) methods. The extent of homogenization and thermal denaturation was evaluated for each lesion. The results of this work showed that the degree of mechanical tissue disruption and the amount of heat generated in large BH lesions can be tailored to result in a range of desired tissue effects dependent on the target clinical application. [Work supported by NIH EB007643, RFBR 16-02-00653, K01 EB015745, and NSBRI through NASA NCC 9-58.]

1:30

2pBAb3. Designing a multi-element array transducer for abdominal boiling histotripsy applications. Pavel Rosnitskiy, Petr Yuldashev (M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Adam Maxwell, Wayne Kreider, Michael Bailey (Univ. of Washigton, Seattle, WA), Oleg Sapozhnikov (M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), and Vera Khokhlova (Univ. of Washigton, 1013 NE 40th St., Seattle, WA 98105, va.khokhlova@gmail.com)

The boiling histotripsy (BH) method to mechanically fractionate tissue using high intensity focused ultrasound relies on the presence of high amplitude shocks. Recently, a method of determining single-element transducer parameters to achieve desired shock amplitudes at the focus was developed. It was shown that the transducer F-number is the main parameter that determines the focal pressure level at which fully developed shocks form. Here, a 256-element HIFU array transducer of 1.5 MHz frequency was designed for abdominal BH applications using the proposed method. An opening was included at the center of the transducer to fit an ultrasound imaging probe and a compact 16-arm spiral layout of uniformly sized elements was chosen for maximizing the transducer power output. The F-number of the array and the diameter of its elements were determined to satisfy technical limitations on the intensity level at the array elements as well as the required shock amplitudes of 90-100 MPa at the focus assuming 9 dB attenuation losses in tissue. For creating volumetric lesions, the region of safe and efficient steering of the array focus was evaluated using the T-Array software package (www.limu.msu.ru). [This work was supported by RSF 14-12-00974 and NIH R01 EB7643.]

1:45

2pBAb4. Characterization of high-amplitude fields of an annular array using acoustic holograms of its individual elements. Petr V. Yuldashev (Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, petr@acs366.phys.msu.ru), Martijn Hoogenboom (Dept. of Radiology, RadboudUMC, Nijmegen, Netherlands), Erik Dumont (Image Guided Therapy, Pessac, France), Martijn H. den Brok (Dept. of Tumor Immunology, RadboudUMC, Nijmegen, Netherlands), Pavel B. Rosnitskiy, Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Jurgen J. Fütterer (Dept. of Radiology, RadboudUMC, Nijmegen, Netherlands), Gosse J. Adema (Dept. of Tumor Immunology, RadboudUMC, Nijmegen, Netherlands), and Vera A. Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

High-intensity focused ultrasound system (Image Guided Therapy, Pessac France) with magnetic resonance guidance was developed and used for evaluation of thermal and mechanical ablation methods in mouse tumors. The system comprises a 3 MHz annular array transducer (48 mm diameter and 35 mm radius of curvature) consisting of 16 elements that allow moving the focus electronically along the transducer axis. Experimental characterization of nonlinear acoustic field generated by such high-frequency strongly focused transducer is technically challenging because of the fine spatial structure of the field, small dimensions of the focal lobe, and limited hydrophone bandwidth. Here we evaluated pressure levels and shock-forming conditions for the system using a combination of numerical modeling based on the 3D Westervelt equation with holographic boundary condition obtained from low-power measurements. The holograms were measured separately for each element and combined together to obtain a boundary condition for the array with all operating elements. Then, nonlinear field simulations were performed at increasing power output levels. It was shown that the transducer is capable to produce focal waveforms with 120 MPa shock amplitude at 110 W acoustic power and thus is well suited for shock-based ablation therapies in small animals. [Work supported by Radboudumc Ph.D. grant and RSF-14-12-00974.]

Invited Papers

2:00

2pBAb5. Pathology and immune effects of magnetic resonance imaging-guided Boiling Histotripsy in murine tumor models. Gosse J. Adema (Tumor Immunology, RadboudUMC, RIMLS, Geert Grooteplein 26/28, Nijmegen 6525 GA, Netherlands, gosse.adema@radboudumc.nl), Martijn Hoogenboom (Dept. of Radiology, RadboudUMC, RIMLS, Nijmegen, Netherlands), Renske van den Bijgaart, Dylan Eikelenboom (Tumor Immunology, RadboudUMC, RIMLS, Nijmegen, Netherlands), Pieter Wesseling (Dept Pathol., radboudUMC, RIMLS, Nijmegen, Netherlands), Arend Heerschap (Dept. of Radiology, RadboudUMC, RIMLS, Nijmegen, Netherlands), Martijn den Brok (Tumor Immunology, RadboudUMC, RIMLS, Nijmegen, Netherlands), and Jurgen Futterer (Dept. of Radiology, RadboudUMC, RIMLS, Nijmegen, Netherlands)

In situ tumor ablation techniques are successfully applied for the destruction of local, often inoperable tumor masses. Following ablation tumor antigens become instantly available for immune cells, but systemic abscopal effects have only occasionally been reported after ablation monotherapy. Which ablation technique combines optimal local destruction with effective antigen release for induction of anti-tumor immunity is largely unknown. We study non-invasive *MRI-guided high intensity focused ultrasound*-ablation (MRgHiFU) in

murine tumor models for local destruction by heating or mechanical disruption using Boiling histotripsy (BH). BH mechanically fragmentizes soft tissue into submicron fragments that are absorbed as part of a physiological healing response. BH treatment was performed using a MR compatible animal HIFU system (Image Guided Therapy, Pessac, France) with a 3 MHz transducer (focal spot size 0.5 x 0.5 x 2.0 mm). A 7T animal MR scanner was used for treatment guidance and evaluation. Here, we will present the pathological response and efficacy of BH treatment in three mouse models with different tumor characteristics: a soft-tissue melanoma (B16OVA), a compact growing thymoma (EL4), and a highly vascularized neuroblastoma (9464D) tumor. Furthermore, the impact of the type of ablation on immune cell infiltration and immune cell activation will be discussed.

2:20

2pBAb6. Healing and the immune response following boiling histotripsy ablation of renal carcinoma in the Eker rat. George R. Schade, Yak-Nam Wang, Kayla Gravelle, Stella Whang, Venu Pillarisetty, Joo Ha Hwang, W. Conrad Liles, Vera Khokhlova, Michael R. Bailey, and Tatiana D. Khokhlova (Dept. of Urology, Univ. of Washington, 5555 14th Ave. NW, Apt. 342, Seattle, WA 98107, grschade@uw.edu)

Boiling histotripsy (BH) is an experimental non-invasive focused ultrasound (FUS) technology that uses milliseconds-long ultrasound pulses at low duty cycle to mechanically homogenize targeted tissue. Here, we report the evolution of BH lesions and the resulting immune response to *in vivo*ablation of renal carcinoma (RCC) in a rat model. RCC bearing Eker rats and syngeneic wild-type rats were randomly assigned to transcutaneous BH or FUS SHAM procedure targeting ~0.5 cc of RCC or non-tumor bearing normal kidney. BH was delivered with a 1.5 MHz US-guided small animal FUS system (VIFU-2000, Alpinion) operated at duty cycles of 1-2%, 10-20 ms pulses, and 525-600 W electric power. Rats were survived for up to 56 days post-treatment. BH lesions evolved from sharply demarcated regions of homogenized tissue to small fibrous scars by 56 days. Compared to sham procedure, BH produced significant alterations in plasma and intrarenal cytokines and tumor/renal infiltrating leukocyte populations. These data describe the immunologic changes and time course of healing following BH ablation *in vivo*. Future studies will further address tumor control and the impact on metastases. [The Focused Ultrasound Foundation, The Urology Care Foundation, and NIH K01EB015745 and R01CA154451.]

2:40–2:55 Break

2:55

2pBAb7. Bubble-seeded histotripsy for cancer treatment. Ken-ichi Kawabata (Res. & Development Group, Hitachi, Ltd., 1-280 Higashi-Koigakubo, Kokubunji, Tokyo 185-8601, Japan, kenichi.kawabata.ap@hitachi.com), Takahi Maruoka, Rei Asami, Hideki Yoshikawa (Res. & Development Group, Hitachi, Ltd., Tokyo, Japan), and Reiko Ashida (Osaka Medical Ctr. for Cancer and Cardiovascular Diseases, Osaka, Japan)

An approach to applying the histotripsy-like mechanical effects of ultrasound for cancer treatments is studied. We investigated the combinational effects of pulsed ultrasound at intensities used for HIFU (high intensity focused ultrasound) therapy and a type of locally injected superheated perfluorocarbon droplet (PCND). The droplets produce microbubbles when exposed to ultrasound. In our approach, chemical agents are also added to obtain combined mechanical and chemical antitumor effects. It was found that, in the presence of locally injected PCND, spatially controlled mechanical effects similar to histotripsy can be achieved with pulsed ultrasound (pHIFU) at several kW/cm^2*ex vivo*. Further, it was suggested that even if the target volume is too vast to distribute PCND with a single injection, one injection is enough because PCND particles migrate inside tissues due to pHIFU. Further, the antitumor effects of pHIFU in combination with the local injection of PCND and an antitumor agent (Adriamycin) were investigated *in vivo* using murine tumors (Colon 26). Experiments were performed under acoustic conditions and with drug doses not severe enough to induce significant antitumor effects. It was found that PCND and Adriamycin were effective at suppressing tumor growth and tumor regrowth after suppression, respectively. The obtained results are very promising for developing novel cancer treatments.

Contributed Papers

3:15

2pBAb8. A method for phase aberration correction of high intensity focused ultrasound fields using acoustic nonlinearity. Tatiana Khokhlova, Adam Maxwell, Wayne Kreider, Vera Khokhlova, Matthew O'Donnell, and Oleg Sapozhnikov (Univ. of Washington, 325 9th Ave., Harborview Medical Ctr., Box 359634, Seattle, WA 98104, tdk7@uw.edu)

High intensity focused ultrasound (HIFU) therapies are often affected by aberrations induced by soft tissue heterogeneities. To mitigate these aberrations, phase corrections at different elements of a HIFU transducer array can be introduced, if suitable feedback is available. Existing feedback approaches include maximization of focal tissue displacement induced by acoustic radiation force or scattering from a cavitation bubble nucleated at the focus. Here we propose an aberration correction method based on backscattering of strongly nonlinear HIFU waves that are present only at the focus. A 1.5 MHz, 12-element sector array was integrated with a coaxially aligned imaging probe (P7-4). Each element of the array emitted a short, high-amplitude pulse through the aberrative layer toward the focus in tissue, and the imaging probe recorded the backscattered signal at higher frequencies. The focusing gain of each sector element was sufficient to generate a highly nonlinear focal waveform, which allowed localizing the "beacon" for phase correction feedback. The relative transmit time delays for array elements were calculated based on cross-correlations between the nonlinear backscattered signals corresponding to the other elements. The results of hydrophone measurements and *ex vivo* tissue experiments demonstrate the feasibility of the proposed approach. [Work was supported by NIH K01EB015745 and R01EB7643.]

3:30

2pBAb9. Histotripsy pulse-reflection for 3D image forming and bubble cloud localization in transcranial applications. Jonathan R. Sukovich, Zhen Xu, Timothy L. Hall, Jonathan J. Macoskey, and Charles A. Cain (Biomedical Eng., Univ. of Michigan, 1410 Traver Rd., Ann Arbor, MI 48105, jsukes@umich.edu)

Here, we present results from experiments using histotripsy pulses scattered off the surface of the skull, as well as bubble clouds generated within, to reconstruct 3D images of the exterior skull surface and localize bubbles within. These capabilities have the potential to provide image coregistration and real-time ultrasound monitoring for transcranial histotripsy treatment, without the need for MRI guidance. Histotripsy pulses were delivered to an

2p TUE. PM

ex vivo human skullcap mounted centrally within a 500 kHz, 256-element histotripsy transducer with transmit-receive capable elements. Straight-line ray tracing approximations based on the times-of-flight of the emitted pulses and the known geometry of the array were used to calculate points on the skull surface and to localize bubble clouds generated within. Using these methods, we were able to accurately locate and orient the skull within the array and generate a 3D map of its surface for coregistration with an a priori 3D scan. The points calculated based on the pulse-reflection from the skull were found to be within 1.25 mm of the surface measured in the a priori 3D scan. Using these signals, the calculated centroid of the generated bubble clouds was likewise found to be within 500 um of the measured focal point of the array through the skull.

Invited Papers

3:45

2pBAb10. Monitoring and guidance of histotripsy using ultrafast imaging. Bastien Arnal (Institut Langevin, ESPCI Paris, INSERM U979, CNRS UMR7587, 17 rue Moreau, Paris 75012, France, bastien.arnal@gmail.com), Wei-Ning Lee (Institut Langevin, ESPCI Paris, INSERM U979, CNRS UMR7587, Hong Kong, Hong Kong), Mathieu Pernot, and Mickael Tanter (Institut Langevin, ESPCI Paris, INSERM U979, CNRS UMR7587, Paris, France)

To use histotripsy safely, it is required to visualize the cavitation clouds and the lesion formation in real-time. Conventional ultrasound imaging (USI) is a good candidate for the real-time monitoring of the procedures. However, bubble clouds are not always clearly visible *in vivo* using USI because of flows, motion or high echogenicity of the target region. Ultrafast imaging, based on plane wave or diverging waves sonications, is able to record a larger amount of data, providing broad information about how signals originating from different sources vary in time. Here, we apply a spatiotemporal singular value decomposition of ultrafast acquisitions that enables the discrimination of cavitation bubbles from moving tissue *in vivo*. An automation to determine the filtering parameters will be introduced. Moreover, USI depicts the histotripsy lesion only at an advanced stage and can be hindered by the presence of dissolving gas in the tissue. As the tissue microstructure is greatly affected by the treatment, a high stiffness contrast is observed. We will show that quantitative stiffness evaluation allows measuring tissue damage from the beginning of the treatment with a 2 to 6-fold contrast increase compared to USI.

4:05

2pBAb11. Comparison of passive cavitation imaging and plane wave B-mode imaging for monitoring histotripsy ablation. Kenneth B. Bader, Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3935, Cincinnati, OH 45267-0586, kenneth.bryan.bader@gmail.com), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), and Christy K. Holland (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Histotripsy utilizes the oscillations of bubbles within a cloud for transcutaneous mechanical ablation of tissue and is currently under development to treat benign prostatic hyperplasia. The bubble clouds are hyperechoic, enabling standard B-mode imaging to be used for image guidance. The volumetric oscillations of the bubbles also generate acoustic emissions that can be imaged. The purpose of this study was to demonstrate the feasibility of monitoring histotripsy-induced ablation with passive cavitation images. Prostate tissue phantoms were exposed to mechanically ablative, 1-MHz histotripsy pulses over a range of pulse durations (5—20 ms) and peak negative pressures (12—23 MPa). Bubble clouds generated by the histotripsy pulses were monitored with a linear array using passive cavitation imaging (PCI) and plane wave B-mode imaging. The utility of PCI and B-mode imaging to predict the phantom ablation zone was assessed using receiver operating characteristic (ROC) curve analysis. The area under the ROC curve, accuracy, and sensitivity were greater for PCI relative to B-mode imaging ($p < 0.05$). These results will be discussed in the context of the potential advantages of PCI relative to B-mode imaging for the prediction of histotripsy-induced ablation.

Contributed Papers

4:25

2pBAb12. Effects of temperature on the histotripsy intrinsic threshold for cavitation. Eli Vlaisavljevich, Zhen Xu (Univ. of Michigan, 1111 Nielsen Ct. Apt. 1, Ann Arbor, MI 48105, evlaisav@umich.edu), Adam Maxwell (Univ. of Washington, Seattle, WA), Lauren Mancia, Xi Zhang, Kuang-Wei Lin, Alexander Duryea, Jonathan Sukovich, Tim Hall, Eric Johnsen, and Charles Cain (Univ. of Michigan, Ann Arbor, MI)

A histotripsy cavitation cloud can be formed by a single acoustic pulse with one high amplitude negative cycle, when the negative pressure exceeds a threshold intrinsic to the medium. The intrinsic threshold in water-based soft tissues, which is similar to the intrinsic threshold of water, has been experimentally verified in the range of 24-30 MPa over a frequency range of 0.3-3 MHz at 20°C. In this study, the effects of temperature on the intrinsic threshold was investigated both experimentally and theoretically. Single pulses with one high amplitude negative cycle at 1 MHz were applied to distilled, degassed water at temperatures ranging from 10°C-90°C. Cavitation was detected and characterized by passive cavitation detection and high-speed photography, from which the probability of cavitation was measured vs. pressure amplitude. The results indicated that the intrinsic threshold significantly decreases with increasing temperature, showing a nearly linear decreasing trend from 29.8±0.4 MPa at 10°C to 14.9±1.4 MPa at 90°C. Overall, this study supports our hypothesis that the intrinsic threshold is highly dependent upon the temperature of the medium, which may allow for better predictions of cavitation generation at body temperature*in vivo* and at the elevated temperatures commonly seen in high intensity focused ultrasound (HIFU) regimes.

4:40

2pBAb13. The effects of heat and mass transfer on free oscillations of a bubble in a viscoelastic, tissue-like medium. Eric Johnsen and Carlos Barajas (Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, ejohnsen@umich.edu)

Free oscillations of a bubble in soft tissue is of relevance to a variety of diagnostic and therapeutic ultrasound applications. Heat and mass transfer effects have been explored in the context of bubble oscillations in water; however, the extent to which they influence bubble oscillations in soft materials is presently unknown. Our objective is to use numerical modeling to predict bubble oscillations in viscoelastic, tissue-like media, while

accounting for heat and mass transfer. We numerically solve the Keller-Miksis equation to compute the Rayleigh collapse of a spherical bubble in a Kelvin-Voigt viscoelastic medium with finite-strain elasticity. The energy equation is solved inside and outside the bubble, and a mass conservation equation is solved for the vapor inside the bubble; equilibrium phase change is assumed. Using linear, small-amplitude perturbation theory, we investigate the bubble response. We show how the damping and oscillatory behavior depends on the eigenvalues of the full system; in particular, the time constant does not follow a monotonic relationship with respect to shear modulus. We also identify regimes in which solving the fully partial differential equations for energy and vapor concentration is not necessary. Finally, we interpret our results in the context of ultrasound applications.

4:55

2pBAb14. Modeling tissue response to a cavitation bubble in histotripsy. Lauren Mancia (Mech. Eng., Univ. of Michigan, 2016 Walter E. Lay Automotive Lab, Ann Arbor, MI 48109-2133, lamancha@umich.edu), Eli Vlaisavljevich, Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), and Eric Johnsen (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Histotripsy is a noninvasive focused ultrasound procedure that uses cavitation bubbles generated by high-amplitude ultrasound pulses to mechanically homogenize soft tissue. Experimental studies of histotripsy-induced cavitation in tissue phantoms and animal models have shown that tissue mechanical properties such as viscosity and elasticity affect cavitation threshold and bubble behavior. At present, however, the mechanisms responsible for tissue damage observed in histotripsy and other cavitation-inducing ultrasound treatments remain difficult to quantify. In this study, we simulated the dynamics of a single, spherical bubble in a Kelvin-Voigt-based viscoelastic solid, with nonlinear elasticity to better represent nanometer to micron-scale bubble growth. We applied the numerical model to calculate stress, strain, and strain rate distributions produced by a cavitation bubble exposed to forcing representative of a tensile histotripsy cycle. We found that stress and strain in excess of the ultimate tensile strength and fractional strain of most soft tissues occur at the bubble wall and decrease by at least two orders of magnitude within 50 microns from the bubble. Tissue mechanical properties were found to affect the magnitudes of stress and strain developed at different distances from the bubble. We will relate these results to experimentally observed correlates of tissue damage.

5:10

2pBAb15. Investigation of mechanical and dynamic characteristics for multifunctional contrast agents using atomic force microscopy and acoustic assessments. Gepu Guo, Qingyu Ma (School of Phys. and Technol., Nanjing Normal Univ., 1 Wenyuan Rd., Qixia District, Nanjing, Jiangsu 210023, China, guogepu@njnu.edu.cn), Juan Tu, Dong Zhang (Inst. of Acoust., Nanjing Univ., Nanjing, Jiangsu, China), and Shaotong Feng (School of Phys. and-Technol., Nanjing Normal Univ., Nanjing, Jiangsu, China)

Although multi-parameter fitting algorithms are often used for the characterization of coated-bubbles, it is inevitable to introduce uncertainty into the results. Therefore, it is urgent to develop some improved techniques to analyze the mechanical properties of the multifunctional microbubbles (MBs) accurately and systematically. By combining the measurements of atomic force microscopy, optical and acoustic detection with the simulations of coated-bubble dynamics, a comprehensive technology was developed to determine the size distribution, shell thickness, elasticity, and viscosity of multifunctional MBs precisely. Moreover, the impact of magnetic nanoparticles (MNPs) concentration on the multifunctional MBs' dynamic properties was studied systematically. It is demonstrated that, with the increasing MNPs concentration, the MB mean diameter and shell stiffness increased and ultrasound scattering response and inertial cavitation activity could be significantly enhanced. Although the shell thickness does not depend on the MNPs concentration, the increased MNPs concentration would generally result in enhanced bulk modulus and reduced bulk viscosity. The results of current studies demonstrate that the proposed single-parameter evaluation method could be helpful for further design and development of MB agents in clinic applications.

Session 2pEAa

Engineering Acoustics: Acoustic Transduction Material, Sensors, and Array Technologies

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Yasuhiro Oikawa, Cochair
Department of Intermedia Art and Science, Waseda University, 3-4-1 Okubo, Shinjuku-ku, Tokyo 169-8555, Japan

Invited Papers

1:00

2pEAa1. A nanotube thermophone projector array. Benjamin Dzikowicz, James F. Tressler, and Jeffery W. Baldwin (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, ben.dzikowicz@nrl.navy.mil)

Thermophone projectors fabricated using newly available nanoscale materials as elements hold the promise of a new transducer technology for the Navy. With no moving parts, they can operate over a broad frequency range and can be designed to be lighter and thinner than competing technologies. Scaling arguments and numerical models are explored to understand the parameter space for a gas-enclosed thermophone applicable to micro- and nanoscale elements. These models led to the development of an element made from a sparse horizontal array of single-walled nanotubes. Experiments are performed in NRL's Laboratory for Structural Acoustics using a specifically designed enclosure for underwater operations. Results and analysis of experiments using various fill gasses and internal pressures will be presented. [Work supported by NRL.]

1:20

2pEAa2. Investigation of carbon nanotubes for acoustic transduction applications. Thomas R. Howarth, Dehua Huang, Christian R. Schumacher, Nathanael K. Mayo, Donald L. Cox, Jeffrey E. Boisvert (NAVSEA Div. Newport, 1176 Howell St., B1346 R404A, Newport, RI 02841, thomas.howarth@navy.mil), Ali E. Aliev, and Ray H. Baughman (Alan G. MacDiarmid NanoTech Inst., The Univ. of Texas at Dallas, Richardson, TX)

Traditional acoustic transduction sources typically begin with the generation of an electrical excitation pulsed through an amplifier into an electroacoustic material to create a mechanical vibration which is then converted into an acoustic wave to produce sound. The lower the preferred transmitting frequency (and hence, longer acoustic range) desired, the larger the required size of the source. Often this means that for acoustic projectors producing sound at frequencies below a few kHz, that the electroacoustic device will need to be very large in order to produce very long sound waves. This has a limitation for incorporating low frequency sonars on smaller autonomous underwater vehicles (AUVs). The topic of our presentation is an acoustic source technology that relies on the conversion of thermal energy to acoustic energy. Though the concept was first demonstrated in 1917, the recent advent of carbon nanotubes (CNT) now makes it possible to transmit acoustic waves in small and affordable packages using the thermophone approach. The presentation begins with an overview of thermophones and a method for incorporating the CNTs into a useable transduction device. Discussions will include detailing on-going efforts for the encapsulation, packaging and further insight of the thermoacoustic energy conversion process. Recent 2016 experimental data will be shown. New start numerical and analytical modeling efforts and future development thrusts will be discussed.

1:40

2pEAa3. Thermophone modeling. Donald L. Cox, Thomas R. Howarth, Christian R. Schumacher, and Nathanael K. Mayo (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, donald.l.cox@navy.mil)

The use of carbon nanotube (CNT) films encapsulated within a pouch design pressurized with an inert gas has demonstrated promise for underwater sound projection. To augment this experimental demonstration, a modeling approach has been undertaken to capture the requisite physics and eventually provide a design tool to optimize performance. The modeling approach makes use of the finite element method as programmed in the commercially available COMSOL software. A multi-physics approach is taken where models are coupled that include: electrically stimulated heat, heat transfer to a compressible, viscous fluid, structural response of a thin shell to this computational fluid domain, and external acoustic propagation from the outer surface of the structure. The intent is to capture the physical response of the sensor tested and understand the interaction dynamics of the various media. The multiphysics solution approach taken in this work involves the use of a segregated, time dependent solution of coupled, linear and nonlinear problems. As a result, the process is complicated and computationally intensive. Modeling results will be presented and compared with experimental data.

2:00

2pEAa4. A design of broadband constant beam pattern transducers. Dehua Huang (NAVSEANPT, Howell St., Newport, RI 02841, DHHuang@cox.net)

A constant beam pattern (CBP) transducer is an acoustic transducer, where its beam patterns are independent of frequency. The theory and numerical simulations for constant beam pattern transducer design are introduced. For a hemispherical design, if the radial velocity distribution on the surface of a conventional hemisphere transducer is a single complete Legendre polynomial, the far field angular beam pattern shows the same Legendre polynomial distribution for all frequencies above its cut-off frequency, based on spherical Hankel function asymptotic approximation to the solutions from Helmholtz wave equation. Because of orthogonality, Legendre polynomials form a complete set, per Sturm-Liouville theory that an arbitrary velocity shading function can be expanded by Legendre series. Each polynomial within this Legendre series contributes its share to the far field, such that the converging acoustic beam pattern displays the same shape as the original shading function itself. Numerical simulations with various samples include (cos(theta))^3 and Gaussian, a well as equal sidelobe suppression classic Dolph-Chebyshev functions used as shading and achievable as beam patterns for a broadband frequency range.

2:20

2pEAa5. Optical sensing of sound fields: Non-contact, quantitative, and single-shot imaging of sound using high-speed polarization camera. Kenji Ishikawa, Kohei Yatabe, Yusuke Ikeda, Yasuhiro Oikawa (Intermedia Art and Sci., Waseda Univ., Ohkubo 3-4-1, Bldg 59, 407-2, Shinjuku-ku, Tokyo 169-8555, Japan, k-ishikawa@fuji.waseda.jp), Takashi Onuma, Hayato Niwa (Photron Ltd., Chiyoda-ku, Tokyo, Japan), and Minoru Yoshii (Kiyohara Optics, Shinjuku-ku, Tokyo, Japan)

Imaging of a sound field aids understanding of the actual behavior of the field. That is useful for obtaining acoustical spatial characteristics of transducers, materials and noise sources. For high spatial resolution imaging, optical measurement methods have been used, thanks to its contactless nature. This paper presents sound field imaging method based on parallel phase-shifting interferometry, which enables to acquire an instantaneous two-dimensional phase distribution of light. Information of sound field is calculated from the phase of light based on the acousto-optic theory. The system consists of a polarization interferometer and high-speed polarization camera, whose measurement points of a single image are 512×512 and spatial resolution is about 0.2 mm $\times$ 0.2 mm. Therefore, the system can image a sound field with much higher spatial resolution compared with conventional imaging methods. The maximum framerate, which is corresponding to the sampling frequency, is 1.55 M frames per second. This paper contains the principle of optical measurement of sound, the description of the system, and several experimental results including imaging of a transducer-generated sound field and propagation and reflection of a sound wave.

2:40

2pEAa6. Study to fabricate high-quality and portable parametric speakers. Jun Kuroda (Dept. of Intermedia Art and Sci., Waseda Univ., 3–4–1 Ohkubo, Tokyo 169–8555, Japan, jun-kuroda@suou.waseda.jp), Shota Minami, and Yasuhiro Oikawa (Dept. of Intermedia Art and Sci., Waseda Univ., Shinjuku-ku, Tokyo, Japan)

Parametric speakers provide narrow directional audible sounds using modulated ultrasounds. Parametric speakers are employed to build audio communication with privacy protection, which is expected to be installed into various instruments. To respond to this requirement, slim-sized parametric speakers delivering high-quality sound must be fabricated. Ultrasonic transducers usually consist of piezoelectric elements and metal resonators, which have a number of resonant modes. To demodulate audible sounds by the non-linearity of a finite amplitude sonic wave, a large amplitude carrier wave must be emitted. Therefore a predominant resonant mode of an ultrasonic transducer is used to transmit carrier wave. Moreover, frequency band around the predominant resonance is used to transmit modulated Fourier components of signals. While considering these principals of sonic transmission, two issues must be solved to fabricate a slim-sized and high quality parametric speaker, i.e., 1) widening of available bandwidth and improvement of electroacoustic efficiency and 2) optimization of transmitter circuits and modulation method based on characteristics of ultrasound transducers. It is necessary to study parametric speakers as a total system comprised by ultrasound transducers, transmit circuits, and signal processing of modulation, to solve these issues. This paper denotes outline of these issues and the present status of our studies.

Session 2pEAb

Engineering Acoustics: Ocean Acoustics Analysis

Robert Taylor, Cochair
Mechanical Engineering, The Catholic University of America, 620 Michigan Ave. NE, Washington, DC 20064

Veronica Koh, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Contributed Papers

3:15

2pEAb1. Numerical investigation on transmission loss of sound above sea surface. Robert Taylor, Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, 45taylor@cua.edu), Teresa Ryan (Mech. Eng., Eastern Carolina Univ., Greenville, NC), and Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC)

Determination of the transmission loss of sound travelling over the ocean with various sea states involves a number of variables. One method to approximate the random roughness of the surface is to use the variance spectrum model as proposed by Pierson and Moskowitz. A numerical solution of the wave equation is presented for a small elevation angle over the sea surface at different sea states. This new model has the capability to include both the thermal gradient and wind profile above the sea. The solutions are derived with an implementation of the Crank-Nicolson Parabolic Equation, and the ocean wave roughness profile is introduced into the numerical model by the Generalized Terrain Parabolic Equation method. These results are used to investigate the effect of the sea roughness on the sound transmission loss for different sea states.

3:30

2pEAb2. Study of two-dimensional underwater particle velocity pickup sensor. Zhao Tianji (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, ztj13613621886@126.com)

Addressing the subject of a suspended co-oscillating vector hydrophone application platform, a new sound wave receiving theory model of underwater particle velocity pickup sensor was established. Based on the study of the inner sound field of an elastic cylinder vibrating freely under the action of sound waves in theory, the influence of material and geometry parameters on the frequency response of particle velocity has been analyzed. According to the results of parameter optimization and taking into account engineering needs, one sample of two-dimensional particle velocity pickup sensor has been designed and tested. By comparing the results of sensitivity and directivity tested in a calibration device, the general rules of theoretical analysis have been verified. The theory and experiments confirmed the feasibility of particle velocity pickup sensors in engineering applications.

3:45

2pEAb3. Adaptive statistical learning models for long-range sound propagation. Carl R. Hart, D. K. Wilson (U.S. Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@usace.army.mil), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), and Edward T. Nykaza (U.S. Engineer Res. and Development Ctr., Champaign, IL)

Uncertainties in source characteristics, meteorological conditions, and topographic variations present formidable challenges for accurately predicting long-range outdoor sound propagation. Numerical propagation models inherently assume perfect knowledge of these uncertain variables and are fixed in a modeling sense. In contrast, statistical learning models can incorporate new observations to update the underlying prediction model. Past work has shown that statistical learning models trained on synthetic data for predicting long-range sound propagation have, at best, an overall root-mean-square error (RMSE) of about 5 dB. This limit appears to be imposed by modeled atmospheric turbulence. It is hypothesized that this prediction limit may be lowered as observational data are incorporated into trained statistical learning models. Furthermore, data are assimilated by a Kalman filtering process for the purpose of updating knowledge of the atmospheric and source characteristics. Within the prediction phase three different statistical learning models are compared: an ensemble neural network, a cluster-weighted model, and a random forest regression model. The efficacy of data assimilation is evaluated with respect to each statistical learning model.

4:00

2pEAb4. Hydrophone calibration system. Alper Biber and Ata C. Corakci (Matererials, Tubitak Mam, Dr. Zeki Acar Cad., Gebze, Kocaeli 41470, Turkey, alper.biber@tubitak.gov.tr)

A system to calibrate underwater electro-acoustic hydrophones by primary method, namely, Reciprocity method, in wide frequency range from few kilohertz to megahertz is presented. The aim is to realize primary level "Hydrophone Calibration System" covering from few kilohertz to upper limit of underwater applications with very limited restrictions in one simple system. This system not only covers wide frequency range of underwater applications but also offers low-cost solution to reach enough uncertainty level required for primary calibration according to standards [1]-[3]. The system is based on very high resolution PC oscilloscope of Pico Technology, namely, PicoScope 4262, used for primary data acquizition and digitizing. Primary signal generator and power amplifier, available in market, are integrated into system for purpose. The detailed description is presented and uncertainty budget is discussed.

4:15

2pEAb5. Design improvements of a low cost underwater projector based on clarinet acoustics. Veronica Koh and Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vwk102@psu.edu)

Low frequency (ka<<1) underwater projectors are desired for long range sonar applications and oceanography due to their low absorption losses. However, such low frequency projectors (LFP) tend to be large in size and costly to build. This study aims to assess the feasibility of an underwater clarinet as a low cost LFP. Previous proof-of-concept work on a lab prototype showed stable tone production but with resonance frequencies that were lower than predicted and significant harmonic levels. In this study, a new pressure chamber design for housing the clarinet mouthpiece is built and tested. Improvements in the equivalent circuit model for the underwater clarinet are also made to include the elastic effects of the clarinet wall and the pressure chamber impedance, to better predict its performance. The current measurement results using both PVC and steel resonator for the underwater clarinet will be presented and compared with the improved model. The trade-off considerations between the design parameters of the underwater clarinet and its performance will also be discussed.

4:30

2pEAb6. Comparison of computational schemes in the design of a large-scale projector array. Eunghwy Noh, Wonjong Chun, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., Eng. Bldg. A391, 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, hwyya@yonsei.ac.kr), Woosuk Chang, and Hongwoo Youn (Maritime, LIG Nex1, Seongnam, South Korea)

Modeling and simulation for the design of a projector array involves computations in the transducer and acoustic domains. In the transducer domain, each transducer is often described by the two-port, equivalent-circuit model, in which the radiation impedance containing the influence of the neighboring transducers as well as the medium is not known a priori. To obtain the radiation impedance and the subsequent transducer response a set of coupled transducer equations must be solved simultaneously in conjunction with the (computationally expensive) wave-field calculation in the acoustic domain. In this talk, we compare two computational schemes for speed, which differ in the way the mutual interaction of transducers is treated: the first scheme explicitly accounts for the mutual interaction in terms of the mutual impedance, whereas in the second scheme the interaction is reflected implicitly in the iterative solution of the transducer equations. Here, the speed is largely determined by the number of wave-field calculations required for the desired accuracy. Comparison is made for the case of a cylindrical array of 288 tonpilz transducers. [This work was sponsored by LIG Nex1 Co., Ltd.]

4:45

2pEAb7. Active control of low-frequency underwater echoes using an array of tile projectors. Wonjong Chun, Eunghwy Noh, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 120-752, South Korea, wonjong.chun@yonsei.ac.kr), Jaepil Kim, Youna Ji, Youngcheol Park (Comput. & Telecommunications Eng. Div., Yonsei Univ., Wonju, Gangwon-do, South Korea), and Youngsoo Seo (Agency for Defense Development, Changwon, Gyeongsangnam-do, South Korea)

With the introduction of low-frequency active sonar in anti-submarine warfare, there is a growing need for a novel underwater invisibility device that could replace the existing passive anechoic tiles. In this talk, we describe an experiment on active reduction of underwater echoes at low frequencies using an array of tile-shaped projectors. Each tile projector was designed with the aid of finite-element computations and tested in an acoustic tank for transmit voltage sensitivity and directivity. An array of tile projectors, covering a scale model submarine in a large acoustic tank, were driven by control signals that were intended to produce the impedance match between water and the object. Depending on the frequency of the incident wave, echo reduction as large as 6 dB was achieved. [This work has been supported by the Low Observable Technology Research Center program of Defense Acquisition Program Administration and Agency for Defense Development.]

5:00

2pEAb8. Underwater parametric array source transducer composed of PZT rods and thin polymer plate with high power efficiency for wideband sound generation. Hongmin Ahn, Yonghwan Hwang, and Wonkyu Moon (Mech. Eng., POSTECH, Hyoja-dong, Nam-gu, Pohang, Gyeongsangbuk-do 790784, South Korea, idealcircuit@postech.ac.kr)

Nowadays 1-3-type PZT composite transducers are widely used for underwater parametric array (PA) sound source since they can generate highly directional acoustic beam with high intensity, which is the requirements for the PA acoustic phenomenon that results from nonlinearity of a medium. Thanks to the relatively small mechanical quality factor of 1-3 PZT composite, the transducer using it can generate high intensity sounds by relatively low input voltage in the relatively wide frequency band around its resonance frequency. The low mechanical quality factor, however, increase the internal mechanical loss inside the transducer. Consequently, the acoustic radiation efficiency may be lower than expected. In this paper, a dual-resonant-frequency PZT rods are integrated into a thinner polymer plate. Then, the radiation surface increases to enhance radiation power efficiency. The PZT rods with two different lengths, are integrated in the process of molding a polymer plate. The fabricated transducer are operated with the out-of-phase driving method for a dual resonance transducer to generate direct sound beam with a wide frequency bandwidth. The sound pressure levels (SPLs) of the primary waves of the first and second resonant frequencies were 197 dB and 203 dB (re = 1μP), respectively, at 9 m. The acoustic radiation efficiencies at 98 kHz and 135 kHz were 50% and 33%, respectively. The DFW generated by the PA had an SPL of 150 dB (re = 1μP @ 30 kHz) and a high directivity of 3.4° half-power beam width.

Session 2pMU

Musical Acoustics and Signal Processing in Acoustics: Music Signal Processing II

Masataka Goto, Cochair
National Institute of Advanced Industrial Science and Technology (AIST), IT, AIST, 1-1-1 Umezono, Tsukuba 305-8568, Japan

James W. Beauchamp, Cochair
School of Music and Electrical & Computer Eng., University of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824

Invited Papers

1:15

2pMU1. Songle Widget: A web-based development framework for making animation and physical devices synchronized with music. Masataka Goto (National Inst. of Adv. Industrial Sci. and Technol. (AIST), IT, AIST, 1-1-1 Umezono, Tsukuba, Ibaraki 305-8568, Japan, m.goto@aist.go.jp), Kazuyoshi Yoshii (Kyoto Univ., Kyoto, Japan), and Tomoyasu Nakano (National Inst. of Adv. Industrial Sci. and Technol. (AIST), Tsukuba, Ibaraki, Japan)

This paper describes a web-based multimedia development framework, *Songle Widget* (http://widget.songle.jp), that makes it possible to control computer-graphic animation and physical devices such as lighting devices and robots in synchronization with music publicly available on the web. Development of applications featuring rigid synchronization with music playback on the web was difficult because audio signals of the playback are not accessible on a web browser and manual annotation is time-consuming. To overcome such difficulties, Songle Widget makes it easy to develop web-based applications with rigid music synchronization by leveraging music-understanding technologies based on signal processing. Songle Widget is implemented by using our public web service called *Songle* (http://songle.jp) that automatically analyzes songs on the web and annotates four important types of musical elements (music structure, hierarchical beat structure, melody line, and chords). More than 1,000,000 songs on music- or video-sharing services have been analyzed by Songle and can readily be used by music-synchronized applications. Since errors are inevitable when elements are annotated automatically, Songle Widget takes advantage of a Songle's crowdsourcing interface that enables users to correct errors. This is effective when applications require error-free annotation. We made Songle Widget open to the public, and its capabilities and usefulness have been demonstrated.

1:35

2pMU2. Rethinking audio production tools. Bryan Pardo (EECS, Northwestern Univ., 2133 Sheridan Rd., Evanston, IL 60208, pardo@northwestern.edu)

Potential users of audio production software, such as equalizers and reverberators, may be discouraged by the complexity of the interface and a lack of clear affordances in typical interfaces. We seek to simplify interfaces for audio production (e.g., mastering a music album with ProTools), audio tools (e.g., equalizers), and related consumer devices (e.g., hearing aids). Our approach is to center the interaction around user-provided examples, an evaluative paradigm ("I like this sound better than that sound") and descriptive language (e.g., "Make the violin sound 'warmer.'"). To build interfaces that use descriptive language, a system must be able to tell whether the stated goal is appropriate for the selected tool (e.g. making the violin "warmer" with a panning tool does not make sense). If the goal is appropriate for the tool, it must know what actions need to be taken (e.g., add some reverberation). Further, the tool should not impose a vocabulary on users, but rather understand the vocabulary users prefer. In this talk, Prof. Pardo describes recent work in evaluative interfaces, crowdsourcing a vocabulary for language-based production tools, and language-based interfaces for production tools.

1:55

2pMU3. Smart loop sequencer: An audio-based approach for ease of music creation. Tetsuro Kitahara (Nihon Univ., 3-25-40, Sakurajosui, Setagaya-ku, Tokyo 1568550, Japan, kitahara@chs.nihon-u.ac.jp)

A loop sequencer is expected to be a good tool for non-musicians to compose music. However, it is not easy to appropriately select music loops because a loop sequencer usually has a large scale of loop collection. In this paper, we propose a smart loop sequencer that automatically selects music loops based on the degree of excitement input by the user. The user expresses a temporal evolution of a desired excitement as a continous curve using the mouse of the PC. Then, the system searches for a sequence of music loops that generates the most similar excitement to the user's input. This process is formulated with a hidden Markov model (HMM). A combination of music loops is regarded as a state of the HMM, and an excitement input by the user is regarded as an emission value. Given a sequence of excitement, the most likely combination of music loops for every measure is estimated with the Viterbi algorithm. Experimental results show that non-experts of music easily compose musical pieces with our system.

2:15

2pMU4. Audio scene transformation using informed source separation. Sylvain Marchand (Univ. of La Rochelle, Laboratoire Informatique, Image et Interaction (L3i), Ave. Michel Crepeau, La Rochelle 17042, France, sylvain.marchand@univ-lr.fr)

Giving the listener the freedom to transform the audio scene in real time is a challenging task. An important example of transformation is (re)spatialization: moving sound sources in space. This is equivalent to source separation. Indeed, moving all the sources of the scene but one away from the listener separates that source. And moving separate sources then rendering from them the corresponding scene is easy. But allowing this spatial transformation/source separation with a sufficient quality is a (too) challenging task for classic approaches, since it requires an analysis of the audio scene with inevitable (and often unacceptable) estimation errors. We introduced the informed approach, which consists in inaudibly embedding some additional information in the audio signal. This information, coded with a minimal rate, aims at increasing the precision of the analysis/separation. Thus, the informed approach relies on both estimation and information theories. Since the presentation at the ASA meeting in 2010, several informed source separation methods were proposed. Among the best methods is the one based on spatial filtering (beamforming), with the spectral envelopes of the sources (perceptively coded) as additional information. This allows the manipulation of the sound sources with an unequaled (controllable) quality.

2:35

2pMU5. Evaluation of singing voice by amateur singers from various aspects. Akinori Ito (Graduate School of Eng., Tohoku Univ., 6-6-5 Aramaki-aza-Aoba, Aoba-ku, Sendai, Miyagi 980-8579, Japan, aito@spcom.ecei.tohoku.ac.jp)

Many research works have been conducted for evaluating singing voice so far. Most of the works aimed to investigate the voice of professional vocalists. Besides, evaluation of amateur singers is useful for developing entertainment applications such as scoring of Karaoke. In this presentation, two of such applications are introduced: an evaluation of singing enthusiasm and the detection of mistakes of sung lyrics. The evaluation of singing enthusiasm is based on three kinds of features: the A-weighted power, the vibrato-related feature, and the fall-down feature. Using these features, perceived singing enthusiasm can be estimated relatively accurately. In addition, singers' intention of enthusiasm can also be estimated from the fluctuation of the segmental power of the voice. Detection of mistakes of sung lyrics is based on the dynamic time warping between the input singing voice and the reference singing voice. To normalize the singer-dependent variation of voice features, we employed singer normalization based on linear regression. Based on the DTW alignment, frame-by-frame distances between the two voices are obtained. After temporal smoothing of the distances, the lyrics errors are detected based on the peak detection of the distance. High detection accuracy is obtained when the errors are based on phrases.

2:55

2pMU6. Statistical models for the indirect acquisition of violin bowing controls from audio analysis. Alfonso Perez-Carrillo (Music Technol. Group, Universitat Pompeu Fabra, Roc Boronat 138, Barcelona 08018, Spain, alfonso.perez@upf.edu)

Acoustical studies of sound production in bowed string instruments show that there is a direct relationship between bowing controls and the sound produced. During note sustains the three major bowing parameters that determine the characteristics of the sound are the bowing force (bforce), the bowing distance to the bridge (bbd) and the bowing velocity (bvel). During note attacks the third parameter is acceleration (bacc) rather than velocity. We are interested in understanding this correspondence between bowing controls and sound but in the opposite direction, i.e., mapping sound features extracted from an audio recording to the original bowing controls used. This inverse process is usually called indirect acquisition of gestures and is of great interest in different fields ranging from acoustics and sound synthesis to motor learning or augmented performances. Indirect acquisition is based on the processing of the audio signal and it is usually informed on acoustical or physical properties of the sound or sound production mechanism. In this paper, we present indirect acquisition methods of violin controls from an audio signal based on training of statistical models with a database of multimodal data (control and sound) from violin performances.

3:15–3:30 Break

3:30

2pMU7. Adaptive aggregation of regression models for music emotion recognition. Satoru Fukayama and Masataka Goto (National Inst. of Adv. Industrial Sci. and Technol. (AIST), IT, AIST, 1-1-1 Umezono, Tsukuba, Ibaraki 305-8568, Japan, s.fukayama@aist.go.jp)

We present a method for music emotion recognition which adaptively aggregates regression models. Music emotion recognition is a task to estimate how music affects the emotion of a listener. The approach works by mapping acoustic features into space that represents emotions. Previous research has centered on finding effective acoustic features, or applying a multi-stage regression to aggregate the results obtained by using different acoustic features. However, after training regression models, the aggregation happens in a fixed way and cannot be adapted to acoustic signals with different musical properties. We indicate that the most effective feature in estimation differs depending on what kind of emotion we are estimating, and propose a method that adapts the importance of each feature in estimating the emotion. We exploit the variance obtained with Gaussian process regression to measure the confidence of the estimated result from each regression model. The formula for aggregating results from different regression models is derived through the maximum likelihood estimation. We confirmed with an experiment comparing different aggregation approaches that our adaptive aggregation is effective in improving recognition accuracy.

3:50

2pMU8. Experimental modal analysis/synthesis of saxophone input impedances. Esteban Maestre (Music Technol., McGill Univ., Roc Boronat 138, Barcelona 08018, Spain, esteban.maestre@upf.edu) and Gary P. Scavone (Music Technol., McGill Univ., Montreal, QC, Canada)

We present an experimental modal analysis/synthesis framework for the accurate modeling and efficient digital simulation of saxophone input impedances. Starting from impedance measurements, we perform spectral-domain modal analysis by iteratively minimizing the error between measurements and synthetic impedance responses. Synthesized impedances are formulated as passive digital filters in parallel form, enabling the construction of passive digital reflectances with potential use for efficient sound generation via coupling to a reed non-linear model. In this talk, we provide an overview and discuss our ongoing developments, pointing toward future directions in using this technique for real-time sound synthesis.

4:10

2pMU9. Toward a real-time parametric percussion instrument based on a waveguide mesh. Tamara Smyth and Jennifer Hsu (Music, Univ. of California San Diego, 9500 Gilman Dr. MC 0099, La Jolla, CA 92093-0099, trsmyth@ucsd.edu)

In this work, we aim to create a real-time parametric percussion instrument in which tone quality and note duration may be controlled by the user. Though the well-known waveguide mesh can be used to model two-dimensional wave propagation on drum heads of different size/shape, with variable boundary conditions ultimately influencing rate of sound decay, the prohibitive computational cost of its time-domain implementation makes it unsuitable for real-time performance. Here, the square waveguide mesh, having variable size and parametric boundaries, is replaced by its (exactly) equivalent transfer function—the ratio of the output to the input when excited and tapped at its center. The transfer function is obtained by first generating a system of difference equations describing the output of scattering junction ports—a process that is simplified by the symmetry in the square mesh when excited and tapped at its center. The final transfer function is stored for various mesh sizes, with coefficients being represented symbolically as a function of boundary parameters that can be set (for example, according to desired note duration) during real-time performance.

4:30

2pMU10. Recent extensions to topological and nonlinear aspects of wave digital filter theory. Kurt J. Werner (CCRMA, Stanford Univ., 223 Ayrshire Farm Ln., Apt. 205, Stanford, CA 94305, kwerner@ccrma.stanford.edu)

The Wave Digital Filter approach can be used to create computational models of lumped reference systems, including rectilinear mechanical, rotational mechanical, acoustical, and electronic systems. When they are applicable, Wave Digital Filters enjoy properties including modularity, accuracy, and guaranteed incremental passivity that make them attractive in the context of musical instrument and audio effect simulation. However, the class of reference systems that can be modeled with Wave Digital Filters has historically been restricted to systems with simple topologies (which can be decomposed entirely into series and parallel connections) and which contain only a single "nonadaptable" element (e.g., a diode, ideal source, or switch). This talk details recent Wave Digital Filter advances from Stanford University's Center for Computer Research in Music and Acoustics (CCRMA), which broaden the applicability of Wave Digital Filters to reference systems with any topology and an arbitrary number of nonlinearities or other nonadaptable elements. These advances have enabled many new and previously intractable musical circuit simulations at CCRMA, including active tone stages and transistor/diode/feedback clipping from guitar distortion pedals, the electromechanical Hammond organ vibrato/chorus system, guitar tube pre-amplifiers and tone stacks, circuits involving operational amplifiers, drum machine circuits, and transistor amplifiers.

4:50

2pMU11. Music signal processing by the human brains: Studies on the strategies used by professional pianists to efficiently sight-read music. Eriko Aiba (Dept. of Mech. and Intelligent Systems Eng., Univ. of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 1828585, Japan, aiba.eriko@uec.ac.jp)

The human brain is an ingenious signal processing system. When musicians play instruments, their brains must process a huge amount and various types of information in parallel. Particularly, the sight-reading of piano music requires the processing of an enormous amount of information as piano music includes many chords and is written on the great staff (or grand staff). Pianists have to read the score, interpret the music, search for the keys to be played while planning the motions of fingers, and control their fingers. In addition, they must adjust the sound intensity and usage of the sustaining pedal according to the output sound. All these are performed simultaneously and successively. The sensory information or signals to be processed include many different modalities, such as visual, auditory, tactile, and motion sensing. In order to complete this complicated task, it is important that the pianists efficiently process this information. In this presentation, I introduce our experimental studies investigating the strategies used by professional pianists to improve the efficiency of information processing while sight-reading music.

Session 2pNS

Noise: Community Response to Transportation Noises

James E. Phillips, Cochair
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Takashi Yano, Cochair
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Chair's Introduction—1:00

Invited Papers

1:05

2pNS1. Annoyance due to transportation noises in Vietnam. Thulan Nguyen, Takashi Yano (Graduate School of Sci. and Technol., Kumamoto Univ., 2-39-1 Kurokami, Chuo-ku, Kumamoto 8608555, Japan, nguyen@kumamoto-u.ac.jp), Tsuyoshi Nishimura (Graduate School of Eng., Sojo Univ., Kumamoto, Japan), and Tetsumi Sato (Faculty of Eng., Hokkai Gakuen Univ., Sapporo, Japan)

Though understanding differences in the community response to noise among countries is essential to an effective implementation of noise policies, very little noise annoyance data have been accumulated in countries outside Europe and North America. To contribute to Vietnamese and global noise policies, socio-acoustic surveys on the community response to transportation noise have been carried out in Vietnam since 2005. A total of around 9,900 responses were obtained from these surveys. In this paper, a method of drawing exposure-response curves for noise annoyance in Vietnam is presented. Representative exposure-response relationships are proposed for road traffic and aircraft noise annoyance in Vietnam and compared with those of Europe and Korea. The results show that the aircraft noise annoyance curve for Vietnam was slightly higher than the curve obtained by the EU but considerably lower than that for Korea. Vietnamese respondents were less annoyed by road traffic noise than respondents in Europe and Korea. The frequency of use and the attitude to motorbikes and airplanes were found to moderate noise annoyance in Vietnam. This study also shows a method to quantify the difference in the prevalence of annoyance measured by five-point verbal and 11-point numerical scales.

1:25

2pNS2. Community response to aircraft noise: A step-change study around Hanoi Noi Bai International Airport. Linh T. Nguyen (Sound Traffic Environment Planning INC, Kamikumamoto 3-21-21, Princess Mansion B 402, Kumamoto 8600079, Japan, nguyen.thaolinh0912@gmail.com), Lan T. Nguyen (Graduate School of Sci. and Technol., Kumamoto Univ., Kumamoto, Japan), Takashi Yano, Tsuyoshi Nishimura (Faculty of Information, Sojo Univeristy, Kumamoto, Japan), Tetsumi Sato (Faculty of Eng., Hokkai Gakuen Univ., Sapporo, Japan), Makoto Morinaga (Defense Facilities Environment Improvement Assoc., Tokyo, Japan), and Ichiro Yamada (Aviation Environment Res. Ctr., Tokyo, Japan)

A new terminal building in Hanoi Noi Bai International Airport (Vietnam) opened in December 2014 enhanced the capacity of serving more flights, entailing a step-change in aircraft noise exposure. Three social surveys were conducted around the airport in September 2014, March 2015, and September 2015 to study such a circumstance. Albeit the number of flights increased about 21-24% after the opening of the new terminal, aircraft noise exposure (L_{den}) stayed during the two first surveys (44, 45—66 dB) but increased in the last survey (49—69 dB). Also, noise level at particular sites varied considerably among surveys, especially at sites located near the airport. Curves of exposure—response relationships of three surveys were drawn by using logistic regression. The curve for the second survey fits on the first one with approximately 5% higher while the last survey's curve is remarkably steeper than others. All three curves of this study located above and notably steeper than the curve of EU's position paper, which means people in Hanoi were more annoyed than people in EU. One of the worth mentioning factors for causing these outcomes was the re-operation of the runways which was closed for maintenance since August to December 2014.

1:45

2pNS3. Activity disturbances caused by Shinkansen railway noises through the meta-analysis in Japan. Takashi Morihara (National Inst. of Technol., Ishikawa College, Kitachujo, Tsubata, Ishikawa 929-0392, Japan, morihara@ishikawa-nct.ac.jp), Shigenori Yokoshima (Kanagawa Environ. Res. Ctr., Hiratsuka, Japan), and Takashi Yano (Kumamoto Univ., Kumamoto, Japan)

More than 50 years have passed since the Tokaido Shinkansen railway, the first high-speed railway, opened in 1964. The network of Shinkansen lines has been continuously expanded in Japan: the Hokuriku and Hokkaido Shinkansen lines were partially opened in 2015 and 2016, respectively. To preserve the living environment and to protect human health, the Japanese government enacted the Environmental Quality Standards for Shinkansen Superexpress Railway Noise in 1975. Many social surveys on community response to Shinkansen railway noise and vibration have been conducted. A part of these surveys data have been accumulated in Socio-Acoustic Survey

Data Archive. We reanalyzed data from seven social surveys along six Shinkansen lines carried out from 1995 to 2013. The number of response to the question of daily activities was from 5441 to 1997 samples. TV/radio listening disturbance was greater than other listening disturbances at the same noise level. The evaluation of awakening exceeded 10% above the night time noise level of about 45 dB. Through the comparison between the survey, it was indicated that the effect of differences in the number of train service per day.

2:05

2pNS4. Community response to transportation noises in Japan: Secondary analysis using the Japanese socio-acoustic survey data archive. Shigenori Yokoshima (Kanagawa Environ. Res. Ctr., 1-3-39, Shinomiya, Hiratsuka, Kanagawa 2540014, Japan, yokoshima@k-erc.pref.kanagawa.jp), Makoto Morinaga (Defense Facilities Environment Improvement Assoc., Minato-ku, Japan), Takashi Yano (Kumamoto Univ., Kumamoto, Japan), Takashi Morihara (Ishikawa National College of Technol., Tsubata-town, Japan), and Keiji Kawai (Kumamoto Univ., Kumamoto, Japan)

A number of socio-acoustic surveys on community response to transportation noises have been carried out in Japan since 1970s. The findings from early precious surveys contributed to the establishment of Noise Regulation Law and Environmental Quality Standards regarding noises. However, micro datasets consisting of noise exposure and reactions to noise from recent studies have yet to be accumulated into a unified system. Dose-response relationships obtained through re-analysis of micro-data form the basis of the progress of effective noise policies. Thus, the Japanese Government faces many difficulties in reviewing noise policies. To develop an archive of community responses to noise, we established the Japanese Socio-Acoustic Survey Data Archive (J-SASDA) and have started the secondary analysis since 2011. In this presentation, we introduce the outcomes of our activity. First, we compare exposure-annoyance relationships among the modes of transportation using the J-SASDA and determine the curves per mode of transportation. Next, we focus on the comparison of exposure-annoyance relationships for transportation noises among European and American countries, Vietnam and Japan. Finally, we intend to establish the Asian-SASDA (A-SASDA) in collaboration with researchers from neighboring Asian countries and explain the current status and problem.

2:25

2pNS5. Centerline rumble strip sound level evaluation—Comparison of four designs. David Braslau (David Braslau Assoc., Inc., 6603 Queen Ave. S, Ste. N, Richfield, MN 55423, david@braslau.com), Edward Terhaar (Wenck Assoc., Inc., Maple Plain, MN), and Katie Fleming (Minnesota Dept. of Transportation, St. Paul, MN)

Results of sound level monitoring of four alternative centerline sinusoidal rumble strip designs are discussed: a single strip 14 inches wide and a double strip of two 8 inch wide strips 4 inches apart each with depths—3/8 inch or $1/2$ inch. These results supplement those presented at the Pittsburgh 2015 meeting (3aNS5). Tests were performed at 60 mph with three different vehicles—passenger car, pickup truck, and a six-wheel heavy maintenance truck. One-third octave band sound levels were taken 50 and 75 feet from the edge of the roadway, as well as inside the vehicle. Digital audio recordings were also taken. The single 14 inch design with $1/2$ inch depth was recommended for implementation by MnDOT since it provided good driver feedback but less exterior noise than the other designs for passenger cars but also good results for pickup trucks, a significant portion of the vehicle fleet. Based upon an evaluation of alternative rumble strips by motorcycles and bicycles at the MnROAD test facility, the single strip was also found more desirable. However, it was recommended that, in areas where there is extreme sensitivity to exterior rumble strip noise, the shallower 3/8 inch design would still provide adequate driver feedback.

2:45

2pNS6. Applications of automatic equipment identification to studies of rail noise and vibration in North America. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

When conducting studies of noise and vibration impacts of existing rail lines, it is useful to identify and log train movements. Since the early 1990s, all railcars and locomotives in North America have been outfitted with automatic equipment identification (AEI) tags. AEI is an RFID system that allows for automated electronic identification of passing equipment with unattended trackside readers. These data reveal the makeup, direction, length, speed, and other useful parameters of each passing train. In order to assess the feasibility of using AEI readers in noise studies, a field study was conducted using a temporary trackside reader and correlated sound and vibration monitors. This paper will present a technical overview of the AEI system and the results of the field study.

3:05–3:20 Break

Contributed Papers

3:20

2pNS7. Transportation noise: Nuisance or disability? Abigail Bristow (Loughborough Univ., School of Civil and Bldg. Eng., Loughborough LE11 3TU, United Kingdom, a.l.bristow@lboro.ac.uk)

This paper reviews the state of the art with respect to the valuation of transportation noise nuisance. It considers recent movements away from values based on revealed or stated preferences (SP) to values based on disability weights. Posing the question as to whether the experience of noise nuisance constitutes a disability. Environmental noise imposes amenity losses and adverse health outcomes on the population. As the world becomes more urban, increasingly motorized and aviation growth persists, noise nuisance is likely to affect ever more people. In identifying and prioritising interventions to reduce experienced noise, it is essential to identify the benefits not just in terms of decibel reductions but in terms of the economic benefits of achieving such changes so these may be compared with costs of interventions to ensure best value for money. The values from a recent meta-analysis of SP studies are compared with those derived using the more conventional hedonic pricing (HP) approach including meta-analyses of HP studies of aircraft noise and more recent developments with respect to assessing annoyance in the form of disability adjusted life years (DALYS) and applying a value of a quality adjusted life year (QALY) to this.

3:35

2pNS8. Higway traffic noise effects on annoyance. Aybike Ongel (Civil Eng., Bahcesehir Univ., Ciragan Cad No: 4 Besiktas, Istanbul 34349, Turkey, aybike@gmail.com), Mustafa Ilicali (Istanbul Ticaret Univ., Istanbul, Turkey), Didem Oguz (Baskent Univ., Istanbul, Turkey), Betul Ogutmen (Numune Hospital, Istanbul, Turkey), Tolga Guvenc (Siyami Ersek Hospital, Istanbul, Turkey), and Nurten Sayar (Marmara Univ., Istanbul, Turkey)

Noise pollution has been a major issue in the last decade. Noise was shown to cause adverse health effects including noise-induced hearing impairment, annoyance, sleep disturbance, and cardiovascular diseases. Transportation noise has been the most common one among all noise sources in urban areas. A study has been initiated in Istanbul, Turkey in order to evaluate the effects of highway noise on annoyance and myocardial infarction. A total of 2000 surveys will be conducted in 6 hospitals in Istanbul. The highway noise levels are obtained from the noise maps prepared by Environmental Protection Department of Istanbul Metropolitan Municipality. This manuscript presents the highway noise exposure levels in Istanbul and early findings regarding the annoyance effects of highway traffic noise.

3:50

2pNS9. Multilevel modeling of recent community noise annoyance surveys. Maurice E. Hayward (Mathematics, College of Williams & Mary, 200 Ukrop Way, College of William & Mary Mathematics Dept., Williamsburg, VA 23185, mehayward@email.wm.edu), Jonathan Rathsam (NASA Langley Res. Ctr., Hampton, VA), D. K. Wilson (Cold Regions Res. Eng. Lab., U.S. Army Engineer Res. Dev. Ctr., Hanover, NH), Edward T. Nykaza (ERDC-CERL, US Army Corps of Engineers, Champaign, IL), and Nicole M. Wayant (Geospatial Res. Lab., U.S. Army Engineer Res. Dev. Ctr., Alexandria, VA)

Large scatter in community noise annoyance survey data has recently been linked to systematic variations in noise tolerance. We hypothesize that noise tolerance is related to secondary factors, including environmental (e.g., population density, ambient noise) and attitudinal (e.g., noise sensitivity). If noise tolerance and secondary factors can be linked, the secondary factors provide a means to generalize noise annoyance from surveyed communities to non-surveyed communities. This approach would be useful for generalizing annoyance caused by quiet supersonic flight or military blast noise from a limited number of surveyed communities to other locations. A simple multilevel modeling approach (D.K. Wilson, *et al.* "Community noise annoyance: Connecting regression methodologies and theoretical models," J. Acoust. Soc. Am. **139**(4), 1983 (2016)) offers a useful framework to study the relationship between noise tolerance and secondary factors. The framework has already been applied to historic survey data on transportation noise annoyance. However, the historic surveys contain limited secondary factor information. This presentation applies the multilevel modeling approach to two recent community surveys about which more secondary factors are known. The first survey was conducted on a community exposed to quiet sonic booms. The second survey was conducted on multiple communities exposed to military blast noise.

4:05

2pNS10. Individual differences in judging the annoyance of the sounds of helicopters. George A. Luz (Luz Social & Environ. Assoc., Inc., 288 Ninth St. Alley, Ashland, OR 97580, luz_associates@msn.com) and Nicholas P. Miller (HMMH, Burlington, MA)

A rarely used procedure for studying aviation noise annoyance is asking people ro rate indiviidual aircraft events experienced in their homes. Our study employs this procedure with five subjects who self-identified as "highly annoyed" by helicopters. Subjects lived within 400 m of each other. Flights were recorded at three homes over a 10 day survey period during which subjects registered their judgments on a cell phone app with which a subject initiated a survey by tapping "hear a helicopter." Questions included location (indoors/outdoors), window position (open/closed), annoyance (5 pt scale), loudness (3 pt scale), and house rattles (3 pt scale). Recordings corresponding to completed surveys were analyzed for ASEL, CSEL, subjective and objective duration, loudness, loudness level, fluctuation strength, and roughness. Stepwise regression was used to find the most robust predictors of annoyance. The data confirmed the importance of ASEL and/or measures of loudness as predictors. Compared with in-home observers in a 1983 study at Oklahoma City Airport, these subjects showed a sharp increase in high annoyance when the SEL exceeded 90 dBA.

4:20

2pNS11. Three dimensional noise mapping system with aerial blimp robot. Ryouzi Saitou (Dept. of Intermedia Art and Sci., Waseda Univ., Minamishitaura, 1617-3 308, Miura, Kanagawa 238-0103, Japan, ryouzi5136-saitou@fuji.waseda.jp), Yusuke Ikeda, and Yasuhiro Oikawa (Dept. of Intermedia Art and Sci., Waseda Univ., Tokyo, Japan)

Recently, Unmanned Aerial Vehicle (UAV) is used in various situations and research fields, because of low cost and high convenience. For instance, there are many measurement and monitoring systems for taking environmental information by analyzing the image data which is acquired from the air with the UAVs. The UAVs with some propellers, such as quadcopter, always needs to rotate the propellers during flying. The noises of rotating propellers and makes collecting sound information with UAV difficult. In this study, we proposed the environmental sound recording system using a blimp which is filled with helium gas. The system has four propellers in the horizontal direction and two propellers in the vertical direction. This system has also a control board which has a SoC (including both FPGA and CPU) with embedded Linux OS to control six DC motors attaching to the each propeller and to communicate with PC through wireless LAN. The control board can record a sound signal with an A/D converter and an external memory. The experimental results show that it is efficient to make noise maps in horizontal and vertical planes by using the proposed system.

4:35

2pNS12. Probability of receiving a blast noise complaint based on population density. Dawn A. Morrison (US Army Corps of Engineers, ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61826, dawn.a.morrison@usace.army.mil), Jose M. Gonzalez (Statistics, Univ. of Illinois, Urbana-Champaign, Champaign, IL), and Edward T. Nykaza (US Army Corps of Engineers, ERDC-CERL, Champaign, IL)

This study extends previous work done by Nykaza et al. (2012) on the probability of receiving a blast noise complaint to include population density in the analysis. Nykaza et al. specifically asked, "what is the probability of receiving a blast noise complaint on a given day within 5 km of a noise monitor?" The original study examined 102,116 blast events and 148 complaints recorded over 641 study days around one military installation. Results from the original study found that while the overall probability of receiving a blast noise complaint was very low, blast noise complaint tolerances around individual noise monitors varied significantly. The current study combines dasymetric mapping techniques—an overlay method for redistributing arbitrarily organized data based on real world circumstances—and regression analysis to examine the different complaint tolerances, and explores what role, if any, characteristics of surrounding population may play in determining the probability of receiving a blast noise complaint. The results indicate that population density has a direct relationship with blast noise complaint tolerance and suggest that population density may be used as a proxy variable in noise complaint forecasting models.

4:50

2pNS13. Influence of visual information presentation method and sound presentation method on subjective evaluation of boat traffic noise. Suguru Hosono, Kenji Muto (Graduate School of Eng. and Sci., Shibaura Inst. of Technol., 3-7-5 Toyosu, Shibaura Inst. of Technol., Muto lab., Koto-ku, Tokyo 135-8548, Japan, ma15072@shibaura-it.ac.jp), and Yasunobu Tokunaga (National Inst. of Technol., Maizuru College, Kyoto, Japan)

Information on the acoustic environment such as the engine noise of cruising boat on the canal is necessary for town planning. Miyagawa has investigated the influence of the visual of a sound source on an impression of the sound. The study has evaluated the shift on the impression of the sound by the visual stimulus. We focus on human subjective evaluation of the noise stimulus of moving source in this study. Our objective is to clarity the influence of visual information on the subjective evaluation of one moving boat noise. In this experiment, we examined the subjective evaluation of

the moving source comparison between the stimulus of the sound and the stimulus of the sound and visual in the case of moving boat noise. This experiment used a 24-inch monitor, a headphone, or loudspeakers. The sound pressure level of 42–70 dB. As the result, the value of subjective evaluation was decreased by the moving source of loud boat noise, in the case of visual information of the moving boat.

TUESDAY AFTERNOON, 29 NOVEMBER 2016 SOUTH PACIFIC 2, 1:15 P.M. TO 5:00 P.M.

Session 2pPA

Physical Acoustics and Noise: Acoustics of Supersonic Jets: Launch Vehicle and Military Jet Acoustics II

Seiji Tsutsumi, Cochair
JEDI center, JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara 252-5210, Japan

Kent L. Gee, Cochair
Brigham Young University, N243 ESC, Provo, UT 84602

Alan T. Wall, Cochair
Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, OH 45433

Chair's Introduction—1:15

Invited Papers

1:20

2pPA1. Liftoff and time equivalent duration data evaluation of exploration flight test 1 Orion multi-purpose crew vehicle. Janice Houston (Fluid Dynam., NASA Marshall Space Flight Ctr., Bldg. 4203, Huntsville, AL 35812, janice.d.houston@nasa.gov)

The liftoff phase induces high acoustic loading over a broad frequency range for a launch vehicle. These external acoustic environments are used in the prediction of the internal vibration responses of the vehicle and components. There arises the question about time equivalent (Teq) duration of the liftoff phase and similarity to other launch vehicles. Vibroacoustic engineers require the fatigue-weighted time duration values for qualification testing inputs. In order to determine the Teq for the Space Launch System, NASA's newest launch vehicle, the external microphone data from the Exploration Flight Test 1 (EFT-1) flight of the Orion Multi-Purpose Crew Vehicle (MPCV) was evaluated. During that evaluation, a trend was observed in the data, and the origin of that trend is discussed in this paper. Finally, the Teq values for the EFT-1 Orion MPCV are presented.

1:40

2pPA2. Development of aero-vibro acoustics methods for predicting acoustic environment inside payload fairing at lift-off. Seiji Tsutsumi, Shinichi Maruyama (JEDI Ctr., JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa 252-5210, Japan, tsutsumi.seiji@jaxa.jp), Wataru Sarae, Keita Terashima (JAXA, Tsukuba, Japan), Tetsuo Hiraiwa (JAXA, Kakuda, Japan), and Tatsuya Ishii (JAXA, Chofu, Japan)

Prediction of harmful acoustic loading to payloads inside launcher fairing due to intense acoustic wave generated from propulsion systems at lift-off is an important design issue. Aero-vibro acoustics method developed in this study for predicting the acoustic loading consists of four elements. Hydrodynamics of jet flowfield generating aeroacoustic wave is computed by the high-fidelity Large-Eddy simulation. Computational aeroacoustics based on the full Euler equations is conducted to simulate propagating acoustic wave from the jet to the payload fairing. Then, finite element method is applied to simulate the structural vibration that radiates acoustic wave inside the fairing. Finally, acoustic behavior inside the payload fairing is also computed by the finite element method. An overview of the methods and recent work for validation and verification will be presented.

2:00

2pPA3. Large eddy simulations of acoustic waves generated from hot and cold supersonic jets at Mach 2.0. Hiroaki Nakano (Mech. Eng., Tokyo Univ. of Sci., 6-3-1,Nijuku, Katsushika, Tokyo 125-8585, Japan, nakano@flab.isas.jaxa.jp), Taku Nonomura, Akira Oyama (Inst. of Space and Astronautics Sci., JAXA, Sagamihara, Japan), Hiroya Mamori, Naoya Fukushima, and Makoto Yamamoto (Mech. Eng., Tokyo Univ. of Sci., Katsushika, Japan)

Strong acoustic waves are generated from the rocket plumes in the supersonic jet condition. The acoustic waves are significantly affected by temperature of jets. In this study, we perform large eddy simulations of the acoustic waves generated from the hot and cold jets. The temperature ratio of chamber to ambient air is set to be 4.0 and 1.0 for the hot and cold jets, respectively. Mach waves are radiated from the region close to the nozzle exit. In the hot jet case, we confirmed that the shorter potential core length, the larger angle of Mach waves, and the higher sound pressure level, as compared with those in the cold jet case.

2:20

2pPA4. Quantitative evaluation of the acoustic waves generated by a supersonic impinging jet. Daisuke Kato (Aeronautics and Astronautics, Univ. of Tokyo, Yoshinodai, Sagamihara, Kanagawa 2520206, Japan, dkato@flab.isas.jaxa.jp), Taku Nonomura, and Akira Oyama (Japan Aerosp. Exploration Agency, Sagamihara, Japan)

Strong acoustic waves are generated when rocket launch. The accurate prediction of these acoustic waves are important for the design of the rocket launch site because the acoustic waves possibly damage the payload. However, it is difficult for numerical simulation to predict accurately because the numerical simulation overestimatesthe strength of acoustic wave by several decibels compared to that observed in the experiment. The objective the present study is to obtain the knowledge for quantitative evaluation of the acoustic waves generated from supersonic impinging jet. According to the free jet studies, the reason for difference between numerical simulations and experiments in acoustic field is considered to be turbulence intensity in nozzle exit. Thus, the flow inside the nozzle is calculated, together with the turbulent disturbance is added to the boundary layer. In the numerical simulation, three-dimensional compressible Navier-Stokes equations are solved with the high resolution shock capturing scheme (WCNS). The effects of the acoustic field on nozzle internal inflow condition of supersonic impinging jet are maninly discussed.

2:40

2pPA5. Modeling the generation of supersonic turbulent jet noise by large-scale coherent structures. Oliver T. Schmidt, Tim Colonius (MCE, California Inst. of Technol., 1200 E California Blvd., MC 104-44, Pasadena, CA 91125, oschmidt@caltech.edu), and Guillaume A. Brès (Cascade Technologies, Inc., Palo Alto, CA)

Large-scale coherent structures, or wavepackets, are a salient feature of turbulent jets, and the main source of jet mixing noise at aft angles. They are extracted from a high-fidelity Mach 1.5 LES database as spectral POD mode estimates. These most energetic wavepackets obtained via POD and their acoustic far-field radiation patterns are compared to solution to the one-way Euler (OWE) equations recently introduced by Towne & Colonius (AIAA Paper 2013-2171, 2013; AIAA Paper 2014-2903, 2014). Within the OWE framework, the linearized Euler equations are modified such that all upstream propagating acoustic wave components are removed from the solution. The resulting spatial initial value problem can be solved in a stable and computationally efficient manner by downstream marching the solution. Additionally, the scenario of stochastic forcing of wavepackets by the surrounding turbulence is considered in a resolvent analysis. The resolvent analysis allows for the computation of optimal forcing distributions and corresponding responses. It is based on a singular value decomposition of the transfer function of the governing linear operator. The results of the both methods, OWE and resolvent analysis, are compared to the most energetic POD modes with a special focus on far-field radiation patterns and computational efficiency.

Contributed Papers

3:00

2pPA6. Self-similarity of level-based wavepacket modeling of high-performance military aircraft noise. Tracianne B. Neilsen, Kent L. Gee, Blaine M. Harker (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Construction of the equivalent acoustic source model for high-performance military aircraft noise presented in this paper begins with a decomposition of sound levels, from ground-based microphones 11.7 m from a high-performance military aircraft, into portions matching the similarity spectra associated with large-scale and fine-scale turbulent mixing noise (LSS and FSS). [Tam *et al*. AIAA Paper 96-1716 (1996)]. A Fourier transform of the spatial distribution of the decomposed levels yields frequency-dependent wavenumber spectra [Morris, Int. J. Aeroacoust. **8**, 301-316 (2009)]. Each LSS-educed wavenumber spectrum corresponds to a wavepacket, consisting of a spatially varying amplitude distribution with a constant axial phase relationship. This wavepacket model produces a directional, coherent sound field. However, the asymmetry in the LSS-educed wavenumber spectra may indicate that a nonuniform phase relationship is a more physical choice that could be correlated to the axial variation in convective speed. The FSS component is modeled with a line of incoherent monopoles whose amplitude distribution is related to the FSS-educed wavenumber spectrum. While this level-based model is limited, the addition of the FSS-related source yields a better match to the noise environment of a high-performance military aircraft than is found using a single coherent wavepacket. [Work supported by ONR.]

3:15

2pPA7. Wavepacket source modeling of high-performance military aircraft jet noise from cross-beamforming analysis. Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen (Dept. Phys. & Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@gmail.com), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Wavepacket models provide a convenient representation of jet noise source phenomena because of the extended, partially correlated nature of the turbulent mixing noise. When treated as an equivalent source model, they are useful to estimate features of both the radiated noise as well as the source characteristics to assist in jet noise reduction efforts. In this study, advanced cross-beamforming techniques are applied to measurements in the vicinity of a high-performance military aircraft. These results are then decomposed into an azimuthally-averaged multi-wavepacket representation of the data, which can then be treated as an equivalent source. Estimates of

the field levels and coherence properties using the equivalent source are compared with measurements, and results from the multi-wavepacket model show good agreement with benchmark measurements over a range of frequencies that contribute significantly to the overall radiation. The capabilities and limitations of the model to estimate field properties are quantified with respect to benchmark measurements in the mid and far fields as a function of frequency. Results indicate that the multi-wavepacket representation is an improvement over single-wavepacket models, which do not incorporate spatiotemporal features of the radiation. [Work supported by ONR and USAFRL through ORISE.]

3:30–3:45 Break

3:45

2pPA8. Azimuthal coherence of the sound field in the vicinity of a high performance military aircraft. Kevin M. Leete (Phys. and Astronomy, Brigham Young Univ., Provo, UT 84604, kevinmatthewleete@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, U.S. Air Force Rersearch Lab., Dayton, OH), Kent L. Gee, Tracianne B. Neilsen, Blaine M. Harker (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Res. and Consulting LLC, Asheville, NC)

Mixing noise from a jet engine originates from an extended spatial region downstream of the nozzle and is partially correlated both spatially and temporally. Previously, the coherence properties in the downstream (axial) direction of the sound field of a tethered military aircraft were investigated, resulting in the identification of different spatial regions based on coherence length [B. M. Harker *et al.*, AIAA J. **54**, 1551-1566 (2016)]. In this study, a vertical array of microphones to the side of the jet plume is used to obtain the azimuthal coherence of the sound field. Although multipath interference effects and a limited angular aperture make coherence length calculation impossible, information about upper and lower bounds can be extracted. The measured azimuthal coherence as a function of downstream distance and frequency is then compared to that predicted by sound field reconstructions using multisource, statistically optimized near-field acoustical holography (M-SONAH) [A. T. Wall *et al.*, J. Acoust. Soc. Am., **139**, 1938-1950 (2016)]. This comparison helps to benchmark the performance of a reduced-order M-SONAH algorithm that employs only axisymmetric cylindrical basis functions to represent the direct and image sound fields. [Work supported by USAFRL through ORISE (2016).]

4:00

2pPA9. In defense of the Morfey-Howell Q/S as a single-point nonlinearity indicator: An impedance-based interpretation. Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, ohm@yonsei.ac.kr), Kent L. Gee, and Brent O. Reichman (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Since the Morfey-Howell Q/S was proposed as a nonlinearity indicator for propagation of intense broadband noise [AIAA J. **19**, 986-992 (1981)], there has been considerable debate as to its meaning and utility. Perhaps the most contentious argument against Q/S is about its validity as a *single-point* nonlinearity indicator: the importance of nonlinearity is often judged by observing cumulative effects over some propagation distance, whereas Q/S is based on a pressure waveform at a single location. Studies to address these criticisms have emerged over the years, most recently by Reichman et al. [J. Acoust. Soc. Am. **139**, 2505-2513 (2016)] in support of Q/S. In this talk we show that the Burgers equation (from which the Q/S was originally derived) can be recast in terms of specific impedance, linear absorption and dispersion coefficients, and normalized quadspectral (Q/S) and cospectral (C/S) densities. The resulting interpretation is that Q/S and C/S represent the additional absorption and dispersion, introduced by the passage of a finite-amplitude wave to the existing linear absorption and dispersion. In other words, a nonlinear wave process alters the apparent material properties of the medium, the extent of which can be used as a single-point indicator of the relative strength of nonlinearity.

4:15

2pPA10. Near-field array measurements of F-35 aircraft noise. Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Kent L. Gee, Tracianne B. Neilsen, Kevin M. Leete (Brigham Young Univ., Provo, UT), Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC), and Richard L. McKinley (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson Air Force Base, OH)

Noise source measurement and modeling efforts are being conducted for current and future generations of high-performance military aircraft. A near-field microphone array was used in a noise measurement of F-35A and F-35B aircraft at Edwards Air Force Base in 2013 in order to extract source models. The two aircraft measured were tethered to the ground on a concrete run-up pad. The array consisted of 76 sensors, spanned 32 m, and was placed on the ground approximately 7 m from the jet plume. The engines operated over a full range of power settings from idle through full afterburner. In preparation for the execution of source modeling efforts, these near-field data are explored and compared to far-field data published previously [James *et al*., AIAA Paper 2015-2375]. [Work supported by USAFRL through F-35 JPO.]

4:30

2pPA11. Noise measurements of the F-35C on an aircraft carrier flight deck. Nourhan K. Abouzahra, Alan T. Wall, Richard L. McKinley, Billy J. Swayne (Air Force Res. Lab., 2610 Seventh St. Bldg. 441, Wright-Patterson AFB, OH 45433, nourhan.abouzahra.ctr@us.af.mil), Michael J. Smith, and Allan C. Aubert (NAVAIR, Patuxent River Naval Air Station, MD)

In 2015, scientists from the Air Force Research Laboratory (AFRL) in collaboration with other teams performed noise measurements of the F-35C aircraft aboard a Nimitz-class carrier. The purpose of the measurements was to quantify the crewmembers' exposure to noise. The measurements were taken using hand-held noise recorder systems, and the recording engineers shadowed actual locations of crew. The near-field noise levels caused by aircraft are reported in the areas experienced by crew on board aircraft carriers. High noise levels can interfere with communications and may pose a risk for hearing loss. These data represent a unique measurement; the previous measurement of aircraft noise from catapult launches and wire arrestments aboard a carrier occurred more than 15 years ago.

4:45

2pPA12. Nonlinear characteristics of F-35 flyover waveforms. Brent O. Reichman, Kent L. Gee, Tracianne B. Neilsen (Brigham Young Univ., 453 E 1980 N, #B, Provo, UT 84604, brent.reichman@byu.edu), and Sally A. McInerny (Univ. of Louisiana at Lafayette, Lafayette, LA)

Metrics indicating the presence of acoustic shocks are calculated from noise measurements of F-35 aircraft during flyover operations. The sound pressure level, average steepening factor, and skewness of the first time derivative of the pressure waveform are calculated for the A and B variants of the aircraft. Other metrics more specific to flyover measurements are also shown. Comparisons of nonlinear indicators are made between engine thrust conditions, aircraft type, microphone height, microphone size, and sampling rate. Nonlinear indicators are also compared against other full-scale military aircraft. Comparisons are made with similar conditions during ground run-up operations.

Session 2pPP

Psychological and Physiological Acoustics: Binaural Processing and Localization

Christopher Brown, Chair
University of Pittsburgh, 4033 Forbes Tower, 3600 Forbes at Atwood, Pittsburgh, PA 15260

Contributed Papers

1:15

2pPP1. Coherence influences sound-field localization dominance in a precedence paradigm. Julian Grosse, 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), and Constantine Trahiotis (Departments of Neurosci. and Surgery (Otolaryngology), Univ. of Connecticut Health Ctr., Farmington, CT)

The ability to localize sound sources in reverberant environments is dependent upon first-arriving information, an outcome commonly termed “the precedence effect.” For example, a combination of a leading (direct) sound followed by a lagging (reflected) sound is localized in the direction of the leading sound. We measured how the compactness/diffuseness (i.e., the interaural coherence) of leading and lagging sounds, respectively, influences performance. The compactness/diffuseness of leading or lagging sounds was varied by either presenting a sound from a single loudspeaker or by presenting mutually uncorrelated versions of similar sounds from nine adjacent loudspeakers. The listener pointed to the perceived location of leading and lagging sources of sounds which were 10-ms long, low-pass filtered white noises or 2-second long tokens of speech. The sounds were presented either from speakers located directly in front of the listener or from speakers located 45° to the right or left. Leading compact/coherent sounds, be they Gaussian noises or tokens of speech, influenced perceived location more so than did leading diffuse/incoherent sounds. The results strongly support the hypothesis that stimulus coherence, often modeled by “straightness” within a cross-correlogram, plays an important role in precedence.

1:30

2pPP2. Better-ear glimpsing with symmetrically-placed interferers in bilateral cochlear implant users. Hongmei Hu (Medizinische Physik and Cluster of Excellence Hearing4all, Universitat Oldenburg, Carl von Ossietzky-Universitat Oldenburg, Oldenburg 26129, Germany, hongmei.hu@uni-oldenburg.de), Mathias Dietz (National Ctr. for Audiol., Western Univ., London, ON, Canada), and Stephan D. Ewert (Medizinische Physik and Cluster of Excellence Hearing4all, Universitat Oldenburg, Oldenburg, Germany)

For a frontal speaker in spatially symmetrically placed maskers, normal hearing (NH) listeners can use an optimal “better-ear glimpsing” strategy selecting time-frequency segments with favorable signal-to-noise ratio in either ear. It was shown that for a monaural signal, obtained by an ideal monaural better-ear mask (IMBM), NH listeners can reach similar performance as in the binaural condition, but interaural phase differences at low frequencies can further improve performance. In principle, bilateral cochlear implant (BiCI) users could use a glimpsing strategy; however, they cannot exploit (temporal fine structure) interaural phase differences. Here, speech reception thresholds of NH and BiCI listeners were measured in two symmetric maskers ($\pm 60°$; speech-shaped stationaty noise, non-sense speech, single talker) using head-related transfer functions and headphone presentation or direct stimulation in BiCI listeners. Furthermore, a statistically independent masker in each ear, diotic presentation with IMBM processing, and a noise vocoder based on the BiCI electrodogram was used in NH. Results indicate that NH with vocoder and BiCI listeners show a strongly reduced binaural benefit in the $\pm 60°$ condition relative to the co-located condition when compared to NH. However, both groups greatly benefit from IMBM processing (as part of the CI stimulation strategy). Individual differences are compared.

1:45

2pPP3. Speech intelligibility by bilateral cochlear implant users when interaural level cues are made more consistent across frequency. Christopher Brown (Univ. of Pittsburgh, 4033 Forbes Tower, 3600 Forbes at Atwood, Pittsburgh, PA 15260, cbrown1@pitt.edu)

Interaural level differences (ILDs) aid speech understanding for listeners with both normal hearing (NH) and bilateral cochlear implants (BCIs). Naturally-occurring ILDs are less than ideal for this outcome; however, as they are generally restricted to frequencies above about 2000 Hz, while the energy in speech is typically concentrated below 2 kHz. It has recently been demonstrated that applying larger-than-normal ILDs in the low-frequency region can significantly improve speech intelligibility for BCI users. Despite the benefit, this may result in sound sources that appear to move, depending on where in frequency the speech energy resides at a given moment. The current study examined the effect of applying larger-than-normal ILDs both above and below 2 kHz. Speech intelligibility was measured in simulation by 16 individuals with NH, and by eight BCI users, of a target talker along the midsagittal plane in the presence of symmetrically placed masker talkers on either side. An improvement of about 20 percentage points was observed at a number of masker locations for both populations tested when larger-than-normal ILDs were extended into the high-frequency region, although perceptual movement of sound sources appeared to be more noticeable for the NH group. [Work supported by the NIDCD.]

2:00

2pPP4. The benefits of cochlear implants for single-sided deafness for sound localization with multiple concurrent sources. Joshua G. Bernstein, Gerald I. Schuchman, Olga A. Stakhovskaya (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., 4954 N. Palmer Rd., Bethesda, MD 20889, joshua.g.bernstein.civ@mail.mil), Arnaldo L. Rivera (Otolaryngol. - Head and Neck Surgery, Univ. of Missouri School of Medicine, Columbia, MO), and Douglas S. Brungart (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

Individuals with single-sided deafness (SSD) who receive a cochlear implant (CI) realize some benefit for localization and speech perception in noise. Previous studies of SSD-CIs examined simple conditions involving speech perception in the presence of a single noise masker or single-source localization. However, some of the most important benefits of two ears arise in multisource situations where binaural cues and head turning can facilitate the perceptual separation of spatial separated sources. Eight SSD-CI listeners performed a sound-localization task in a multisource environment pre-implantation and at least 3 months post-implantation. Environmental sounds were presented from a spherical loudspeaker array. The target sound was presented alone, or concurrently with one or three additional sources. The

2p TUE. PM

target was cued by being added to or removed from the mixture after a 6-s delay. A head-mounted tracker monitored head movements during the task. The CI substantially improved localization accuracy, reducing the mean azimuthal error by as much as 34%, while reducing the amount of head turning by as much as 44%, even during periods when the target sound was not present. For SSD-CI listeners, bilateral auditory input can improve environmental-sound localization in a complex mixture, while reducing the amount of head-turning effort required. [The opinions and assertions presented are the private views of the authors and are not to be construed as official or as necessarily reflecting the views of the Department of Defense.]

2:15

2pPP5. The impact of cochlear implantation on spatial hearing and listening effort. Ruth Litovsky, Sara Misurelli, Shelly Godar, Tanvi Thakkar, Alan Kan (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Waisman Ctr. Rm. 521, Madison, WI 53705, litovsky@waisman.wisc.edu), and Matthew Winn (Speech and Hearing, Univ. of Washington, Seattle, WA)

We studied the potential benefits of bilateral hearing in cochlear implants (CI) users, and in patients with single-sided deafness (SSD) who receive a CI in the deaf ear and have normal hearing in the other ear. We hypothesized that listening effort, measured with pupil dilation, can reveal benefits of bilateral hearing that may not be consistently observed when localization or spatial release from masking (SRM) are measured. Result from 12 bilateral CI users showed reduction in listening effort with bilateral hearing compared to the poor ear or better ear alone. In patients with SSD, benefits of adding a CI to a normal hearing ear can emerge over a protracted period of a year or longer. In addition, for at least some of the subjects bilateral hearing (adding a CI to the normal hearing ear) produced release from listening effort even in conditions where SRM was not observed. That is, speech intelligibility did not always improve with spatial separation of target and competing speech, but pupil dilation was reduced, suggesting that pupillometry can reveal subtle aspects of bilateral benefits. The benefits shown with pupillometry are consistent with the patients' subjective reports of improved ability to function in complex, noisy acoustic environments.

2:30

2pPP6. Uncertainty in binaural hearing linked to inherent envelope fluctuations. Gabrielle O'Brien and Matthew Winn (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, andronovhopf@gmail.com)

It was recently shown that supra-threshold changes in interaural level differences (ILDs) in 1/3-octave bands of noise were more quickly and reliably perceived when carried by high-frequency (4000 Hz) rather than low-frequency (500 Hz) noise. For fixed-octave ratio noises, however, center frequency differences are accompanied by differences in inherent envelope modulations stemming from differences in absolute bandwidth. We propose that envelope fluctuations (rather than frequency differences) are a more parsimonious explanation of response latency and certainty. In the current study we investigate the hypothesis that for noisier envelopes, there is less certainty that an ILD is a true diversion from noise perceived at center. Differences in inherent envelope modulation strength were controlled using three types of stimuli: low-noise noise (with negligible envelope fluctuations), high-noise noise, and an equal mixture of the two. Perception of ILD changes was measured using anticipatory eye movements elicited by ILD changes. Results confirm that sensitivity and response latency to ILD changes were affected by degree of envelope fluctuation. In addition, high-noise noise provoked more phantom percepts, which were nearly absent for low-noise noise. Finally, we propose a mathematical model of accumulating certainty in this task as a function of time and degree of envelope fluctuation.

2:45

2pPP7. Front-back sound source localization confusions when sources and listeners are rotated. William Yost (ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Sound source localization identification judgments were made in the azimuth plane with the goal of determining front-back and back-front confusions. The sounds were filtered 125-ms noise bursts: two and 1/10th octave wide at center frequencies of 500 and 4000 Hz, and a wideband (125-8000 Hz) condition. Sounds were either presented from one of six evenly spaced (60^o) random loudspeaker locations (random) or the sound presentation rotated clockwise (rotating) around the six loudspeaker array except on 20% of the trials when the location of the loudspeaker was randomly determined. Four conditions were tested: Listener Stationary with Eyes Open (L-O), Listener Stationary with Eyes Closed (L-C), Listener Rotating with Eyes Open (R-O), and Listener Rotating with Eyes Closed (R-C). Listener rotation was clockwise at 45^o/s. Localization accuracy and the type of errors, especially front-back and back-front errors, were determined for 18 normal hearing listeners. Sound source localization accuracy and the number of front-back and back-front confusions varied as a function of the different filtered noises, the type of sound source presentation (random or rotating) and the listening condition (L-O,L-C,R-O,R-C). There were very large individual differences, especially in front-back and back-front confusions. [Work supported by Grants from AFOSR and NIH.]

3:00–3:20 Break

3:20

2pPP8. Audiometrically-defined "slight" or "hidden" hearing losses can be manifested as changes in binaural detection. Leslie R. Bernstein and Constantine Trahiotis (UConn Health Ctr., UConn Health Ctr., MC3405, 263 Farmington Ave., Farmington, CT 06030, lbernstein@uchc.edu)

We hypothesized that audiometrically-defined "slight" or "hidden" hearing losses might be associated with losses in binaural detection via degradations in precision of coding of interaural temporal disparities (ITDs). Thirty-one listeners were tested using masking configurations that, taken together, allow one to describe how precision of ITD-coding varies as a function of reference ITD and center frequency. Binaural detection thresholds at 500 Hz and 4 kHz were found to increase magnitude of ITD, consistent with there being a loss of precision of ITD-coding that increases with magnitude of ITD. Binaural detection thresholds were elevated, across all ITDs, for those listeners whose absolute thresholds at 4 kHz exceeded 7.5 dB HL, but were not greater than 23 dB HL. No such elevations were observed in conditions having no binaural cues available to aid detection (i.e., "monaural" conditions). Partitioning and analyses of the data revealed the elevated thresholds: 1) were more attributable to hearing level than to age; 2) result from increased levels of internal binaural-processing noise. Our conclusion is that listeners whose high-frequency monaural hearing status would be classified audiometrically as being normal or "slight loss" could still suffer from perceptually meaningful losses of binaural processing.

3:35

2pPP9. Comparison between individualization methods of spatialization for bone conducted sound in navigation task for walking subjects. Akiko Fujise (Panasonic Corp., Tokyo Fashion Town West, 3-4-10 Ariake, Koto-ku, Japan, fujise.akiko@jp.panasonic.com) and Joseph A. Paradiso (MIT Media Lab, Cambridge, MA)

Recent work in human-machine interaction and audiology has indicated the effectiveness of using spatialized bone conduction (BC) hearing in augmented reality (AR) environments. For broader applications, individualization methods are preferred, as they require simpler computation and measurement, despite the fact that spatial hearing for BC is not yet well investigated at individual levels. Accordingly, this study examines different individualization techniques for spatial audio using BC applied to a navigation task with a moving virtual sound object in external noise, which simulates a common situation in AR environments. The tested methods include one based on individually measured head-related transfer functions (HRTFs), generic HRTFs, and a modified version based on these. Subjects are instructed to follow a virtual sound source presented from the BC headphones while walking. The virtual sound source is updated so that the source is located ahead of the subject according to the subject's location and head rotation tracked by the motion capture system. The subjects' performance are evaluated as the error of the subjects' actual path from the simulated pathway. Results are discussed on the comparison between individualization methods and in regards to recent reports on sound localization in BC hearing without the listener's body movement.

3:50

2pPP10. Spatial and temporal competitors bias auditory localization of impulse events. Jeremy R. Gaston (US Army Res. Lab., Dept. of the Army, Attn: RDRL-HRS, Aberdeen Proving Ground, MD 21005, jgaston2@gmail.com), Kelly Dickerson, Ashley Foots, Timothy Mermagen, Mark Ericson, and Brandon Perelman (US Army Res. Lab., APG, MD)

Sound events in the real-world rarely occur in isolation and when competing sounds are present, the effect on perception can be significant (e.g., various forms of masking). The present study examines auditory localization for impulse sounds modeled after the real-world sound event of small-arms fire. Small-arms fire consists of two distinct sounds, the muzzle blast and ballistic crack. Across different shooter-observer positions, there can be substantial offsets in spatial and temporal stimulus relationships between the sounds. To examine these relationships, impulse targets were presented across a loudspeaker array in the context of impulse competitors varying across timing and spatial relationships. Results showed that there was a significant increase in absolute localization error as a function of the magnitude of these relationships, and signed error indicated significant bias toward the direction of the competing impulse. These results suggest that listeners integrated information from the competing stimulus into their localization decision.

4:05

2pPP11. Reverberation enhances the relative potency of onset cues for sound localization. G. C. Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu), Asawer Nihal (Loyola Univ. Chicago, Chicago, IL), Julie M. Stecker, and Travis M. Moore (Hearing and Speech Sci., Vanderbilt Univ., Nashville, TN)

Onset dominance in spatial hearing has been repeatedly demonstrated through studies of the precedence effect, Franssen effect, and temporal weighting in sound localization. Such effects are often cited as evidence that onset dominance enhances spatial perception, particularly in reverberant environments where post-onset cues are degraded by echoes and reverberation. Most studies, however, have presented sounds in anechoic space or via headphones. Thus, the actual effects of reverberation on onset dominance are not well understood. This study adapted the approach of Stecker & Hafter (2002; *JASA* 112:1046-57) to measure temporal weighting functions (TWF) for freefield localization of Gabor click trains (4 kHz; 200 Hz click rate) in the presence or absence of simulated reverberation. Reverberant conditions simulated a 10m x 10m room with modestly absorptive walls ($\alpha = 0.5$). For each potential source location, direct sound and all reflections up to 13th order were synthesized at correct azimuths, intensities, and delays. Anechoic conditions presented only direct sound (i.e., source locations were single loudspeakers). Anechoic TWFs revealed modest onset dominance consistent with Stecker & Hafter (2002). Reverberant TWFs exhibited significantly stronger onset dominance. The results suggest a greater role of onsets in real-world listening than previously estimated. [Work supported by NIH R01 DC011548.]

4:20

2pPP12. Interaural level difference processing as a function of frequency. Matthew J. Goupell and Beth Rosen (Hearing and Speech Sci., Univ. of Maryland, College Park, 0119E Lefrak Hall, MD 20742, goupell@umd.edu)

To localize in the azimuthal plane, humans utilize interaural time differences and interaural level differences (ILDs). For sounds presented in the free field at different azimuthal locations, ILDs generated by the head are complex functions that are frequency dependent and non-monotonic, and are thought to contribute towards sound localization primarily at frequencies above 1500 Hz. However, previous studies have shown that ILD sensitivity is relatively frequency independent for tones presented over headphones, even at frequencies below 1500 Hz. The purpose of this study was to carefully re-investigate the frequency dependency of ILD sensitivity in normal-hearing listeners. Ten normal-hearing listeners were presented ILDs using insert earphones at 31 frequencies between 100 and 11,000 Hz. A 20-dB range of level roving was used to limit listeners' ability to use monaural level cues to perform the task. On average, ILD sensitivity had several high and low sensitivity frequency regions, was best at 7000 Hz (1.5 dB) and worst at 1000 Hz (2.5 dB). Individuals also showed some unique frequency dependent patterns. These results suggest that ILD sensitivity is frequency dependent and neural models of ILD processing need to accommodate this dependency.

4:35

2pPP13. Sound image movement of approaching and retreating sounds. Tatsuya Hirahara and Shuhei Okada (Toyama Prefectural Univ., 5180 Kurokawa, Imizu 939-0398, Japan, hirahara@pu-toyama.ac.jp)

We investigated how listeners perceive sound image movements of moving sound sources horizontally approaching and retreating from a listener's head under three conditions: directly listening to approaching and retreating real sound sources, listening to binaurally recorded sounds with headphones, and listening to binaurally synthesized sounds with headphones. White noise was emanated from a small loudspeaker put on a slider. The moving directions of the loudspeaker were eight radial horizontal directions at 45° intervals, and the moving distance was 5 to 105 cm from the head. Personal earplug microphones were used for the binaural recording. Personal head-related transfer functions were used to synthesize the binaural signals. Sounds approaching and retreating from listeners' heads in most radial horizontal directions produced straight approaching and retreating sound images. Those in the 45° and 315° radial horizontal directions, however, yielded anomalous sound image movement. The sound image moved straight forward, bent to the rear as the sound approached, and then followed the same path in reverse as the sound retreated for the real, binaurally recorded, and binaurally synthesized sounds. The analysis of HRTFs suggests that the growth of the inter-aural level difference very close to the head can be the cause of the anomalous sound image movement.

Session 2pSA

Structural Acoustics and Vibration and Physical Acoustics: Acoustic Metamaterials II

Michael R. Haberman, Cochair
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Christina J. Naify, Cochair
Acoustics, Naval Research Lab, 4555 Overlook Ave. SW, Washington, DC 20375

Jeffrey Cipolla, Cochair
Weidlinger Associates, Inc., 1825 K St. NW, Suite 350, Washington, DC 20006

Invited Papers

1:00

2pSA1. Superresolution using an acoustic vortex wave antenna. Matthew D. Guild (US Naval Res. Lab., Code 7160, 4555 Overlook Ave. SW, Washington, DC 20375, matthew.guild@nrl.navy.mil), Christina J. Naify (Jet Propulsion Lab., California Inst. of Tech., Pasadena, CA), Theodore P. Martin, Charles A. Rohde, and Gregory J. Orris (US Naval Res. Lab., Code 7160, Washington, DC)

Acoustic waves, like their electromagnetic counterpart, exhibit a diffraction limit, which restricts the ability to resolve or generate fine structures in the pressure field. Previously proposed methods to overcome this limit, and therefore achieve superresolution, have mainly been limited to operating within the near-field region of the aperture. In this work, we describe how far-field superresolution can be achieved using shaped acoustic vortices, created by utilizing the topological diversity of an acoustic vortex wave antenna. Theoretical and numerical results will be presented for the design of an acoustic vortex wave antenna with arbitrary planar arrangement which is capable of generating superresolved far-field features in the radiated acoustic pressure. Variation of the antenna geometry enables different shaped acoustic vortex patterns to be achieved, which propagate from the near-field into the far-field through an arrangement of stable integer mode vortices. Despite the total aperture size being less than a wavelength in diameter, the proposed acoustic vortex wave antenna is shown to achieve far-field superresolution with feature sizes 4-9 times smaller than the resolution limit. Several examples will be presented and discussed in detail. [Work supported by the Office of Naval Research.]

1:20

2pSA2. The response of an acoustic trace wavelength enhanced array to an isotropic noise field and optimum processing. Ivars P. Kirsteins and Benjamin A. Cray (NUWCDIVNPT, 1176 Howell St., Newport, RI 02841, ivars.kirsteins@navy.mil)

Recently, a new array based on generating Bragg-scattered acoustic wavelengths along the surface of an array was proposed (Enhanced directivity with array grating (J. Acoust. Soc. Am. **136** (2), 2014) and Experimental verification of acoustic trace wavelength enhancement (J. Acoust. Soc. Am. **138** (6), 2015)). The technique, denoted acoustic trace wavelength enhancement, relies on embedding periodic structures within an array, chosen to precisely replicate and shift the incident acoustic wavenumber into higher wavenumber regions. In principle, an important advantage of this array over conventional hydrophone-based arrays of the same size is its greatly improved directivity. A prototype array was built and demonstrated in a test tank under idealized high signal-to-noise ratio conditions. However, two important open questions which remain are how this array responds to ambient noise and how to optimally beamform it when multiple signals are present. Using idealized analytical models for the array's plane wave response, we calculate its spatial wavenumber response to an isotropic noise field and investigate instrumentations for optimum beamforming.

1:40

2pSA3. Constitutive laws for the design of metallic foam. Elizabeth A. Magliula (NAVSEA Newport, 1176 Howell St., Bldg. 1302, Newport, RI 02841, elizabeth.magliula@navy.mil), James G. McDaniel (Dept. of Mech. Eng., Boston Univ., Boston, MA), and Andrew S. Wixom (Appl. Res. Lab., The Penn State Univ., State College, PA)

Cellular materials have recently found attention in structural applications to buffer impacts and for energy absorption. In order to choose the most suitable cellular material based on its intended application, one must understand its mechanical response either through experimentation and/or modeling. Accurate modeling of the response of a material has many advantages, as it helps avoid extensive experimental laboratory tests and provides a fast and efficient avenue to explore the design space and optimize with a specific intent. The problem solved by the present research is as follows: How does one estimate the foam properties, such as porosity and void fraction, to produce a metallic foam with desired mechanical properties, such as Young's Modulus and Poisson's ratio? The solution to this problem is vitally important as it will yield an entirely new class of materials with tunable mechanical properties and dramatically improved shock and crack resistance. Therefore, the present work seeks to develop constitutive laws for the design of metallic foams. The resulting

semi-analytical micro-mechanical model is applicable to positive Poisson's ratio (PPR), negative Poisson's ratio (NPR) and zero Poisson's ratio (ZPR) metallic foams. It is developed quasi-analytically, implemented numerically, and compared to experimental data for validation.

Contributed Papers

2:00

2pSA4. Demonstration of acoustic holographic rendering with two-dimensional metamaterial-based passive phased array. Yangbo Xie (Elec. and Comput. Eng., Duke Univ., 3417 CIEMAS, Durham, NC 27705, yx35@duke.edu), Chen Shen, Yun Jing (Mech. and Aerosp. Eng., NC State Univ., Raleigh, NC), and Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC)

Acoustic holograms in analogy with optical holograms are useful for a variety of applications, such as multi-focal lensing, multiplexed sensing and synthesizing three-dimensional complex sound fields. We previously presented the designs and simulation results of several metamaterial-based acoustic holograms (*J. Acoust. Soc. Am. 138, 1751 (2015)*). Here, we demonstrate the experimental realization of two of them: one projects a complex pattern on the image plane, while the other focus energy onto multiple circular spots of different sizes. We showcase the holographic reconstruction measurements carried out in an anechoic chamber. The demonstrated passive holograms, without phase-shifting circuitry and transducer arrays, are advantageous for applications where simplicity, robustness, and small footprint are preferred over adaptive control. We also discuss the higher frequency versions of such holograms and their applications in robotic sensing and wireless power transfer. Such metamaterial-based holograms can serve as versatile platforms for various advanced acoustic wave manipulation and signal modulation, leading to new possibilities in acoustic sensing, imaging and energy deposition.

2:15

2pSA5. A helical-structured acoustic metamaterial. Jie Zhu (Hong Kong Polytechnic Univ., Hong Kong Polytechnic University, Kowloon 00000, Hong Kong, jiezhu@polyu.edu.hk) and Xuefeng Zhu (Huazhong Univ. of Sci. & Technol., Wuhan, China)

Here, we would like to present a helical-structured acoustic metamaterial that can effectively decelerate sound propagation. Unlike currently avaliable solutions which are mostly dispersive due to the requirement of local resonators., this non-dispersive helical-structured metamaterials is able to provide high effective refractive index. By adjusting the helicity of the acoustic metamaterial structures, we can further modify such refractive index. The performance of this helical-structured acoustic metamaterial has been numerically and experimentally verified. We even designed a lens base on multiple inhomogeneous helical-structured acoustic metamaterial unit cells. This lens can turn normally travelling plane wave into a self-accelerating beam.

2:30

2pSA6. New frontiers in elastic metamaterials for controlling surface waves. Andrea Colombi (Mathematics, Imperial College, South Kensington Campus, Huxley Bldg., London SW7 2AZ, United Kingdom, andree.colombi@gmail.com), Philippe Roux (ISTerre, Grenoble, France), Daniel Colquitt (Mathematical, Univ. of Liverpool, Liverpool, United Kingdom), Sebastien Guenneau (Institut Fresnel, Marseille, France), Victoriya Ageeva, Matt Clark (Optics and photonics, Univ. of Nottingham, Nottingham, United Kingdom), and Richard Craster (Mathematics, Imperial College, London, United Kingdom)

In this presentation, we show recent results obtained by our research group on two types of elastic metamaterial that are capable of controlling the propagation of surface elastic waves at various physical scales: from tens of meter in geophysics to millimeters in ultrasonic. We begin from the geophysical scale introducing a metamaterial that uses soft soil inclusions in the ground to create a lens for detouring seismic surface waves around an obstacle at frequency $<$10 Hz. We then move to the smaller scale presenting a metamaterial that exploits the local resonances in a subwavelength collection of resonators (vertical rods) attached to a finite (plate) or semi-infinite (halfspace) substrate to control either the A_0 mode in the plate or Rayleigh waves in the halfspace. We focus on the halfspace case and we illustrate how Rayleigh waves can be (1) mode converted into S-waves or stopped and trapped by a graded array of resonators (seismic rainbow); (2) focused, or detoured using a resonant metalens (Luneburg, Maxwell, and Eaton type) made with the vertical rods. The metamaterial are presented using results from time domain numerical simulations, laboratory and geophysical experiments to highlight the broad applicability of these objects across the wavelength spectrum.

2:45–3:00 Break

3:00

2pSA7. A new radial flexible acoustic metamaterial plate. Nansha Gao (School of Marine Sci. and Technol., Northwestern PolyTech. Univ., Xi'an, 710072, China; E-mail:gaonansha@hotmail.com., gaonansha@hotmail.com)

This paper presents the low frequency acoustic properties of a new proposed acoustic metamaterial, which is arranged under axial coordinate. Band structures, transmission spectra, and eigenmode displacement fields are different from previous acoustic metamaterial structures. Numerical calculation results show that first order band gap of radial flexible acoustic metamaterial plate is below 100 Hz. Multiple vibrations coupling mechanism is proposed to explain the low frequency band gaps. Radial flexible acoustic metamaterial plate can restrain low frequency vibration, which can potentially be applied to infrasound protection, generate filters, and design acoustic devices.

3:15

2pSA8. Experimental measurement of the Willis coupling coefficient in a one-dimensional system. Michael B. Muhlestein (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 3201 Duval Rd. #928, Austin, TX 78759, mimuhle@gmail.com), Caleb F. Sieck (Dept. of Elec. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson, and Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

The primary objective of acoustic metamaterial research is to design subwavelength systems that behave as effective materials with novel acoustical properties. Willis coupling is one such property. Initially described by J. R. Willis [Wave Motion **3**, pp. 1-11 (1981)], subwavelength structural asymmetry and/or nonlocal effects couple the stress-strain relation to the momentum-velocity relation. This coupling is absent in typical materials. While various theoretical models have predicted the existence of Willis coupling, experimental observation of Willis coupling has not, to our knowledge, been reported in the literature. We present here the first experimental evidence of Willis coupling in a one-dimensional unit cell consisting of a 0.12-mm-thick polyimide membrane under tension, a 5.9-mm-thick layer of air, and a perforated sheet of 0.45-mm-thick paper (perforation surface fraction 0.48, circular holes 3.1 mm in diameter). The properties of the unit cell were extracted using reflection and transmission data measured in a plane-wave impedance tube operating in the long-wavelength limit relative to the unit cell dimensions. The measured material properties, including the Willis coupling coefficient, are in good agreement with the properties predicted using a transmission line model. [Work supported by ONR.]

3:30

2pSA9. Experimental realization of acoustic double zero index metamaterials. Chengzhi Shi, Marc Dubois, Xuefeng Zhu, Yuan Wang, and Xiang Zhang (Dept. of Mech. Eng., Univ. of California, Berkeley, 3112 Etcheverry Hall, Berkeley, CA 94720, chengzhi.shi@berkeley.edu)

Acoustic double zero index metamaterial with simultaneous zero density and infinite bulk modulus induced by Dirac cone at the Brillouin zone center provide a practical solution for applications. The resulted finite impedance of this metamaterial can be designed to match with surrounding materials. However, such metamaterial consists of scatterers with lower sound speed than the matrix, which is fundamentally challenging for air acoustics because the sound speed in air is among the lowest in nature. Here, we experimentally realize double zero index metamaterial by periodically varying waveguide thickness, which modulates the speed sound of high order waveguide mode. With proper designed dimensions of the scatterers, a Dirac cone at 18.7 kHz formed by the degeneracy of one monopolar and two dipolar modes at the Brillouin zone center is obtain. The monopolar mode modulates the bulk modulus and the dipolar modes modulate the density, resulting in simultaneous zero density and infinite bulk modulus. In our experiment, a point source is mounted at the center of our metamaterial sample. The double zero index metamaterial collimates the wave emitted from the point source to a planar wavefront at 18.7 kHz. This point source in a pure waveguide emits cylindrical wave. The experimental result shows that the amplitude of the collimated plane wave is confined within 11 ± 1 degrees, closed to the theoretical limit 10.6 degrees.

3:45

2pSA10. 3D printed membrane-type acoustic metamaterials with structured masses. Alexandre Leblanc and Antoine Lavie (Université Artois, Faculté des Sci. Appliquées, Technoparc Futura, Béthune 62400, France, alexandre.leblanc@univ-artois.fr)

As society evolves, new technologies emerge. They should be considered to address persistent problems such as sound absorption at low frequencies. 3D solid prototyping printers are already used to obtain efficient sound diffusers, but remain marginal for creating acoustic absorbers. Recently, sub-wavelength absorbers have been proposed, particularly in the form of membrane-type acoustic metamaterials. These lasts usually consist on a decorated membrane resonator with tunable weight. In this work, membrane-type metamaterials are fabricated by fused deposition modeling, and both the membrane and the added masses are all made by the same flexible material. This allows to study other geometries, while structuring the added masses. Indeed, they can be split to increase peak efficiencies of this metamaterial to obtain an efficient band-stop filter. This application is illustrated with a base decorated membrane, using a divided central platelet and/or an additional split ring. Also, preliminary numerical simulations show that these structured membranes are likely to be more efficient than membranes with rigid inclusions when subjected to a diffuse sound field.

4:00

2pSA11. Elastic wave dispersion in pre-strained negative stiffness honeycombs. Benjamin M. Goldsberry and Michael R. Haberman (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bmgoldsberry@gmail.com)

A doubly periodic array of curved beams, known as a negative stiffness (NS) honeycomb, has recently been shown to exhibit tunable impact isolation by exploiting nonlinear mechanical behavior [Correa *et al.*, Rapid Prototyping J., **21** (2), 2015]. The primary benefit of NS honeycombs over standard honeycomb materials lies in their ability to return to their original shape after experiencing an impact. The recoverable nature of their response is a result of the curved beam structure which has a single stable configuration but experiences one or more buckling events when loaded. The complex nonlinear elastic response of NS honeycombs lead to large variations in mechanical stiffness, which makes these materials compelling candidates for tunable control of elastic waves. The present work investigates linear elastic wave propagation of a representative NS honeycomb lattice at varying levels of pre-strain. Eigenfrequency finite element analysis is performed on the unit cell subjected to finite uniaxial deformation. The longitudinal and transverse phase speeds are shown to be anisotropic and highly dependent on uniaxial pre-strain. This is especially true near points of instability, where one observes very different functional dependence of longitudinal and transverse wave motion on pre-strain.

4:15

2pSA12. Metamaterial ground cloak with anisotropic solid inclusions. Benjamin Beck, Amanda Hanford, and Peter Kerrian (Appl. Res. Lab., Penn State Univ., PO Box 30, MS 300B, State College, PA 16804-0030, benbeck@psu.edu)

There has been much research on the use of metamaterials within ground cloaks to hide objects on a rigid surface. Within the literature, ground cloaks made with metamaterials based on unit cells with solid inclusions have required the use of at least two distinct materials within the unit cell to achieve the necessary anisotropic material properties calculated from a transformation method. Here, we present a method to calculate the material properties of an anisotropic solid inclusion for use in a metamaterial ground cloak. The generated materials properties are then shown to be realized through an additively manufactured anisotropic solid inclusion. Finally, numerical results of cloaking performance of the metamaterial ground cloak are presented.

4:30

2pSA13. Effect of non-uniform mean flow to the scattering pattern of acoustic cloak. Wonju Jeon (Korea Adv. Inst. of Sci. and Technol. (KAIST), 291 Daehak-ro, Yuseong-gu, Daejeon 34141, South Korea, wonju.jeon@kaist.ac.kr)

During the last decade, most of acoustic cloak researches have been done within a theoretical framework in which the medium is at rest. However, such acoustic cloaks cannot preserve its unique properties or functions in the presence of flow. In this study, we propose a theoretical framework to consider the effect of non-uniform mean flow to acoustic cloak for the purpose of understanding the physics of acoustic cloak within flow and designing a new concept of metamaterial so called aeroacoustic cloak. We formulate a convective wave equation in which the differential operator contains the non-uniform velocity vector coupled with gradient operator and the equivalent source terms due to the mean flow are divided into two terms with their own physical meanings. The performance of existing convective cloak is investigated by using the present formulation in the presence of non-uniform flow. By comparing the existing and present formulations, the polarity and magnitude of equivalent source are compared: the present source shows hexapole pattern whereas the existing source shows quadrupole pattern. The non-uniformity effect was dominant in hexapole pattern. Finally, we found that although the compressibility effect is much smaller than non-uniformity effect in low Mach number range, it is not negligible in high Mach number flow, especially for small Helmholtz number. Understanding the nature of equivalent source is expected to provide a physical insight to design a new acoustic cloak within fluid convection.

Session 2pSC

Speech Communication: Articulatory and Acoustic Phonetics: Data and Methods (Poster Session)

Grant L. McGuire, Chair
Linguistics, UC Santa Cruz, 1156 High Street, Stevenson Academic Services, Santa Cruz, CA 95064

All posters will be on display from 1:00 p.m. to 5:30 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:15 p.m. and authors of even-numbered papers will be at their posters from 3:15 p.m. to 5:30 p.m.

Contributed Papers

2pSC1. Acoustic phonetic study of phonemes and tonemes of spoken Punjabi language. Shyam S. Agrawal, Shweta Bansal, Shambhu Saran, and Amritpal Singh (College of Eng., KIIT, Sohna Rd., Near Bhondsi, Gurgaon, Haryana 122102, India, dr.shyamsagrawal@gmail.com)

Punjabi language is one of the important languages among 22 official languages in India. It is spoken by about 105 million people in India. The present paper describes a study and results of detailed acoustic analysis of vowels, consonantal phonemes and tonemes as spoken by the speakers of Malwai dialect of Punjabi language. A database of 1500 words containing all the phonemes and tonemes, selected from a text corpus of 300,000 words were used for the study. These words were recorded and segmented by usingsignal processing tools to analyze the samples of speech. Fundamental frequency, first three formants, and bandwidths for nasal and non-nasal vowels were measured. For the study of consonants, duration of sub-phonemic segments such as occlusion, burst, and VOT have been computed and compared. Special features of tonemes have been studied and compared with non-tonal phonemic segments. Tones are fully phonemic and words with similar spellings are distinguished by varying tones: low, mid, and high and corresponding accent marks. It has been observed that intonation plays a significant role in the discrimination of phonemes and tonemes. These results can be used to create PLS and phonetic dictionary of Punjabi speech.

2pSC2. Intonation contours in Hawai'i Creole and Hawai'i English show influence from Hawaiian. Victoria B. Anderson and M. J. Kirtley (Dept. of Linguist, Univ. of Hawai'i, 569 Moore Hall, 1890 East-West Rd., Honolulu, HI 96822, vanderso@hawaii.edu)

Hawai'i Creole (ISO 639-3 hwc) and Hawai'i English (HE) are speech varieties spoken by local residents of the Hawaiian Islands. Three native speakers of HWC, seven native speakers of HE, and three native speakers of Hawaiian (ISO 639-3 haw) were recorded speaking in casual interviews, participating in map tasks, and saying phonetically controlled sentences. Analysis focused on utterances containing the high (or rise) plus steep fall contour, which is not found in Mainstream American English but is often used for continuations and polar questions in HWC, HE, and Hawaiian (Vanderslice & Pierson 1967, Anderson 2003, Murphy 2013). Results for all three speech varieties showed alignment of F0 minimums with lexically stressed syllables at the end of an intonation phrase, with F0 peaks occurring on immediately preceding syllables. All three speech varieties show a preponderance of this steep drop in continuations and polar questions, confirming the idea that this is a feature borrowed from Hawaiian (Bickerton and Wilson 1987, Anderson 2003.)

2pSC3. The effect of context on the articulation of harmonic vowels in Assamese: An ultrasound study. Diana Archangeli and Jonathan Yip (Linguist, Univ. of Arizona/Univ. of Hong Kong, Pok Fu Lam Rd., Hong Kong, Hong Kong, darchang@hku.hk)

Tongue root (TR) harmony in Assamese affects high back vowels and mid vowels (Mahanta 2007). The description (based on perception and limited acoustic measurements) is that Assamese shows the following five properties.—Constrast: TR contrasts in monosyllabic words. — [bel] "bell" vs. [bɛl] "stupid person"—Harmony: Vowels agree with a following vowel, e.g., triggered by suffixes. — [nomori] "not-die.non-finite" vs. [nɔmɔra] "not-die.habitual.non-finite"—CC-blocking: Harmony crosses one consonant only, no harmony in $\_\__{CCi}$ vs. harmony in $\_\__{Ci}$. — [bɔnti] "lamp" vs. [nomori] "not-die.non-finite"—N-blocking: There is no harmony in [….ɔNi…] (*[…oNi…], N = any nasal). — [k^{h}ɔmir] "leavening agent" vs. [nomori] "not-die.non-finite"—Idiosyncrasy: /a/, which otherwise does not harmonize, appears as [e]/[o] before an idiosyncratic set of suffixes. — [kɔpal] "destiny" vs. [kopolija] "destined." This ultrasound study explores Assamese vowel articulation to determine the nature of tongue position in different harmonic contexts, based on data collected in Assam from 10 subjects. Limitations of ultrasound technology prevented a clear image of the tongue root; indirect measures based on tongue dorsum height and backness are used instead. Results confirm perceptual/acoustic descriptions, showing a contrast in tongue position in all contexts: (i) Contrast, (ii) Harmony, (iii) CC-blocking, (iv) N-blocking, and (v) Idiosyncrasy.

2pSC4. An acoustic analysis of morphological glottalization in Hul'q'umi'num' Salish. Kevin M. Baetscher (Linguist, Univ. of Hawaii at Mānoa, 3239 Huelani Dr., Honolulu, HI 96822, kbaetsch@hawaii.edu)

Glottalization is lexically contrastive for both obstruents and resonants in Hul'q'umi'num' Salish. Glottalization of resonants is also a morphological process involved in the marking of progressives forms, among others. In these forms, all non-initial resonants (but not obstruents) become glottalized. This study investigates the question of whether the glottalization involved in the formation of progressives is a process that affects resonants locally, or whether glottalization as a feature is applied to the whole word. To this end, the glottal pulse periods of vowels in progressive forms have been measured and compared to modal schwas, to see whether glottalization in these words applies to vowels, as well. The results are negative and thus support the hypothesis that morphological glottalization (at least of progressive forms in Hul'q'umi'num') applies to resonants only, as opaque as this may appear from a phonological point of view.

2p TUE. PM

2pSC5. Vowel duration before ejective stops. Gašper Beguš (Dept. of Linguist., Harvard Univ., Boylston Hall, 3rd Fl., Cambridge, MA 02138, begus@fas.harvard.edu)

It has been well established that vowels are phonetically longer before voiced than voiceless stops, all else being equal. To my knowledge, however, no studies have measured vowel durations before *ejective* stops. This paper aims to fill this gap: I present preliminary results from a phonetic experiment with nine speakers of Georgian, establishing for the first time that vowels are (i) significantly longer before ejective stops than before voiceless aspirated stops and (ii) significantly shorter before ejective stops than before voiced stops. While the dependent relationship between vowel duration and obstruent voicing is well known, the cause of this relationship is poorly understood. The results presented in this paper shed new light on our understanding of differences in vowel durations and, in doing so, render several proposed explanations for these differences considerably less likely. In particular, my results show that (i) the voice feature itself likely does not affect vowel duration, (ii) closure duration of the following stop is not inversely correlated with vowel duration, and (iii) perceptual factors likely play no role in determining vowel length. My study shows that laryngeal features are the best predictors of vowel duration, suggesting an articulatory connection between these two factors.

2pSC6. Tenseness in Northern Yi: An ultrasonic and acoustic study. Shuwen Chen (The Dept. of Linguist and Modern Lang., The Chinese Univ. of Hong Kong, G19, Leung Kau Kui Bldg., Hong Kong, Hong Kong, chenshuwen@link.cuhk.edu.hk) and Ziwo Lama (College of Yi Studies, Southwest Univ. for Nationalities, Chengdu, Sichuan, China)

A contrast between tense and lax vowels is a main feature of many Tibeto-Burman (TB) languages spoken in Southwest China. This study investigated the articulation and acoustics of tense-lax contrast in two Northern Yi dialects: Suondi and Yinuo. Both dialects have five pairs of phonologically-defined tense and lax vowels. Previous studies have shown that tense and lax vowels in TB had distinct patterns in phonation types and f0, while the difference in other acoustic dimensions—vowel quality, duration, and intensity are rarely examined, and the difference in the lingual gestures is largely unknown. The current study examined the articulatory and acoustic characteristics of tense-lax contrast using ultrasound imaging and spectrogram analysis. Our preliminary results showed that articulatorily, tense vowels were produced with more retracted tongue root. Acoustically, tense vowels had higher F1 and lower F2, which was caused by the lowered and retracted tongue position. The intensity of tense vowels was lower than that of lax vowels for front vowels. No significant difference was found in duration. Based on the articulatory and acoustic data, as well as the vowel harmony phenomenon in Yi, we propose that the tense-lax distinction in Northern Yi can be better represented as [+RTR] phonologically.

2pSC7. F0 and plosive voicing in Afrikaans. Andries W. Coetzee, Patrice S. Beddor, Dominique Bouavichith, Justin T. Craft (Dept. of Linguist., Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109, coetzee@umich.edu), and Daan Wissing (North-West Univ., PotchefstRm., North-West, South Africa)

Afrikaans plosives are traditionally described as contrasting in voicing ([b d] vs. [p t]). Coetzee *et al.* (2014, *J. Acoust. Soc. Am.* **135**, 2421), however, showed that the voicing contrast is collapsing word-initially, with voiced plosives merging with voiceless plosives. They also found that voicing loss does not result in loss of lexical contrast, which is preserved on the following vowel (high f0 after historically voiceless and low f0 after historically voiced plosives). That study investigated word-initial plosives, leaving unanswered whether word-medial plosives are also devoicing. The current study addresses this question. Acoustic analysis of data collected from nine Afrikaans speakers replicated the results of Coetzee *et al.* for word-initial plosives. For word-medial plosives, a robust voicing contrast was found along with a post-plosive f0 difference. The f0 difference in medial position was comparable to that in initial position in both magnitude and duration, and extended throughout the vowel. While lexical contrasts are hence cued by only f0 word initially, they are differentiated by both plosive voicing and the f0 of the following vowel word medially. The relevance of these results for theories of sound change and theories of phonological contrast will be discussed. [Work supported by NSF.]

2pSC8. Contextual place neutralization of nasal codas in Taiwanese Mandarin: An Elecrtomagentic Articulography Study. Yuehchin Chang and Yi Yeh (Inst. of Lingsuit., National Tsing Hua Univ., 101, Sec. II, Kuang Fu Rd., Hsinchu, Taiwan 300, Taiwan, ycchang@mx.nthu.edu.tw)

It has long been noted that /in/ and /iŋ/ are neutralized to either [in] or [iŋ] in Taiwanese Mandarin. This study aims at investigating this phenomenon and more generally, coarticulatory patterns of VN rimes ([an] and [aŋ] here) from articulatory perspectives. Kinematic data from two speakers were collected using an NDI Wave. Regarding /in/ vs. /iŋ/, a principle components analysis (PCA) was conducted to see if a specific flesh point is more positively (or negatively) related to the distinction of the tongue contours. For the female speaker, Tongue Tip (TT)-height is most positively correlated, whereas Tongue Dorsum (TD)-height is most positively correlated for the male speaker. Regarding the nasal rimes with /a/, TD-height is most positively correlated for the male speaker, while TT-backness is most negatively correlated for the female speaker. The main findings are: (i) the contextual neutralization of place contrasts in nasal codas is not articulatorily confirmed, that is, the phenomenon in question is better characterized as incomplete neutralization, (ii) there is inter-speaker variation in the implementation of articulatory gestures and (iii) "anomalous" tongue shapes are found in the production of [iŋ], suggesting that the actual coarticulatory patterns within VN rimes are more complicated than previously thought. These obtained data will also be compared with the acoustic and airflow data reported in our previous studies.

2pSC9. The vowel system of Nasa Yuwe. Didier Demolin, Angélique Amelot, Lise Crevier-Buchman (LPP Paris 3, 19 rue des Bernardins, Paris 75005, France, ddemolin@univ-paris3.fr), Tulio Rojas, and Esteban Diaz (Universidade de Cauca, Popayan, Colombia)

Nasa Yuwe (sometimes referred as Paes) is a language spoken in Colombia. The vowel system has four vowels timbers [i, e, a, u] that can be oral or nasal (short and long); breathy, breathy nasal; glottal, glottal nasal. This makes a total of 32 contrastive vowels. This study examines the acoustic and articulatory characteristics of each type of vowel. Data come from synchronized acoustic, EGG and fiberscopic measurements. A special focus is given to the acoustic and articulatory features of breathy and glottal vowels. The breathy character appears in the last part of the vowel. This is realized by a slightly wider opening between the vocal folds or by an aperture between the arytenoids. The glottal character of the vowels also appears at the end. The data clearly show that glottal vowels are different from a laryngalized counterpart. Spectrographic and EGG observations show irregularities towards the end glottal vowels. These are followed by a closure or a stricture made either at the glottis or in the epilaryngeal tube. Fiberscopic and EGG data show that there is a glottal closure at the end of these vowels. This closure precedes a closure of the ventricular bands.

2pSC10. Method for recognizing multiple emotions from acoustic features using soft clustering. Joji Uemura, Kazuya Mera, Yoshiaki Kurosawa, and Toshiyuki Takezawa (Graduation School of Information Sci., Hiroshima City Univ., 3-4-1, Ozuka-higashi, Asa-minami-ku, Hiroshima 7313194, Japan, 4dai0715@gmail.com)

When a speaker expresses multiple emotions at the same time, emotion recognition methods using multi-class classification cannot recognize all the emotions because they always choose "an" emotion class. We thus propose a method to recognize multiple emotions expressed in parallel by using soft clustering (fuzzy c-means algorithm). Seven support vector machines for regression (SMOreg) calculate element values of the data. The SMOregs calculate probabilities of the following seven emotions: anger, anticipation, disgust, fear, joy, sadness, and surprise. First, 584 pieces of emotional voice data annotated with one of the seven emotions were used to calculate eight cluster centers on the basis of 384 acoustic features. Next, the characteristic of each cluster was defined on the basis of annotated tags of data belonging to the cluster. The characteristic was expressed in terms of valence (pleasure-displeasure) and arousal (high-low). To evaluate whether voice data that have two high membership grades have both emotion characteristics of the high-membership clusters, calculated membership grades were compared with the results of a questionnaire in which 10 subjects gave their impressions of the voices. We will show the experimental results and the comparison in the presentation.

2pSC11. A nasograph and fibrescopic analysis of nasal diphtongs in Brazilian Portuguese. Rita Demasi, Angélique Amelot, Lise Crevier-Buchman, and Didier Demolin (LPP Paris 3, 19 rue des Bernardins, Paris 75005, France, ddemolin@univ-paris3.fr)

This paper examines velum movements during the production of front and back nasal diphtongs [ẽĩ], [ãõ] in Brazilian Portuguese. Velum movements were measured with a female speaker from Sao Paulo using a nasograph and fibrescopic video-recordings synchronized with acoustic recordings. The nasal diphthongs were always situated at the end of a syllable in real words containing nasal diphthongs. Contrasting oral diphthongs were also recorded for comparison. Our results show that there is a nasal appendix the end of the nasal diphthongs. This is made by a contact between the tongue dorsum and the velum. These observation are based on nasograph and fibrescopic data. They confirm previous measurements made by EMA and aerodynamic measurements with the same subject. This appendix is sometimes perceived as a short velar nasal consonant. The duration of nasal appendix is slightly different when front and back nasal diphthongs are compared. Data also show that the lowering of the Velum occurs gradually for a nasal diphthong. This can start at any time during the first part of the diphthong.

2pSC12. For bilinguals, Enxet vowel spaces smaller than Spanish. John Elliott (Linguist, Univ. of Hawai'i at Mānoa, 1117 3rd Ave., Unit C, Honolulu, HI 96816, johnaell@hawaii.edu)

This study compares the vowel spaces of bilingual speakers of Spanish and Enxet Sur (ISO 639-3: enx), a moderately threatened language spoken by around 4000 speakers in the Paraguayan Chaco which purportedly lacks high vowels. The Enxet vowel system has only three phonemic vowel qualities /e,a,o/ and a length distinction. Using recorded natural speech data from two Enxet/Spanish bilinguals, formant values for vowel tokens are used to establish phonetic targets and acoustic vowel clouds for each of the phonemic vowels in both languages. A comparison of these vowel spaces suggests that an individual bilingual speaker uses a smaller vowel space when speaking Enxet (less vowel height, higher minimum F1 frequencies) than when speaking Spanish. Languages like Enxet lacking in true high vowels present problems for the theory of adaptive dispersion in that they do not "take advantage" of maximal distinctiveness in the acoustic and articulatory quality of their vowels. This is the first known phonetic investigation of an Enlhet-Enenlhet language.

2pSC13. Velar flutter does not Occur on Persian radico-uvular /G/ sound: Case reports. Marziye Eshghi (Speech, Lang. and Hearing Sci., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Craniofacial Ctr., Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu) and David J. Zajac (Dental Ecology, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Nasal turbulence with velar flutter is an obligatory symptom that occurs during production of high-pressure sounds due to a partially closed velopharyngeal (VP) port. A small VP gap creates air-pressure high enough to set the marginal edges of the velum into vibration. The flutter component is characterized spectrally by low frequencies (Zajac & Preisser, 2015) and vibratory activity through auto-correlation functions (Eshghi *et al*., 2015). In this study, two speech-language pathologists independently listened to the audio recordings of stops produced by two Persian-speaking children (1 male, 8 yrs and 1 female, 9 yrs) with repaired cleft palate. Speech samples consisted of Persian plosives including /p, b, t, d, k, g, G/. The two listeners achieved 100% agreement that velar flutter occurred on all stops except /G/, a voiced radico-uvular sound. Spectral evaluation indicated characteristics of flutter were present during all stops except for /G/. It is hypothesized that tissue vibration was inhibited during the production of /G/ due to the the tongue making contact with the posterior part of the soft palate and/or uvula. The findings provide cross-linguistic evidence in support of tissue vibration (velar flutter) rather than displacement of mucous (bubbling) as the primary perceptual component of nasal turbulence.

2pSC14. Voiceless vowels in Miyako. Catherine Ford, Benjamin V. Tucker, and Tsuyoshi Ono (Univ. of AB, Unit 204 7111 80 Ave. NW, Edmonton, AB T6B0C7, Canada, cford1@ualberta.ca)

The present study investigates the nature of vowel devoicing in Miyako, a Ryukyuan language spoken on remote Japanese islands near Taiwan. High vowels in Japanese are commonly described as being devoiced after voiceless consonants in two environments: 1) before a voiceless consonant, 2) at the end of a word (Shibatani 1990). Being a Japonic language, Miyako is also described as having a similar phenomenon but devoices all vowels in its phonemic inventory /i, ɨ, ɑ, u/ (Hayashi 2013). Data from several speakers suggests that there may be cases of vowel deletion as well. Current data suggest vowel deletion between two voiceless consonants, creating the potential for consonant clusters in Miyako—something that is said not to occur in Japanese. The acoustic nature of voiceless vowels in Miyako is investigated using both elicited data and discourse data. Data were collected from the Ikema Island community, limiting data to the Ikema dialect. Data were analyzed in PRAAT to determine whether potential voiceless vowels are indeed devoiced or deleted. The present investigation contributes to our overall understanding of the nature of vowel devoicing/deletion in a highly under-documented language Miyako and Ryukyuan languages in general.

2pSC15. A preliminary ultrasound examination of Upper Sorbian rhotics. Phil Howson (The Univ. of Toronto, Sidney Smith Hall, 4th Fl. 100 St. George St., Toronto, ON M5S 3G3, Canada, phil.howson@mail.utoronto.ca)

Upper Sorbian has been influenced tremendously by German, causing it to adopting the uvular rhotic. The current study examines the uvular rhotic and its palatalized counterpart in three vocalic environments (word initial, intervocalic, and word final) with the vowels /a, e, o, i/ using ultrasound. The preliminary results indicate a striking difference between the two phonemes. The plain uvular rhotic was characterized by a retracted and raised tongue dorsum, accompanied by a slightly retroflexed tongue tip. The retroflexed tongue tip is surprising for a uvular rhotic. It could be related to requirements for rhotic production or a remnant of the alveolar rhotic Upper Sorbian had historically. The environments with /i/ were especially striking: the uvular rhotic was palatalized intervocalically; the tongue body was raised near the palatal region, while the dorsum retracted strongly. In other word positions, the rhotic caused a lowering and retraction of /i/. The palatalized rhotic had a raised tongue body, but the tongue dorsum was as retracted as the plain rhotic and less susceptible to environmental influence. The tongue tip was laminal in articulation, rather than retroflexed. The more consistently retracted tongue dorsum may be related to a need to maintain rhotic qualities. [Work supported by SSHRC.]

2pSC16. Temporal organization of onglides in standard Chinese and Taiwanese Mandarin: A cross-dialectal study. Feng-fan Hsieh, Guansheng Li, and Yueh-chin Chang (Inst. of Linguist., National Tsing Hua Univ., 101 Kuang-fu Rd., Sec. 2, Hsinchu 30013, Taiwan, ffhsieh@mx.nthu.edu.tw)

This study examines the temporal organization of prenuclear glide (onglide) gestures within a syllable. Kinematic data were collected from four Standard Chinese (SC) and four Taiwanese Mandarin (TM) speakers using an NDI Wave. Mean relative timing of C/G gestures to a (heterosyllabic) consonantal anchor /p/ were calculated using the lag between achievement of constriction of the C/G and /p/. By comparing a given subject's mean lags for Cs/Gs with CGs, we calculated the subject's leftward (C vs. CG) and rightward (G vs. CG) shifts associated with /pj/ and /kw/. Results show that in SC, onglides /j/ and /w/ don't pattern alike: for /kw/, all SC speakers show both leftward and rightward shifts (i.e. the C-center effect), except for subject S3, who fails to show a leftward shift (due to a different regional accent). For /pj/, only rightward shifts are found for all SC speakers. In contrast, regarding /kw/, only leftward shifts are found for all TM speakers, except for subject S8 (due to a different accent), while for /pj/, all TM speakers show rightward shifts, again, except for subject S8. Our findings thus suggest a cross-dialectal difference in /kw/ and confirm the possibility of the rightward shift in gestural coordination. Implications for the phonological status of onglides in Mandarin Chinese will also be discussed.

2pSC17. Voicing contrast in Punjabi Singletons and Geminates. Qandeel Hussain (Linguist, Macquarie Univ., Australia, 65 Gwen Meredith Loop, Franklin, ACT 2913, Australia, qandeel.hussain@students.mq.edu.au)

Cross-linguistic studies have shown that voiceless geminates are more frequent than voiced geminates (Blevins, 2004). From an articulatory perspective, maintaining voicing and long closure duration in voiced geminates is challenging (Ohala, 1983). Punjabi, an Indo-Aryan language, has been reported to contrast voiceless/voiced singletons and geminates. However, it is not known if voiced geminates in Punjabi are completely voiced and how their durational properties differ from voiceless geminates. The aim of the present study was therefore to investigate the acoustic characteristics of Punjabi voiceless/voiced singleton and geminate stops. Five native speakers of Punjabi participated in the experiment (24-26 years, M= 24.6). The stimuli consisted of nonce words with word-medial (C_2) voiceless/voiced singleton and geminate stops ($C_1V_1C_2V_2$). The C_2 duration was measured in PRAAT, by using the broadband spectrograms and visual inspection of the waveforms (Ridouane, 2007). The durational analysis of C_2 indicated that unlike voiceless singletons/geminates, voiced singletons/geminates showed complete voicing during the closure phase. These findings differ from Tokyo Japanese where voiced geminates are partially devoiced (Kawahara, 2016). However, voiceless singletons/geminates are slightly longer in duration compared to voiced singletons/geminates. The current study raises questions about the cross-linguistic differences in voiceless/voiced singletons and geminates.

2pSC18. Duration of Japanese singleton and geminate stops with devoiced vowel in various speaking rates. Shigeaki Amano (Faculty of Human Informatics, Aichi Shukutoku Univ., 2-9 Katahira, Nagakute, Aichi 480-1197, Japan, psy@asu.aasa.ac.jp), Kimiko Yamakawa (Culture and Lang., Shokei Univ., Kikuchi-gun, Kumamoto, Japan), and Mariko Kondo (School of Int. Liberal Studies, Waseda Univ., Shinjuku-ku, Tokyo, Japan)

Previous studies have shown that a devoiced vowel makes burst duration of a singleton stop consonant longer than a voiced vowel does. However, the effects of devoiced vowels on stop consonants including a geminate have not been fully investigated. To clarify the effects, durations of a burst and a closure in Japanese stops pronounced by 19 native Japanese speakers were analyzed by stop type (singleton or geminate) and speaking rate (fast, normal, or slow). Analysis revealed that at all speaking rates a devoiced vowel makes the burst duration of singleton and geminate stops longer than a voiced vowel does. Analysis also revealed that at normal and fast speaking rates a devoiced vowel has no effect on closure duration in singleton or geminate stops. However, at a slow speaking rate, a devoiced vowel makes the closure duration of a singleton stop longer. These results indicate that a devoiced vowel consistently lengthens the burst duration of singleton and geminate stops but that its effects on closure duration depend on both stop type and speaking rate. [This study was supported by a special-research grant of Aichi Shukutoku University in 2015-2016, and by JSPS KAKENHI Grants Number JP 25284080 and 26370464.]

2pSC19. Oral/nasal airflow during Japanese stop consonants. Masako Fujimoto (Adv. Res. Ctr. for Human Sci., Waseda Univ., 2-579-15, Mikajima, Tokorozawa, Saitama 359-1192, Japan, mfuji@viola.ocn.ne.jp) and Reiko Kataoka (Adv. Res. Ctr. for Human Sci., Waseda Univ., San José, CA)

Voiced obstruents have inherent susceptibility for devoicing due to the Aerodynamic Voicing Constraints (AVC), and the susceptibility is higher for geminate obstruents than singletons. As a way to investigate how Japanese speakers realize the contrast between the [+/-voice] contrast in obstruents, we examined oral and nasal airflow patterns during intervocalic voiced and voiceless stops, in singletons and geminates. The results showed asymmetry between single and geminate stops in realization of the stop voicing contrast. Airflow pattern clearly differentiates voiced vs. voiceless contrast in singletons, but the airflow patterns are similar in geminates. Acoustic signals also shows the same asymmetry between the singletons and geminates. The observed convergence—clear voicing contrast in singletons vs. the lack of the contrast in geminates both in air flow and acoustic signals indicate neutralization of the voiced geminates into voiceless ones. Our results support the idea of phonetic bases in phonological patterning of voicing neutralization in Japanese geminate stops.

2pSC20. Pitch accent and vocal effort in Japanese. Ryota Iwai (Dept. of Korean Studies, Graduate School of Humanities and Sociology, Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo 113-0033, Japan, 3181957501@snu.ac.kr) and Ho-Young Lee (Seoul National Univ., Seoul, South Korea)

It is implicitly assumed that each mora is pronounced with similar vocal effort in Japanese. Our intuition, however, is that the first or second mora is pronounced with the greatest vocal effort. This paper aims to verify this observation by analyzing vocal effort parameters CPP and H1-A3. We prepared two sets of real words and two sets of nonce words, each set consisting of 3 two mora words, 4 three mora words and 5 four mora words, varying the position and existence of pitch accent. Twelve subjects aged 21-39 participated in the recording. They read randomized test words 6 times. The result of the H1-A3 measurement shows that the first mora is significantly pronounced with the greatest vocal effort irrespective of the place and existence of pitch accent. The results of the CPP measurement show that the initial accented mora is pronounced with the greatest vocal effort, and that the second mora is pronounced with greater vocal effort than the first one if the first mora is not accented. The non-initial moras of three or four mora words are pronounced with similar vocal effort if the third or fourth mora is accented or a test word is unaccented.

2pSC21. Acoustic characteristics of liquid geminates in Japanese. Maho Morimoto (Linguist, Univ. of California, Santa Cruz, 1156 High St., Santa Cruz, CA 95064, mamorimo@ucsc.edu)

Japanese contrasts consonant length. Much research has been done on the acoustic characteristics of obstruent geminates in Japanese, identifying the constriction duration as the primary acoustic correlate. The ratios of the duration of the consonants and the surrounding vowels have also been investigated, and it was pointed out that the ratios differ from some other languages with distinctive consonant length, such as Italian. The present study examines the acoustics of liquid geminates in Japanese. Phonological phenomena in Japanese such as verb inflection demonstrate that the language dislikes liquid geminates. However, while not as common as obstruent or nasal geminates, liquid geminates are not impossible; they are present in certain loanwords and emphatic expressions. I report on the acoustic correlates of liquid geminates focusing on consonant sonority and consonant-vowel ratio, in three environments controlled for vowel quality and pitch-accent position: namely, loanwords (mostly from Italian and Arabic), emphatic forms of adjectives (e.g., karai "spicy" > karrai) and emphatic forms of onomatopoeic expressions (e.g., garagara "expresses emptiness" > garragara), pronounced by three native speakers of Tokyo dialect of Japanese in laboratory settings.

2pSC22. Prosodic analysis of storytelling speech in Japanese fairy tale. Takashi Saito (Appl. Comput. Sci., Shonan Inst. of Technol., 1-1-25, Tsujido Nishikaigan, Fujisawa, Kanagawa 251-8511, Japan, saito@sc.shonan-it.ac.jp)

This paper presents a prosodic analysis of storytelling speech in Japanese fairy tale. Recent advances in TTS technologies bring us high quality synthetic speech especially on acoustic aspects. On prosodic aspects, however, there is still room for improvement in the expressiveness since most systems use one-sentence speech synthesis scheme. For instance, storytelling applications expect speech synthesis to be capable of having a control mechanism beyond one sentence. In this paper, a prosodic database is built for real storytelling speech of professional narrators on Japanese fairy tale aiming at investigating the actual storytelling strategies and ultimately reflecting them on the expressiveness of speech synthesis. After conducting a baseline speech segmentation for phones, words, and accentual phrases in a semi-automatic way, a multi-layered prosodic tagging is manually performed to extract information on various changes of "story states" relevant to impersonation, emotional involvement, and scene flow control. Storytelling speech materials of six professional narrators for Japanese fairy tale are prepared to analyze their storytelling strategies. In particular, the dynamics of pitch contours is investigated in terms of expressiveness control beyond sentence considering narrator dependency as well. Based on the findings obtained, pitch control schemes are discussed for storytelling speech synthesis.

2pSC23. Acoustics of phonation types and tones in Santiago Laxopa Zapotec. Jeffrey M. Adler and Maho Morimoto (Linguist, Univ. of California, Santa Cruz, 1156 High St., Santa Cruz, CA 95064, mamorimo@ucsc.edu)

Zapotec languages are known for their complex tonal and phonation systems. Fieldwork in Santiago Laxopa Zapotec (SLZ), an undocumented language spoken in Oaxaca, Mexico, suggests the implementation of six tones (high, mid, low, high-falling, mid-falling and rising) and four phonation types (modal, creaky, checked and breathy). We report on the reliability of acoustic measures such as (H1-H2) and (H1-A1), which have previously been reported to characterize phonation types. Spectral measurements including H1, H2 (first and second harmonics), A1 (amplitude of F1) and A2 (amplitude of F2) are taken from vowels in near minimal pairs controlled for tones and quality. We also report on the relative implementation of tone and phonation. Silverman (1997) argues that languages with contrastive tone and phonation optimize the realization of laryngeal events such that non-modal vowels have a modal portion where the tonal event is implemented. We discuss if Silverman's predictions are borne out in SLZ.

2pSC24. Formant trajectories in the realization of 3 Malayalam rhotics. Alexei Kochetov and Phil Howson (Linguist, Univ. of Toronto, 100 St. George St., Sidney Smith 4076, Toronto, ON M5S 3G3, Canada, al.kochetov@utoronto.ca)

Malayalam has a typologically rare set of three rhotic consonants—an alveolar trill, an alveolar tap, and a retroflex/postalveolar approximant. The first one is described as velarized, while the other two are somewhat palatalized. Previous acoustic work by Punnoose *et al.* [Phonetica **70**, 274-297 (2013)] has established that the contrast can be differentiated by formant values around the constriction, as well as at the midpoint of adjacent vowels. Less clear, however, is the temporal extent of these formant differences and their overall dynamics. In this study, we analyze three rhotics produced by 10 male Malayalam speakers. Measurements of F1-F4 were taken at 10 points throughout the VCV interval and subjected to a series of Smoothing Spline ANOVAs. The results revealed differences in F2 (tap, approximant > trill) and F3 (tap, trill > approximant) among the rhotics extended through most of the preceding and following vowel duration, while differences in F1 (trill > tap, approximant) were less extensive. While similar to the tap in its F2 trajectory, the approximant showed a more abrupt fall during the following vowel, seemingly indicative of a rapid tongue posture change at the release. [Work supported by SSHRC.]

2pSC25. Acoustics of epenthetic vowels in Brazilian Portuguese. Maria Kouneli (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, mrkouneli@gmail.com)

In this production study involving a task of real words and a task of nonce words, epenthesis in obstruent-obstruent clusters in Brazilian Portuguese was investigated and the acoustic properties of epenthetic and lexical [i] were compared. A number of linguistic factors were found to be significant predictors of the epenthesis rate: voicing of the consonants of the cluster, manner of articulation of the second consonant, place of articulation of both consonants, and position of the cluster with respect to the stressed syllable. Moreover, there was a higher epenthesis rate for nonce words than for real words. As for the quality of the epenthetic vowel, it was found to have the same F1 as the lexical [i] in the language, but it was significantly shorter in duration, and it had a significantly lower F2 in the nonce words, but not in the real words. Finally, we found a relatively high rate of deletion of lexical [i], conditioned by the same factors that conditioned epenthesis rate, but in the opposite direction. It is concluded that epenthesis is a phonological phenomenon in Brazilian Portuguese, but that there is also acoustic reduction in the language that affects both lexical and epenthetic vowels.

2pSC26. Articulatory correlates of /w/ in Korean. Soohyun Kwon (Univ. of Pennsylvania, 3810 Walnut St., Linguist Lab, Philadelphia, PA 19104, suhyunkwon511@gmail.com)

The articulatory properties of /w/ have received relatively little attention compared to those of vowels and consonants. Although lip protrusion as well as dorsum raising and backing have generally been considered the major articulatory correlates of /w/, there is cross-linguistic evidence that the primary articulation of labiovelars, including /w/, is labial in some languages, while in other languages it is velar ([1]). Partly due to little articulatory data, there is little consensus on what articulatory features are involved in the production of /w/ in each language. This study used ultrasound to image tongue shapes for /w/ produced by three native Korean speakers. Also, a video camera was used to capture the movement of lips. The video data of movements of both tongue and lips were subjected to Optical Flow Analysis, a technique used to measure the magnitude of movement (MM) of objects in a video. The results show that there is prominent lip protrusion gestures for /w/ in Korean while there is little movement in tongue dorsum (the average MM of $lip = 1.78$; $tongue = 0.48$; $t(86) = 4.02$, $p < 0.001$). This finding indicates that the glide /w/ in Korean involves labial movements only, contrary to the previous description of Korean glides [2], [3].

2pSC27. Durational and spectral differences in Thai diphthongs and final glides. Phongphat Man-khongdi, Chutamanee Onsuwan (Dept. of English and Linguist, Faculty of Liberal Arts, Thammasat Univ., 2 Phrachan Rd, Phra Boromaharatchawang, Phra Nakhorn, Bangkok 10200, Thailand, phtmankhongdi@gmail.com), and Charturong Tantibundhit (Dept. of Elec. and Comput. Eng., Faculty of Eng., Thammasat Univ., Pathumthani, Thailand)

Acoustic analysis was conducted to compare Thai monophthongs /i, ii, ɯ, ɯɯ, u, uu, a, aa/, diphthongs /ia, ɯa, ua/, and vowel-to-glides (vowel + /j/ or /w/) in terms of duration, formant frequency, and spectral rate of change (TL_{roc}). Preliminary results from multiple repetitions of 30 target monosyllabic words (from 3 males) show that durational values of long monophthongs are approximately two times as long as the short counterparts. The diphthong onsets (/i/, /ɯ/, /u/) and offsets (/a/) as part of diphthongs appear to be more centralized than when they occur in monophthongs. Syllables with final glides appear to be longer than those with other final consonants, reducing length differences between syllables with short and long vowels. Interestingly, not only that Thai diphthongs and vowel-to-glides differ in their articulatory trajectories, but they appear to differ in terms of duration and TL_{roc} values. Average duration of diphthongs is shorter than that of vowel-to-glides. TL_{roc} of diphthongs is on average higher than that of long-vowel-to-glides, but lower than short-vowel-to-glides. Taken together, our preliminary results suggest and agree with the notion that transition of diphthongs is usually slower and more gradual than that of vowel-to-glides.

2pSC28. Tongue body shape during the production of Irish palatalization and velarization. Grant L. McGuire, Jaye Padgett (Linguist, UC Santa Cruz, 1156 High St., Stevenson Academic Services, Santa Cruz, CA 95064, gmcguir1@ucsc.edu), Ryan Bennett (Linguist, Yale, New Haven, CT), Máire Ní Chiosáin (Univ. College, Dublin, Ireland), and Jennifer Bellik (Linguist, UC Santa Cruz, Santa Cruz, CA)

Irish is an endangered Celtic (Goidelic) language that has a rare phonemic contrast between palatalized and velarized consonants for all places and manners of articulation. Using ultrasound tongue body imaging of 15 native speakers we provide comparative data on the phonetic realization of the contrast, specifically focusing on how place, manner, and vowel context each affect tongue body position during production for each of the three major dialects. Using principal components analysis, we find evidence for the role of tongue root advancement in the contrast independent of the (less surprising) roles of tongue body frontness and raising. This data have consequences for our understanding of the relationship between phonetic and phonological categories as well as the role of perceptual saliency in shaping inventories.

2pSC29. Ultrasound evidence for place of articulation and assimilation behaviors of the Japanese moraic nasal /N/. Ai Mizoguchi, Kevin Roon (The Graduate Ctr., City Univ. of New York, 365 Fifth Ave., Rm. 7304, New York, NY 10016, amizoguchi@gc.cuny.edu), and D. H. Whalen (Haskins Labs., New Haven, CT)

The syllable-final Japanese moraic nasal /N/ is commonly transcribed as velar or uvular or even placeless, but very little articulatory has been reported. This study investigated the tongue shape and position for /N/ in various phonological environments using ultrasound. /N/ assimilates to the place of following segments, so a variety of environments was also examined to assess whether the assimilation occurs categorically or gradually. Tens repetitions of 7 target words with a moraic nasal (/aNCa/, /aNa/, /aNaN/, and /ɯN/) and 6 control words without a moraic nasal (/aCa/, /aa/) were spoken by 4 native speakers of Japanese. Although there seems to be an oral target for moraic nasal, the place was different for each of our four speakers. The assimilation also varied among speakers, but a gesture for the moraic nasal remained in at least one phonological environment for all the speakers. Assimilation of Japanese moraic nasal to following segments is not always categorical and a gesture for the target of moraic nasal, even though varying among individuals, remains depending on the phonological environments. The ambiguities in transcribing /N/ seem to be reflecting the state of the language accurately: Even with four speakers, four patterns emerged. [Work supported by NIH DC-002717.]

2pSC30. Gradient phonemic contrast in Nanjing Mandarin. Keith Johnson (Linguist, UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, keithjohnson@berkeley.edu) and Yidan SONG (Nanjing Normal Univ., Nanjing, Jiangsu, China)

The variety of Mandarin Chinese spoken in Nanjing does not have a contrast between [n] and [l]. The standard variety of Mandarin which is now taught in schools in Nanjing does have the contrast. Prior research has shown that listeners' subjective impression of sound similarity is modulated by linguistic experience. When sounds are contrastive in a language they sound more different than when they are not contrastive. We conducted an experiment to test the subjective perceptual consequences of lack of [n]/[l] contrast in Nanjing, and of learning the standard variety of Mandarin in school. The stimuli were seven instances each of [tana] [tala] and [tara]. The listeners were 16 younger (< 45 yrs) and 16 older speakers of Nanjing Mandarin, and 35 speakers of American English. The listeners' task was to rate on a 5 point scale their subjective impression of the difference between the words (1 = similar, 5 = different). For American English listeners [l]/[r] pairs and [n]/[r] pairs were rated as different (3.8) while [n]/[l] pairs were heard as a little more similar than this (3.3). For older Nanjing Mandarin speakers [l]/[r] and [n]/[r] pairs were very different (4.8) and [n]/[l] pairs were very similar (1.5). Younger Nanjing Mandarin speakers also rated [l]/[r] and [n]/[r] pairs as very different (4.6), while the [n]/[l] pairs were not as similar (2.25) (standard errors < 0.1). Exposure to a non-native phonological contrast seems to have perceptually enhanced it gradiently.

2pSC31. A classification method for crowded situation using environmental sounds based on Gaussian mixture model-universal background model. Tomoyasu Tanaka, Sunao Hara, and Masanobu Abe (Graduate School of Natural Sci. and Technol., Okayama Univ., 1-1-1 Tsushimanaka Kita-ku, Okayama-shi, Okayama-ken, Okayama, Okayama 700-8530, Japan, pu4o7k17@s.okayama-u.ac.jp)

This paper presents a method to classify a situation of being crowded with people using environmental sounds that are collected by smartphones. A final goal of the research is to estimate "crowd-density" using only environmental sounds collect by smartphones. Advantages of the approach are (1) acoustic singles can be collected and processed at low cost, (2) because many people carry smartphones, crowd-density can be obtained not only from many places, but also at any time. As the first step, in this paper, we tried to classify "a situation of being crowded with people." We collected environmental sounds using smartphones both in residential area and downtown area. The total duration of the collected data is 77,900 seconds. The sound of "crowded with people" is defined as buzz-buzz where more than one person talked at the same time. Two kinds of classifiers were trained on the basis of the GMM-UBM (Gaussian Mixture Model and Universal Background Model) method. The one was trained with acoustic features that are generally used in speech recognition, and the other was trained with additional parameter of sound power. Experiment results showed that the parameter of sound power improves the F-measure to 0.60 from 0.58.

2pSC32. The phonological preparation unit in the production of English words by Korean learners. Sujin Oh and Jeong-Im Han (English, Konkuk Univ., 120 Neungdong-ro, Gwangjin-gu, Seoul 05029, South Korea, cutmirul@konkuk.ac.kr)

This study investigated how speakers of two languages differing in phonological preparation unit in planning spoken words encode a functional unit of their second language (L2). It was shown that the first phoneme of a word is a functional unit in English, whereas Korean has a strong syllable but not onset phoneme effects. Two groups of native Korean speakers depending on their length of residence in English-speaking countries were asked to name pictures in English, and the names shared the same onsets, rhymes, or had nothing systematically in common. The results showed that 1) no onset effect was shown regardless of length of residence, suggesting that L2 learners' phonological preparation unit is influenced by their native language functional unit; and 2) there was a rhyme interference effect regardless of length of residence, probably due to lexical competition. However, upon close inspection of the results of individual speakers, L2 learners are able to attend to the target language unit if they are extensively exposed to L2 phonology.

2pSC33. Temporal and spectral characteristics of the intervocalic consonant clusters in Korean. Hansang Park (English Education, Hongik Univ., 94 Wausan-ro, Mapo-gu, Seoul 04066, South Korea, phans@hongik.ac.kr)

Park(2008) showed that there is no significant difference in specific segment duration and their temporal structure between $\mathbf{VCC_{[Asp]}V}$ and $\mathbf{VC_{[Asp]}V}$, and between $\mathbf{VCC_{[For]}V}$, $\mathbf{VCC_{[Len]}V}$, and $\mathbf{VC_{[For]}V}$, where V stands for vowel, C for consonant, [Asp] for aspirated, [For] for fortis, and [Len] for lenis. He also showed that there is no significant difference in consonant duration between a singleton and a geminate when the following C is [For] or [Asp], but a significant difference between them when the following C is [Len]. On the other hand, Park(2005) showed that there is a significant difference in phonation type index k (PTI k) across consonant types, where PTI k presents a single and simplified measure of the spectral tilt, which is free from the effects of fundamental frequency and vowel quality. This study, by applying Park(2008)'s and Park(2005)'s methods, investigates temporal and spectral characteristics of the consonant clusters of the other manners and places of articulation. The results showed that temporal characteristics present a similar pattern to those of Park(2008), and that spectral characteristics are associated with the phonation identity of the final consonant of the intervocalic consonant clusters.

2pSC34. Word boundaries attenuate the effects of emphasis in Lebanese Arabic. Andrew N. Pick (Linguist, Univ. of Hawaii Manoa, 1617 Kapiolani Blvd. Apt. 407, Honolulu, HI 96814, pick@hawaii.edu)

Arabic has a series of pharyngealized consonants, called "emphatics," which are phonologically distinct from their plain counterparts. The strongest acoustic correlate of emphasis is a lowering of the second formant on vowels adjacent to the emphatic consonant. Arabic varieties differ in how far emphasis spreads, and previous work suggests that varieties may differ in whether emphasis can cross word boundaries. This study examines the effect of Lebanese Arabic emphatics on preceding vowels in two environments: within the word and within the phrase. Five native Lebanese speakers were recorded reading a list of phrases containing a target vowel preceding either an emphatic or plain consonant, with or without an intervening word boundary. In both environments, vowels preceding emphatic consonants have a lower F2 than vowels preceding plain consonants. Additionally, F2 is lowered more on vowels that are within the same word as the conditioning emphatic. Overall, these results indicate that emphasis in Lebanese Arabic affects preceding vowels within the domain of the word as well as the domain of the phrase, but this effect is stronger within the word.

2pSC35. An ultrasound study of the nasal substitution in Javanese. Lang Qin (Dept. of Linguist, The Univ. of Hong Kong, General Office 9.30, Run Run Shaw Tower, The Centennial Campus, Hong Kong 999077, Hong Kong, qinlang@hku.hk)

Nasal substitution in Austronesian languages refers to replacement of voiceless obstruents by homorganic nasals under certain morphological conditions. In Javanese, for example, the active form of verb [tuku] "buy" is [nuku] and that of [tʃakar] "scratch" is [ɲakar]. However, exception does exist; namely, the sibilant [s] is paired with a palatal nasal [ɲ], e.g., the active form of [sawaŋ] "see" is [ɲawaŋ]. This ultrasound study aims to verify: i) whether the perception of two distinct places of articulation ([s]/[ɲ]) is matched in the articulation, suggesting an abstract relation between the two sounds, ii) or whether the homorganic pattern seen elsewhere in this morphological paradigm is matched in the articulation, suggesting a concrete relation between the two sounds. SS ANOVA results based on data of 8 Javanese speakers reveal that tongue position of /s/ bears more resemblance to /t/ and its nasal counterpart /n/, whereas the the tongue position of the nasal counterpart of /s/ is more posterior and closer to that of /tʃ/ and its nasal counterpart /ɲ/, which supports the abstract relation between [s] and [ɲ].

2pSC36. The Pohnpeian stop contrast between laminal alveolars and apical dentals involves differences in VOT and F2 locus equation intercepts. Bradley Rentz and Victoria Anderson (Linguist, Univ. of Hawaiʻi at Mānoa, 1890 East-West Rd., 569 Moore, Honolulu, HI 96822, rentzb@hawaii.edu)

Pohnpeian (ISO639-3 pon) is a phonetically understudied Oceanic language spoken by about 34,000 people in the Federated States of Micronesia, and 12,000 in the United States. Pohnpeian has a typologically uncommon laminal alveolar stop /t̻/ that contrasts with an apical dental stop /t̺/. Three male and two female native speakers were recorded in Honolulu, saying phonetically controlled wordlists in sentence frames. VOTs, F2s at vowel onset and steady state, and burst spectral moments (center of gravity and skewness) were measured in PRAAT and analyzed in R. Hierarchical linear modeling showed that /t̻/ has a significantly longer VOT than /t̺/. Linear regressions showed similar F2 locus equation slopes, but a higher intercept for /t̻/. Neither COG nor skewness exhibited significant differences. The locus equation intercept results, while similar to those of Sussman *et al.* 1993 for Urdu, have a different cause. Urdu /ɖ/'s higher intercept is due to higher F2 onsets than that of /d̺/, Pohnpeian's /t̻/ and /t̺/ have similar F2 onsets; /t̻/'s higher intercept is due to vowel allophony: a lower F2 steady state for /t̻/. The burst spectral results remain puzzling in light of source-filter theory, which predicts a higher COG for /t̺/ than /t̻/.

2pSC37. Relative fundamental frequency of voiced and voiceless stops and fricatives for different phonation types in normal French speakers. Camille F. Robieux and Christine Meunier (Aix Marseille Univ, CNRS, LPL, Aix-en-Provence, France, 5 Ave. Pasteur, Aix-en-Provence 13100, France, camille.robieux@lpl-aix.fr)

The relative fundamental frequency (RFF) was recently used to assess vocal effort in normal and dysphonic subjects since it reflects the vocal fold tension at the offset and onset of vowels with, in particular, lower values at the edges of voiced consonants compared to their voiceless counterparts. In this study, we explored the RFF variations according to the voicing status and the articulation manner of consonants, in several types of speech production known to involve different levels of vocal effort. Twelve native French speakers (6 females) with normal voice were asked to produce 24 trains of eight syllables in seven vocal conditions: spontaneous, low, high, soft, loud, breathy, and pressed voice. We considered the unaspirated voiceless and voiced consonants /p, b, f, v/ with the middle open vowel /a/. As expected, the RFF was lower for voiced than for voiceless consonants. In addition, it was lower for stops than for fricatives. The RFF was also lower for the high, loud, and pressed conditions while it was higher for the low, soft, and breathy conditions in comparison to the values for the spontaneous voice. The RFF measurement seems to be efficient to discern subtle levels of vocal effort in normal subjects.

2pSC38. Preliminary study on physiological contrasts between long and short vowels in Japanese. Ayako Shirose (Tokyo Gakugei Univ., 4-1-1 Nukuikita-machi, Koganei, Tokyo 184-8501, Japan, shirose@u-gakugei.ac.jp), Yukiko Nota (ATR-Promotions, Kyoto, Japan), and Tatsuya Kitamura (Konan Univ., Kobe, Japan)

The purpose of this study is to identify physiological contrasts between long and short vowels in Japanese. Japanese (Tokyo Japanese; TJ) is known as a mora-timed language. Durational length of vowels and consonants, short or long, is used to discriminate between words in Japanese; in one example, kabu, "bottom," and ka:bu, "curve," are distinguished by Japanese speakers. Previous studies have discussed the durational contrasts acoustically and phonologically; however, an articulatory explanation was not taken into account. In order to understand the articulatory process of durational contrasts, we analyzed movements of the lips and tongue using an electromagnetic articulography (the NDI Wave Speech Research System). Two TJ speakers were asked to produce minimal pairs of words including short or long vowels (e.g., kabu-ka:bu, koto-ko:to). The position of six sensors attached on their lips and tongue was tracked during the speech. The sensors were placed on the upper and lower lips, right angular of the lip and the tongue dorsum. Analysis of the data revealed that the lower lip and posterior tongue moved downward more in articulating long vowels than short vowels. This result suggested that long vowels caused not only acoustic and phonological lengthening of syllable duration, but also overlapping of articulatory movements.

2pSC39. Contact effects on voice-onset time in Patagonian Welsh. Morgan Sleeper (Linguist, Univ. of California, Santa Barbara, 712 Bolton Walk, Apt. #204, Goleta, CA 93117, msleeper@umail.ucsb.edu)

The effects of language contact extend well beyond lexical borrowings and can include morphosyntactic, phonetic, and phonological changes over time (Thomason & Kaufman 1998). One especially common outcome of long-term contact is phonetic transfer (Matras 2009:222). The Welsh spoken in Patagonia, in close contact with Spanish for the past 150 years, offers one potential example: Jones (1984) observes that younger Patagonian Welsh speakers may be developing unaspirated voiceless stops /p t k/ as a result of Spanish contact. This paper explores this hypothesis through a quantitative study of voice-onset time (VOT) of the Welsh voiceless stops /p t k/ in contemporary conversational speech data from Patagonia and Wales, from male and female speakers in three age groups (0-29, 30-59, and 60+). Results show that the tendencies reported by Jones (1984) have held true, and in fact have generalized to become a feature of Patagonian Welsh for speakers of all ages: Patagonian speakers produce the Welsh stops /p t k/ with significantly shorter VOT values than speakers from Wales. These results shed light on an important distinguishing phonetic feature of this understudied variety of Welsh, as well as the dynamics of language contact in action.

2pSC40. Three-dimensional reduction for three-way contrast: Conversational stops in Korean. Jae-Hyun Sung and Daniel Brenner (Dept. of Linguist, Univ. of AB, Edmonton, AB T6G 2E7, Canada, sung@ualberta.ca)

Phonetic reduction in conversational speech is widely acknowledged across different languages and dialects. Previous experimental studies have reported that spontaneous conversations yield greater variability and reduction in acoustic signals when compared to careful speech. This study investigates the acoustic reduction in intervocalic Korean stops—phonemically lenis, fortis, and aspirated stops—from conversational speech, and adds to the increasing body of reduced speech research. Comparisons of Korean stops from careful and conversational speech confirm greater acoustic variability in conversational speech than in careful speech. Furthermore, conversational stops in Korean show that phonetic reduction occurs in multiple acoustic dimensions. Lenis stops exhibit phonetic reduction represented by both a smaller intensity difference and a higher pitch, fortis stops do so by a higher pitch only, and aspirated stops do so by both a smaller VOT and a higher pitch. Results from this study leave open the possibility of manner-specific strategies in phonetic reduction within a language, and merit further cross-linguistic examination of conversational speech.

2pSC41. A preliminary ultrasound study of apical consonants in stressed and unstressed position in Arrernte. Marija Tabain (La Trobe Univ., La Trobe University, Melbourne, VIC 3086, Australia, m.tabain@latrobe.edu.au)

In this study, we present ultrasound data from four female speakers of the Australian language Arrernte. We focus on the apical consonants—alveolar /t n l/ and retroflex /ʈ ɳ ɭ/—in stressed and unstressed position. Previous articulatory results for the Arrernte stops using electropalatography and electro-magnetic articulography (for two of the speakers in the current study) showed that the tongue is most retracted for the retroflex in unstressed prosodic context, but that both apicals have a more advanced tongue position in stressed context. In addition, jaw position was highest for the alveolar in stressed position, and lowest for the retroflex in unstressed position. Current ultrasound results show that the rear-most portion of the tongue is consistently further back for the retroflex in unstressed position. This result confirms our previous conclusion that the most prototypical retroflex is the one found in the weaker prosodic position. By contrast, the stressed retroflex has a more forward position of the rear-most portion of the tongue, while the stressed and unstressed alveolar consonants show varying patterns depending on speaker and on manner class. Results are discussed in light of the variability that is often found for apical articulations in Australian languages.

2pSC42. Implosives in Jakarta Indonesian. Yu Tanaka (Linguist, Univ. of California, Los Angeles, 405 Hilgard Ave., 3125 Campbell Hall, Los Angeles, CA 90095-1543, yutanaka@ucla.edu)

Some Jakarta Indonesian speakers produce implosives as allophones of the voiced stops /b/ and /d/. Implosives are typically characterized by increasing voicing amplitude during closure, shorter closure duration and/or creaky phonation of the following vowel, but their actual realization shows much variation across languages (see Ladefoged & Maddieson 1996; Lindau 1984). This study investigates the acoustic properties of Indonesian implosives. Thirty-three native speakers of Jakarta Indonesian were recorded. Each speaker produced one hundred words with initial /b/, /d/ or /p/ in different vowel contexts (e.g. /baik/, /biru/) embedded in carrier phrases. A trained phonetician judged whether each case of /b/ and /d/ in their speech was imploded or not. I measured the voicing amplitude and duration of each target stop and the creakiness of the following vowel by means of harmonic amplitude differences such as H1-H2 (Keating *et al.* 2010). A preliminary analysis reveals that stops judged as implosives generally had more increasing voicing amplitude than those judged as modal plosives. The former also had a shorter duration and their following vowels had relatively low H1-H2 and H1-A2 values, which indicates creakier phonation. The analysis also shows that implosivization is more likely to occur before a non-high back vowel.

2pSC43. Hejazi Saudi Arabic pharyngeal approximant ’Ayn involves 4 of 5 correlates of creak, but not positive H1-H2 spectral tilt. James M. Thorne (None, 1890 East-West Rd., 569 Moore, Honolulu, HI 96822, jamesmathiasthorne@yahoo.com)

Five correlates of creaky phonation in the Arabic pharyngeal approximant ’Ayn—/ˤ/—were compared in speech samples of five male native speakers of the Hejazi dialect of Saudi Arabic to determine the degree to which creaky phonation could be a phonetic feature of ’Ayn. Jitter, shimmer, harmonics-to-noise ratio (HNR), F_0 lowering, and positive H_1-H_2 spectral tilt were compared in 360 (total) speech tokens of /a/, /aa/, and /aˤa/ vowel sequences. Acoustic analysis indicated significantly increased jitter, increased shimmer, lower HNR, and lower F_0 values in ’Ayn sequences compared to non-’Ayn sequences, yielding strong evidence of creaky phonation. There was, however, no significant increase in degree of positive H_1-H_2 spectral tilt in ’Ayn sequences compared to non-’Ayn sequences. Positive tilt appeared to occur in free variation across vowel sequence types with many non-’Ayn sequences displaying the positive H_1-H_2 spectral tilt traditionally associated with creak. If increased jitter, increased shimmer, lower HNR, and local F0 lowering *are* correlates of creak in these realizations of ’Ayn, we must consider that creak is a key phonetic feature of ’Ayn for these Arabic speakers and that positive H_1-H_2 spectral tilt may not be a reliable indicator of creak in Hejazi Arabic.

2pSC44. Changes in Iu-Mien prevoicing: A focus on speakers in the middle. Ela Thurgood (California State Univ., Chico, California State University, Chico, CA 95929, ethurgood@csuchico.edu)

The focus is on speakers who come from families of 8 to 10 children and are in the middle being the 4^{th} or the 5^{th} child in the family. This paper provides an acoustic analysis of syllable-initial Iu Mien stops produced by these speakers, comparing their stops with those of the older siblings. The distinction is important in that the oldest siblings are Iu-Mien native speakers and English non-native speakers, while the youngest siblings are English native speakers with only some passive knowledge of Iu-Mien. Those in the middle constitute a bridge between the other two groups. They are bilingual in both Iu-Mien and English; as the interviews attest, they regularly use Iu-Mien with their parents and older siblings, and English with their younger siblings. This study discusses in detail the differences in the VOT productions of /b p p^h/ and /d t t^h/. It shows that the presence or lack of prevoicing divides those in the middle into two groups: those who prevoiced all of the 266 voiced VOTs measured, and those who prevoiced in 25-44%. The percentage of prevoicing is shown to correlate with language competence that influences language transmission.

2pSC45. Voice onset time in Cantonese women across the menstrual cycle. Chi Him Li (Univ. of Hong Kong, Rm. 19B, Block 11, Laguna City, Kowloon, Hong Kong, Hong Kong, jerrylego@hotmail.com)

This study explores the relationship between voice onset time (VOT) and the menstrual cycle in Cantonese women from Hong Kong. While researchers have investigated how fertility may be related to voice pitch, its connection with VOT is much less understood, let alone that in the non-English speaking population. To fill this gap, a production experiment is conducted to look at VOT in eight Cantonese women at two phases of their menstrual cycle (Day 1—5: low estrogen and progesterone level; Day 18—25: high estrogen and progesterone level). Target words contrast in place of articulation (/p, t, k/) and tone (High vs. Low). Results show that VOT is longer during the high fertility phase, in line with previous studies. Subsequently, voice quality of the speech data will be analyzed using standard measures (e.g., H1-H2, H1-A1, jitter, shimmer, and harmonicity) to shed light on how longer VOT may be associated with other feminine phonatory strategies. These results are discussed in relation to current theories of vocal attractiveness.

2pSC46. Talker variability and the use of perceptual cues in Hong Kong Cantonese lingual stop codas. Jonathan C. Yip (Linguist, The Univ. of Hong Kong, Rm. 906, Run Run Shaw Tower, Pokfulam, HK 0000, Hong Kong, yipjonat@hku.hk)

Cantonese inaudibly released oral stop codas, whose place cues occur only before constriction and not during or after constriction, appear to be merging among young-adult speakers (Law, Fung, & Bauer, 2001). Prior work (Yip, 2015) revealed substantial articulatory variation among five talkers, ranging from gestural preservation, to coproduction of lingual-lingual sequences, to gestural loss. Coda contrasts were cued by first- and third-formant loci (both lower into /k/ than into /t/) and preceding vowel duration (longer before /t/ than before /k/). In the present study, 18 Cantonese-speaking listeners’ perception of codas produced by talkers in the prior study was tested in an AXB task in which ambiguous coda productions were judged as more similar to each talker’s best coda /t/ or /k/ production, as assessed with ultrasound tongue imaging. Comparisons of listener sensitivity to cues—based on LMER estimates for change in accuracy with respect to relevant acoustic response variables—with their accuracy rates for individual talkers indicate that listeners who were sensitive to particular acoustic cues gained a perceptual advantage for talkers who produced those cues. However, greater articulatory achievement in the stimuli did not correlate with better perception, except for productions by talker T4, who produced gestural loss.

2pSC47. Effects of phonetic naturalness in decay of Korean vowel harmony. Hayeun Jang (Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, hayeunja@usc.edu)

In this paper, based on Korean vowel qualities measured in the production experiments, I point out that /o/-stems are also actively participating in the decay of Korean vowel harmony, and their speed of decay is much faster than /a/-stems' speed. How the late-starter /o/-stems could have the faster speed in the decay of Korean vowel harmony? The role of phonetic naturalness in language learning is the key. Measured vowel qualities show that /o/ is acoustically much closer to /ʌ/ as "disharmonic" pair than to /a/ as "harmonic" pair. In addition, in terms of articulatory, the raising of /o/ and the backing of /ʌ/ make them more similar. In the production experiments, 14 Seoul-Korean speakers participated (7 males, 7 females). Onset and coda of stems are controlled. For each condition, 1 existing word and 2 nonce words are used. Participants are asked to make a natural conjugation form of a given stem and to produce it three times. Unlike previous corpus and production studies reporting that /o/-stems show no disharmonic forms, the current experimental results show that quality of selected vowels after /o/-stems is similar to /ʌ/, not /a/. Compared to /a/-stems, /o/-stems' suffix vowel qualities are clearly closer to /ʌ/.

2pSC48. Acoustic evidence for prosodic units in Navajo. Kayla Palakurthy (Linguist, Univ. of California, Santa Barbara, 714 Gayley Walk, Apt 204, Goleta, CA 93117, kaylaeisman@umail.ucsb.edu)

Although prosody and intonation are generally considered to be linguistic universals, the few prosodic studies of Navajo, a Southern Athabaskan language spoken in the American Southwest, report a lack of meaningful intonation (McDonough 1999; McDonough 2002). This paper revisits the question of Navajo prosody, presenting the results of a quantitative analysis of acoustic correlates of prosodic constituents in personal narratives. Building on previous studies of Athabaskan prosody (Berez 2011; Lovick and Tuttle 2012), the present work employs mixed-effects linear modeling to examine pitch (F0) reset and syllable and pause duration at different prosodic and syntactic boundaries. Results provide evidence for both intonational units marked by final syllable lengthening and also larger prosodic sentences cued by pauses and pitch reset. Furthermore, the demarcation of prosodic units is statistically more robust than clause boundaries, underscoring the orthogonal relationship between prosodic and syntactic structure. These data suggest that Navajo does have systematic prosody, but the cues may only be identifiable in larger stretches of connected speech, rather than individual words or phrases. Results are compared with other work in the small, but growing typological literature on intonational characteristics of large prosodic units.

2pSC49. A sociophonetic study of the effect of gender on emphatic-plain contrast in Jordanian Arabic. Abdulaziz Alzoubi (Dept. of World Lang. and Cultures, Univ. of Utah, Lang. & Commun. Bldg. 255 S Central Campus Dr., Rm. 1400, Salt Lake City, UT 84112, Aziz.Alzoubi@Utah.edu)

Mixed results are reported in the literature concerning the status of gender, as social factor, as a predictor of the degree of emphatic-plain contrast in Jordanian Arabic. The present study argues that these mixed results could be an outcome paying no or little attention to demographic factors, more specifically; the original regional dialect subjects belonged to before settling in Jordanian cities. The present study examines the effect of gender, in Amman City, on the realization of emphasis in the emphatic coronals /tˤ/ and /sˤ/ and their plain counterparts /t/ and /s/. The study depended on the first three formants of vowels adjacent to the target sounds after normalization for sex differences, Center of Gravity for target sounds, and Voice Onset Time (VOT) for the stops, as the acoustic cues of emphasis. The results from the mixed model, with "speaker" and "word" as random effect, indicated that gender is a reliable predictor of emphasis in Amman City as evident by the results from the first three vowel formants and stops' VOT; while both males and females maintained a significant difference between emphatic and plain sounds, males maintained a greater emphatic-plain contrast than females.

2pSC50. Production and perception of the low vowels in American English. Valerie Fridland (English, Univ. of Nevada, 1664 N. Virginia St., M.S. 0098, Reno, NV 89519, fridland@unr.edu) and Tyler Kendall (Linguist, Univ. of Oregon, Eugene, OR)

Two vowel features are especially pivotal in defining contemporary U.S. dialects: The merger of the low back vowels and the variable realizations of the low front vowel. Several scholars (Bigham 2010, Gordon 2005, Labov, Ash and Boberg 2006) suggest a relationship between the low front and low back vowels such that /æ/ raising and subsequent fronting of /ɑ/ in the North inhibits the tendency toward low back merger. However, little work examines the robustness of this "structural linkage" or whether a similar relationship obtains across different regional varieties. Further, little work examines whether differences in production correlate with differences in regional perception patterns. Here, we compare the low vowel system across U.S. regional dialects and also consider interrelationships (i.e., correlations) between low vowel categories using data from eight fieldsites across the U.S. We then look at what these speakers' perceptions of category shift between the low front and low back vowel tells us about this relationship in production. Our results suggest both that the regional vowel shifts create significantly divergent low vowel systems across regions but also that /æ/ and /ɑ/ indeed show a structural linkage across regions, while /æ/ and /ɔ/ and /ɑ/ and /ɔ/ show no such relationship.

2pSC51. Individual differences in the relation between perception and production and the mechanisms of phonetic imitation. Donghyun Kim and Meghan Clayards (Linguist, McGill Univ., 1085 Dr. Penfield, Rm. 111, Montreal, QC H3A 1A7, Canada, heydonghyun@gmail.com)

This study uses phonetic imitation to understand more about how individuals perceive and produce speech and to explore the link between the two. We used manipulated stimuli with the goal of more directly probing the link and to test (1) whether individual listeners' perceptual cue weights are related to their patterns of phonetic imitation and (2) the underlying mechanisms of phonetic imitation. Twenty-three native speakers of English completed a 2AFC identification task followed by a baseline production and a forced imitation task. Perception stimuli were created from productions of *head* and *had* recorded by a native speaker of English. Seven steps varying in formant frequency (created with TANDEM-STRAIGHT) were crossed with 7 duration steps (PSOLA in PRAAT). Imitation stimuli were a subset of stimuli from the perception task plus extended and shortened vowel durations. Our results suggest that phonetic imitation is mediated in part by a low-level cognitive process involving a direct link between perception and production as evidenced by imitation of all vowel durations. However, this study also suggests that imitation is mediated by a high-level linguistic component, i.e., phonological contrasts, which is a selective rather than an automatic process as indicated by imitation of phonologically relevant formant frequencies.

2pSC52. Phoneme distribution and syllable structure of entry words in the Carnegie Mellon University Pronouncing Dictionary. Byunggon Yang (English Education, Pusan National Univ., 30 Changjundong Keumjunggu, Pusan 609-735, South Korea, bgyang@pusan.ac.kr)

This study explores the phoneme distribution and syllable structure of entry words in the CMU Pronouncing Dictionary to provide linguists with fundamental phonetic data on English word components. Entry words in the dictionary file were syllabified using an R script and examined to obtain the following results: First, English words preferred consonants to vowels in their word components. In addition, monophthongs occurred much more frequently than diphthongs. When all consonants were categorized by manner and place, the distribution indicated the frequency order of stops, fricatives, and nasals according to manner and that of alveolars, bilabials and velars according to place. Second, from the analysis of syllable structure, two-syllable words were most favored, followed by three- and one-syllable words. Of the words in the dictionary, 92.7% consisted of one, two, or three syllables. Third, the English words tended to exhibit discord between onset and coda consonants and between adjacent vowels. Dissimilarity between the last onset and the first coda was found in 93.3% of the syllables, while 91.6% of the adjacent vowels were different. The author concludes that an analysis of the phonetic symbols in a dictionary may lead to a deeper understanding of English word structures and components.

2pSC53. Glottalization, reduction, and acoustic variability in function words in American English. Laura Dilley, Meisam K. Arjmandi, Zachary Ireland (Dept. of Communicative Sci., Michigan State Univ., East Lansing, MI 48824, ldilley@msu.edu), Chris Heffner (Univ. of Maryland, College Park, MD), and Mark Pitt (The Ohio State Univ., Columbus, OH)

Function words vary widely in pronunciation; understanding this variation is essential for accurate cognitive modeling of lexical perception and production, as well as for computer speech applications. This study analyzed acoustic variation in pronunciation of vowel-initial words *a*, *are*, *our*, and *or*, as well as *her*, which has a vowel-initial variant, [ɚ]. Sentences were constructed in which each function word was preceded by a phonetically similar open syllable rhyme, with which the word was expected to sometimes blend spectrally. Twenty-nine participants uttered 100 sentences in a casual speaking style. The degree of clarity of function word onsets was measured, focusing on evidence of onset discontinuities: glottal onset gestures (e.g., low fundamental frequency, F0; amplitude dips) for vowel-initial words *a, are, our,* and *or*, and, for *her*, of /h/ (e.g., root-mean-square amplitude, amplitude differences of first and second harmonics). Results showed that a substantial proportion of the time, function words lacked any detectable acoustic onset discontinuity. When discontinuities did occur before vowel-onset function words, there was a high degree of acoustic variability in those realizations. These findings highlight the challenges for speech *perception* to identify word onsets and/or to overcome variability in recognizing words. [Work supported by NSF Grant BCS-1431063.]

2pSC54. Analysis of acoustic features for speech intelligibility prediction models analysis of acoustic features for speech intelligibility prediction models. Katsuhiko Yamamoto (Graduate School of Systems Eng., Wakayama Univ., Sakaedani 930, Wakayama 640-8510, Japan, s149011@sys.wakayama-u.ac.jp), Toshio Irino (Faculty of Systems Eng., Wakayama Univ., Wakayama, Japan), Toshie Matsui (Graduate School of Systems Eng., Wakayama Univ., Wakayama, Japan), Shoko Araki, Keisuke Kinoshita, and Tomohiro Nakatani (NTT, Souraku-gun, Japan)

We have developed a new model to predict speech intelligibility of synthetic sounds processed by nonlinear speech enhancement algorithms. The model involves two recent auditory models: the dynamic compressive gammachirp (dcGC) auditory filterbank and the speech envelope power spectrum model (sEPSM). The dcGC-sEPSM was compared with commonly used prediction models based on perceptual intelligibility scores of speech sounds enhanced by classic spectral subtraction and state-of-the-art Wiener filtering. As a result, the dcGC-sEPSM predicted the scores better than the coherence SII (CSII), the short-time objective intelligibility (STOI), and the original sEPSM using the gammatone filterbank. There was, however, still inconsistency between the prediction and data. In this work, we show the analysis of acoustic features used in the prediction models. The CSII calculates the magnitude-squared coherence between clean and processed spectra to derive a signal-to-distortion ratio. The STOI calculates the correlation coefficients between the short-time frame vectors of clean and degraded sound at the output of the one-third octave filterbank. The sEPSM calculates the signal-to-noise ratio in the envelope modulation domain at the output of the auditory filterbank. We summarize the methods and discuss desirable features that improve speech intelligibility predictions.

2pSC55. Ultrasound tongue image denoising for comparison of first and second language tongue trajectories. Shusuke Moriya and Yuichi Yaguchi (Univ. of Aizu, Tsuruga Ikkimachi, Aizuwakamatsu, Fukushima 965-8580, Japan, s1190242@gmail.com)

The main purpose of this research is to specify articulation difference between native and non-native speakers by digitizing tongue motions and analyzing the difference between utterances. Differences in tongue motion directly influence speaker's pronunciation; therefore, it may be possible to improve non-native speaker's efficiency of pronunciation practice with the relevant feedback and visualization. It is necessary for comparison of native and non-native speakers' tongue motions to that end, however, normalization is absolutely necessary to remove the influence of anything except tongue motion before comparison, because every person has a unique shape and size. In our previous research, we proposed normalization methods and some speaking errors were picked up automatically from tongue trajectory in ultrasound tongue image space. However, it is necessary to improve method to extract pure tongue trajectory with accuracy. In this paper, ultrasound tongue images are separated to 5x20 or 20x5 strip block. If tongue edge lies on the block, gray scale graduation forming a shape of an arch has occurred on a line of long side of block. We proposed some shape-sensitive filter to cut off an image noise which locates excepting neighborhood of tongue edge. Through our filter, tongue trajectory extracted from ultrasound tongue image could have less noises then previous one. However, some image noises locates on neighborhood of tongue edge are emphasized and inhibited comparison of native and non-native tongue trajectories.

2pSC56. The acoustic counterpart to articulatory resistance and aggressiveness in locus equation metrics and vowel dispersion. Hye-Young Bang (McGill Univ., 1085 Dr. Penfield, Montreal, QC H3A 1A7, Canada, hye-young.bang@mail.mcgill.ca)

Research on locus equation metrics (LEs) tend to take it for granted that vowel space is invariable across consonantal contexts. However, articulation-based studies report a mutual influence between neighboring segments such that segments with greater constraints in dorsal articulation are more resistant to and concurrently more aggressive in coarticulation than those with less constraints (Farnetani, 1990). We examine (1) whether articulatory resistance and aggressiveness can be acoustically captured through LEs and vowel dispersion and (2) how the relationship between LEs and the degree of coarticulation is mediated by vowel dispersion and delay in voicing. These questions are investigated in CV sequences in English –where C is one of /p t s ʃ/ that varies in the articulatory constraints imposed on the tongue dorsum. We manipulated the magnitude of coarticulation and voicing lag by contrastively stressing the target consonants. Our results show that there is a tight relationship between LE slopes and vowel dispersion where articulatory resistance and aggressiveness appear as the mirror image of each other in the acoustic signal. However, effects of hyperspeech simultaneously affect vowel dispersion, voicing delay, and LEs creating confounds. This calls for caution in the use of LE metrics as a measure of coarticulation.

2pSC57. Pitch estimation based on instantaneous robust algorithm for pitch tracking using weighted linear prediction-based complex speech analysis. Wei Shan (Graduate School of Eng. and Sci., Univ. of the Ryukyus, Senbaru 1, Nishihara, Okinawa 903-0213, Japan, funaki@cc.u-ryukyu.ac.jp) and Keiichi Funaki (C&N Ctr., Univ. of the Ryukyus, Nishihara, Okinawa, Japan)

Pitch estimation plays an important role on speech processing such as speech coding, synthesis, recognition, and so on. Although current pitch estimation method performs well under clean condition, the performance deteriorates significantly in noisy environment. For this reason, robust pitch estimation against additive noise is required. Modified auto-correlation method is commonly used as pitch estimation in which LP residual is used to compute the auto-correlation. We have previously proposed pitch estimation methods based on Time-Varying Complex AR (TV-CAR) analysis whose criterion is the weighted correlation of the complex residual obtained by the TV-CAR analysis, sum of the harmonics for the complex residual spectrum, or so on. On the other hand, Azarov *et al.* have proposed an improved method of RAPT (Robust Algorithm for Pitch Tracking) using an instantaneous harmonics that is called IRAPT (Instantaneous RAPT). The IRAPT can perform better estimation than RAPT. Since IRAPT uses band-limited analytic signal to obtain harmonic frequencies, the complex residual signal obtained by the TV-CAR analysis can also be applied to the IRAPT. In this paper, novel pitch estimation method using the instantaneous frequency based on the robust WLP (Weighted Linear Prediction) TV-CAR residual is proposed and evaluated.

2pSC58. The relationship of VOT and F0 contrasts across speakers and words in the German voicing contrast. Hye-Young Bang, Morgan Sonderegger, and Meghan Clayards (McGill Univ., 1085 Dr. Penfield, Montreal, QC H3A 1A7, Canada, hye-young.bang@mail.mcgill.ca)

Recent studies on tonogenesis in progress in Seoul Korean (Kang, 2014; Bang *et al.*, 2015) find that the size of the VOT contrast and the f0 contrast between aspirated and lax stops "trade off" across speakers (e.g., male speakers have greater/smaller VOT/f0 contrasts), as well as words (e.g., different frequencies, following vowel heights). We examine whether this parallelism across speakers and words occurs in a language not undergoing tonogenesis by examining the size of the fortis/lenis contrast in VOT and f0 in German, using speech from the PhonDat corpus (Draxler, 1995). Mixed-effect regression models show that the size of the VOT contrast, but not the f0 contrast, is affected by properties of words (e.g., frequency, vowel height), unlike the parallelism observed in Korean. We further investigated whether VOT/f0 parallelism would be observed across speakers by partialing out linguistic factors affecting VOT/f0, and performing one logistic regression per speaker ($n = 76$) of fortis/lenis class as a function of residualized f0 and VOT. The cue weights of F0 and VOT are negatively correlated across speakers ($r = -0.213$, $p = 0.0213$), suggesting parallelism exists across speakers, but not across words.

2pSC59. Voice activity detection in noisy environment using dynamic changes of speech in a modulation frequency range of 1 to 16 Hz. Suci Dwijayanti and Masato Miyoshi (Graduate School of Natural Sci. and Technol., Kanazawa Univ., Kanazawa, Ishikawa 920-1192, Japan, dwijas@stu.kanazawa-u.ac.jp)

Voice activity detection (VAD) is usually placed at the preprocessing stage of various speech applications. There have been a lot of studies to find acoustic features that are effective in distinguishing speech/non-speech segments on a recorded signal. In this report, we utilize speech characteristics in a modulation frequency range of 1-to-16 Hz, which seem beneficial to avoid misjudgment. The signal amplitudes are evaluated in common logarithms not to miss small changes in speech energy that may have relations with the starting or ending point of an utterance. In order to capture such changes, the first and second derivatives of the signal are calculated. In finding the starting point, the positive second derivatives seem effective. In finding the ending point, the combination of the negative first derivatives and the positive second derivatives seem effective. And these features should be calculated for individual subbands. On these conditions, Deep Neural Network (DNN) is successfully trained to determine the speech/non-speech segments. In this report, we will evaluate this method in comparison with conventional ones for various noisy utterances.

2pSC60. Improvement of sensor for electromagnetic articulograpy to reduce detrimental effects of sensor attachment on articulation. Yukiko Nota (ATR-Promotions, 2-2-2 Hikaridai, Sorakugun Seikacho, Kyoto 619-0288, Japan, ynota@atr.jp), Tatsuya Kitamura (Konan Univ., Kobe, Japan), Michiko Hashi (Prefectural Univ. of Hiroshima, Mihara, Japan), and Hiroaki Hatano (ATR Hiroshi Ishiguro Labs., Kyoto, Japan)

Electromagnetic articulography (EMA) tracks the position and direction of small wired sensors attached inside and outside of the mouth during articulation. In this study, we improved a five degree-of-freedom sensor of the NDI Wave speech research system, which is a type of EMA system, to reduce the effects of sensor cable on articulation. This was done because the thickness and stiffness of the original cable appeared to interfere with natural articulation. The new cable has the diameter of 0.1 mm, which is thinner than the original, and is more flexible. The degree of interference due to sensor attachment to the articulators was evaluated by comparing the extent of distortion of speech produced by four elderly speakers with new and original sensor attached to their articulators. The speakers produced a Japanese vowel sequence and VCV sequences with the two types of sensors attached to their articulators, and three Japanese speech language pathologists evaluated their speech for the degree of speech distortion. The results suggest that the new sensors were much less likely to disturb natural articulation. [This research was partly supported by JSPS KAKENHI Grant Nos. 24652085, 25240026, and 25280066.]

2pSC61. Analysis of voice imitation by professional/non-professional impersonators based on Kullback–Leibler divergence between acoustic models. Koji Iwano and Takuto Horihata (Faculty of Informatics, Tokyo City Univ., 3-3-1 Ushikubo-nishi, Tsuzuki-ku, Yokohama, Kanagawa 224-8551, Japan, iwano@tcu.ac.jp)

Considering the practical use of speaker verification systems, it is important to investigate the effect of spoofing attacks by professional/non-professional voice imitation. This research proposes an analysis method of voice imitation using the distances between three GMM-based acoustic models trained from cepstral features of "impostor's original voice," "target person's voice," and "impostor's imitated voice." The distance measure is defined by the Kullback-Leibler (KL) divergence. The analysis uses Japanese imitated voice produced by a male professional impersonator and six male non-professional impostors. Each impostor imitated five or six target persons who have never been tried to imitate by the impostors before the experiments. The analysis results show that 1) although the non-professional imitators drastically change their voice features by the imitation, the averaged acoustical distance between the imitated and target voice is still large, 2) whereas the professional imitator approaches their voice characteristics towards the target voice; the distance between the imitated and target voice is approximately 70% of the original distance. The experiments of speaker verification using HMM-UBM-based framework show that the professional imitation certainly yields higher equal error rates than that of non-professional imitation.

2pSC62. Probabilistic simulation for analysis of quantal biomechanical-acoustic relations. Ian Stavness, Francois Roewer-Despres (Comput. Sci., Univ. of SK, 110 Sci. Pl., Saskatoon, SK S7N5C9, Canada, ian.stavness@usask.ca), and Bryan Gick (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Acoustic signals that are stable across a range of articulatory parameters have been suggested as an important feature of speech production [Stevens 1989, *J. Phonetics,* **17**, 3-45]. These so-called quantal effects have also been suggested to arise from the biomechanics of the vocal tract [Fujimura 1989, *J. Phonetics*, **17**, 87-90; Gick & Stavness 2013, *Front Psychol* **4**, 977]. Assessment of potential biomechanical-acoustic quantal relations is hampered by the difficulty of measuring biomechanical parameters, such as muscle excitations, during speech production. Computer modeling has been widely used to probe vocal tract biomechanics, but previous modeling studies have been limited to a small number of deterministic simulations [Gick *et al.* 2014, *CMBBE Imag Vis.,* **2**, 217-22]. We propose a novel probabilistic simulation framework in order to assess how variation in speech motor signals manifests in acoustic variation. We use a detailed 3D biomechanical model of the vocal tract coupled to a source-filter acoustics model [Stavness *et al.* 2014, *Siggraph Asia Tech,* 9] in order to generate acoustic output from muscle excitation inputs. Monte Carlo sampling of muscle excitation inputs are used to characterize variation in formant frequencies for vowel production. These large-scale simulations permit us to evaluate the hypothesis that quantal acoustic signals originate from regions of biomechanical stability. If found, quantal biomechanical-acoustic relations would provide a simple, principled mechanism for feedforward control of speech production.

2pSC63. Optimization of topic estimation for the domain adapted neural network language model. Aiko Hagiwara, Hitoshi Ito, Manon Ichiki, Takeshi Mishima (NHK Sci. and Technol. Res. Labs., 1-10-11 Kinuta, Setagaya-ku, Tokyo 157-8510, Japan, hagiwara.a-iy@nhk.or.jp), Akio Kobayashi (NHK Eng. System, Tokyo, Japan), and Shoei Sato (NHK Sci. and Technol. Res. Labs., Tokyo, Japan)

We present a neural network language model adapted for topics fluctuating in broadcast programs. Topic adapted n-gram language models constructed by using latent Dirichlet allocation for topic estimation are widely used. The conventional method estimates topics by separating the corpora into chunks that have few sentences. While the n-gram model uses several preceding words, the recurrent neural network and long short-term memory can learn to store huge amounts of past information in the hidden layers. Consequently, chunks for language models trained by using neural networks may have a longer optimal length than the chunks for language models trained by using the conventional methods. In this paper, the length of chunks and topic estimation process are optimized for the neural network language models. For the topic estimation, k-mean clustering, latent Dirichlet allocation, and word2vec were compared. On the basis of the results of comparison, we designed a neural network language model.

2pSC64. Acoustic features of prosodic boundary by Chinese learners of English as a second language. Yuanyuan Zhang and Hongwei Ding (Ctr. for Cross-Linguistic Processing and Cognition, School of Foreign Lang., Shanghai Jiao Tong Univ., 800 Dong Chuan Rd., Minhang District, Shanghai 200240, China, yyuan.zhang@sjtu.edu.cn)

The production and perception of prosodic boundary have been well studied in English, but the research regarding to English as a second language (L2) is sparse. It has been shown that Chinese and English employ different acoustic features to mark prosodic boundary in production. Thus, this study aims to find out whether the Chinese learners of English employ the same strategy to convey prosodic boundary in English as the native speakers do. Ten pairs of syntactically ambiguous utterances composed of two or three food items were designed, with 10 pairs of filler utterances half randomized together. Ten speakers (5 males and 5 females) from Shanghai who passed the annual national examination CET (College English Test) Band 6 were asked to read the utterances. The results showed that, similar to the native speakers, Chinese learners also use pre-boundary lengthening, pause duration and pitch reset to signal the prosodic boundary in English. However, their pitch contour of the pre-boundary syllable is falling under boundary condition, and is rising under non-boundary condition, which is opposite to that of the native English speakers. This result will have an implication in L2 English teaching and learning.

2pSC65. End-to-end neural network modeling for Japanese speech recognition. Hitoshi Ito, Aiko Hagiwara, Manon Ichiki, Takeshi Mishima, Shoei Sato (NHK Sci. and Technol. Res. Labs., 1-10-11 Kinuta, Setagaya-ku Tokyo, Setagaya-ku 157-8510, Japan, itou.h-ce@nhk.or.jp), and Akio Kobayashi (NHK Eng. System, Tokyo, Japan)

This study proposes end-to-end neural network modeling to adapt direct speech-to-text decoding to Japanese. End-to-end speech recognition systems using deep neural networks (DNNs) are currently being investigated. These systems do not need intermediate phonetic representation. Instead, many of them utilize Recurrent Neural Networks (RNNs) trained by using much more data than ever before. The end-to-end approach makes acoustic models simpler to train. Typically, previous works have dealt with phonogram labels such alphabetic characters. Ideograms such as Kanji, however, make end-to-end speech recognition more complex. A single Kanji can have multiple readings, such as On-yomi (Chinese reading) and Kun-yomi (Japanese reading). In addition, whereas alphabets have at most 100 labels, Japanese has over 2000 labels to predict, such as Kanji, Hiragana, Katakana, the Roman alphabet, digits, and punctuation marks. To resolve this problem, we attempt to make end-to-end neural network modeling allows speech recognition of Japanese without phonetic representation. This method trains RNN and adopts the connectionist temporal classification (CTC) objective function. The proposed method was able to deal with a large amount of character labels. We also analyzed the decoding results and examined ideas for improving the accuracy of word error rate (WER).

2pSC66. Aperiodicity analysis of filled pause in the Corpus of Spontaneous Japanese database. Hideki Kawahara (Wakayama Univ., 930 Sakaedani, 930, Wakayama, Wakayama 640-8510, Japan, kawahara@sys.wakayama-u.ac.jp)

We applied a novel aperiodicity analysis method (Kawahara et al. 2016, SSW9) to CSJ database (Maekawa, 2003, ISCA & IEEE Workshop). The applied method derives the amount of aperiodicity as a time-frequency map, using three staged procedures. The first stage derives a probability map of the fundamental component witout prior information. The second stage finds the most probable trajectory of the fundamental component based on a state transition model and calculates the initial estimate of FO. The final stage refines this initial estimat using FO adaptive time axis warping and instantaneous frequency of each harmonic component. Recursive application of this refinemet procedure provides simultaneous estimation of FO and aperiodicity map. Classification of filled pause using the refined FO trajectories and the aperiodicity maps will be discussed. [This work was supported by JSPS KAKENHI Grant Number JP26284062.]

2pSC67. Spoken keyword detection using recurrent neural network language model. Shuhei Koike and Akinobu Lee (Dept. of Comput. Sci., Nagoya Inst. of Technol., Gokiso-cho, Showa-ku, Nagoya, Aichi 466-8555, Japan, koike@slp.nitech.ac.jp)

Recently, spoken keyword detection (SKD) systems that listen live audio and tries to capture user's utterances with specific keywords has been extensively studied, in order to realize a truly usable hands-free speech interface in our life: "Okay google" in Google products, "Hey, Siri" on Apple products and "Alexa" on Amazon Alexa / Amazon echo. Since the keyword detectors are typically built from large number of actually spoken keywords, they are irreplaceable and the users of such systems are forced to speak only the keyword they defined. On the other hand, a SKD method based on keyword-filler model using generic phoneme model and garbage filler sequence model is promising in that, since the acoustic pattern of the keyword will can be given as phoneme sequence, it is task-dependent and anyone can use his own keyword. In this study, an improvement of the latter method is studied. Recurrent neural network language model (RNNLM) is introduced as linguistic constraint for both filler-filler and filler-keyword instead of N-gram, and experimental result on actual spoken data for a spoken dialogue system showed that our method can improve the keyword detection performance.

2pSC68. Voice activity detection in movies using multi-class deep neural networks. Ikumi Suga, Ryu Yasuhara, Masashi Inoue, and Tetsuo Kosaka (Yamagata Univ., 4-3-16 Jonan, Yonezawa, Yamagata 9928510, Japan, tkm63745@st.yamagata-u.ac.jp)

For automatic classification of movie genres, detection of the presence or absence of speech is useful. Hence, accurate voice activity detection (VAD) techniques are needed. However, it is difficult to detect speech segments accurately because there are many kinds of noises in movies. Recently, deep learning has received increased attention in the speech processing field. Some VAD techniques based on deep neural networks (DNNs) have been proposed for clean speech conditions and showed better performance than conventional methods. The aim of this study is to improve VAD performance for movies by using DNNs. Generally, VAD is considered to deal with a two-class classification problem, i.e., classification of speech and non-speech segments. However, diverse noises in movies make it difficult. Therefore, it is difficult to detect speech segments accurately by using two-class DNNs. To solve this problem, we propose the use of multi-class DNNs for VAD in movies. In the experiments, we evaluated the proposed method for two types of movies, i.e., conversation-centric (CC) and non-conversation-centric (NCC) movies. The proposed multi-class DNNs showed better performance than conventional DNNs. The results showed that the equal error rates (EERs) for CC and NCC were 7.74% and 2.94%, respectively.

2pSC69. Automatic estimation of extra-linguistic information in speech and its integration into recurrent neural network-based language models for speech recognition. Shohei Toyama, Daisuke Saito, and Nobuaki Minematsu (The Univ. of Tokyo, 7-3-1, Hongo, Bunkyoku, Tokyo 1138654, Japan, toyama@gavo.t.u-tokyo.ac.jp)

When one talks with others, he/she often changes the way of lexical choice and the speaking style depending on various contextual factors. For example in Japanese, we often use gender-dependent expressions and, when we speak to elderly people, we use polite expressions. It can be said that in any language, formality or informality of speaking changes drastically depending on the situation where speakers are involved. The aim of automatic speech recognition (ASR) is to convert speech signals into a sequence of words and the above fact indicates that the performance of ASR can be improved by taking the various contextual factors into account in the ASR modules. In this study, at first, we attempt to estimate those contextual factors as para-linguistic or extra-linguistic information and then, to integrate the results into language models based on Recurrent Neural Network (RNN) for speech recognition. In experiments, from an input utterance, i-vector and openSMILE features were extracted to represent speaker identity and speaking style. These acoustically driven features were integrated into the reranking process of RNN-based language models. Reductions of perplexity of the language models were shown to be 1 to 2% relative.

Session 2pSPa

Signal Processing in Acoustics: Compressive Sensing in Acoustics II

Peter Gerstoft, Cochair
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Invited Papers

1:00

2pSPa1. Super-resolution in sound field recording and reproduction based on sparse representation. Shoichi Koyama, Naoki Murata, and Hiroshi Saruwatari (Graduate School of Information Sci. and Technol., The Univ. of Tokyo, Eng. Bldg. 6-140, 7-3-1 Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, shoichi_koyama@ipc.i.u-tokyo.ac.jp)

Sound field recording and reproduction enables us to construct more realistic audio systems. In practical systems, sound pressure is obtained with microphones in a recording area, and then the sound field is reproduced with loudspeakers in a target area. Therefore, a signal conversion algorithm for obtaining the driving signals of the loudspeakers from the signals received by the microphones is necessary. Most of the current methods are based on the sound field analysis and synthesis in the spatial frequency domain. Although these methods make it possible for stable and efficient signal conversion, artifacts originating from spatial aliasing notably occur, which depends on the intervals of microphones and loudspeakers. We have proposed a signal conversion method based on sparse sound field decomposition, which enables sound field recording and reproduction above the spatial Nyquist frequency. By using sparse decomposition algorithms, the sound pressure distribution can be represented using a small number of fundamental solutions of the Helmholtz equation, such as Green's functions. Group sparse signal models are also required for accurate and robust decomposition. In this presentation, we reports on comparisons between several group sparse signal models and decomposition algorithms as well as their relation to reproduction performances.

1:20

2pSPa2. Compressive sensing methods for passive localization and tracking of undersea acoustic sources. Pedro A. Forero and Paul Baxley (Maritime Systems Div., SPAWAR Systems Ctr. Pacific, 535, San Diego, CA 92152, pforero@gmail.com)

Matched-field processing techniques can achieve localization of undersea acoustic sources in both range and depth when sufficient environmental information is available. Unfortunately, these techniques are sensitive to environmental mismatch and often fail when localizing multiple acoustic sources. This work presents a family of acoustic source-localization techniques that similarly to matched-field processing exploit environmental information for localizing acoustic sources in both range and depth. Unique features of these methods are their explicit use of a sparse representation of the source-localization map and ability to model environmental mismatch. Tools from the areas of compressive sensing and mathematical optimization are leveraged for developing computationally tractable solvers that enable fast processing of high-dimensional source-localization maps. These localization techniques are also extended for tracking multiple acoustic sources. In this case, it is possible to exploit the inherent sparsity of the innovations that occur between consecutive source-localization maps to enhance the localization results at a negligible computational cost. Numerical results on experimental data are shown to illustrate the performance of the proposed methods.

1:40

2pSPa3. Robust multi-frequency sparse Bayesian learning: Theory and simulations. Santosh Nannuru (Marine Physical Lab, SIO, UC, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 452, San Diego, CA 92037, snannuru@ucsd.edu), Kay L. Gemba, William S. Hodgkiss, and Peter Gerstoft (Marine Physical Lab, SIO, UC, San Diego, La Jolla, CA)

Compressive sensing based techniques have recently been applied successfully to underwater acoustics problems such as beamforming and matched field processing. Sparse Bayesian Learning (SBL) is one of the fast compressive sensing methods and is formulated using Bayesian statistics. In the SBL framework, source amplitudes are modeled as complex Gaussian random variables with unknown variance. Evidence maximization is performed to estimate the unknown source amplitude variance and source position. A major advantage of SBL over more commonly used methods such as basis pursuit is that it is computationally faster. In this work, we develop a robust sparse Bayesian learning algorithm that can account for model mismatch leading to errors in the replica vector dictionary. Specifically, the likelihood function is modified so that it takes into account the covariance matrix of error in replica vectors. We also

extend the SBL algorithm to process observations from multiple frequencies. The derived update rule combines observations from all the frequencies in a holistic manner. We demonstrate the robust SBL algorithm with simulations. A comparison is done with other compressed sensing methods including basis pursuit and orthogonal matching pursuit.

2:00

2pSPa4. Robust multi-frequency sparse Bayesian learning: Data results. Kay L. Gemba, Santosh Nannuru, Edward Richards, William S. Hodgkiss, and Peter Gerstoft (MPL, Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 446, La Jolla, CA 92037, gemba@ucsd.edu)

Compressive sensing based techniques have been applied successfully to underwater acoustics problems. We present data results using a robust multi-frequency sparse Bayesian learning (SBL) algorithm that can account for model mismatch leading to errors in the replica vector dictionary. We use the SWellEx-96 Event S5 data set to demonstrate SBL capabilities for the beamforming and matched field processing (MFP) application. Beamforming results using the entire 64 element array (design frequency of 400 Hz) and source frequencies ranging from 50 to 400 Hz indicate that multi-frequency SBL outperforms Bartlett and WNC processors in identifying multipath arrivals over the approximately 10 km source track. MFP results in a multiple-source scenario indicate that SBL offers a degree of robustness in the presence of data-replica mismatch when tracking a quiet source. The data-replica mismatch is especially pronounced at the closest point of approach due to array tilt of approximately 2 degrees. Data results further indicate that the two SBL tuning parameters (diagonal loading of the replica vector covariance matrix and number of sources for the algorithm's noise estimate) do not require excessive calibration.

TUESDAY AFTERNOON, 29 NOVEMBER 2016 SOUTH PACIFIC 1, 2:35 P.M. TO 5:40 P.M.

Session 2pSPb

Signal Processing in Acoustics and Underwater Acoustics: Detection, Tracking, and Classification of Unmanned Aircraft and Underwater Vehicles

Geoffrey H. Goldman, Cochair
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R. Lee Culver, Cochair
ARL, Penn State University, PO Box 30, State College, PA 16804

Chair's Introduction—2:35

Invited Papers

2:40

2pSPb1. Some comments on detecting unmanned underwater vehicles using passive acoustics. Gerald L. D'Spain (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu)

Passive methods of detection rely upon the presence of a field radiated by the phenomenon of interest (in this case, unmanned underwater vehicles—UUVs). This talk illustrates the challenges of using passive acoustic methods to detect UUVs. Two types of vehicles are discussed; a 20-ft wingspan, buoyancy-driven, flying wing autonomous underwater glider and a 21-in-diameter, prop-driven UUV. At-sea measurements of the acoustic field radiated by the glider's large (30 liter) buoyancy engine are presented. Whereas a prop-driven propulsion system operates continuously, a buoyancy engine has very low duty cycle: a few percent. Onboard self noise from the glider's internal fluid-based roll control system (which provides control authority near neutral buoyancy) far exceeds that from an aileron-based system. For the 21-in-diameter prop-driven UUV, modifications were made to its propulsion and steering systems to reduce radiated noise by 20-50 dB. The resulting self noise levels recorded at sea by a hull-mounted hydrophone now are below typical background ocean noise levels above 200 Hz, with the noise below 200 Hz being primarily vibration induced. Highly directional passive acoustic receiving systems are required to detect the transits of these UUVs. [Work supported by the Office of Naval Research and BP.]

3:00

2pSPb2. Acoustic detection, localization, and tracking of tactical autonomous aerial and underwater vehicles. Brian G. Ferguson and Kam W. Lo (Maritime Div., DSTG, PO Box 44, Pyrmont, NSW 2009, Australia, Brian.Ferguson@dsto.defence.gov.au)

By applying single sensor narrowband and multisensor broadband passive acoustic signal processing methods to real microphone data, it is shown that low observable tactical autonomous aerial vehicles can be detected, classified, localized, and tracked by unattended land-based acoustic surveillance systems that exploit a target's acoustic signature. Similarly, the application of passive sonar signal processing methods to real hydrophone data reveals the presence, position and track of an autonomous underwater vehicle by exploiting its radiated noise and its underwater acoustic communication transmissions. A suite of these narrowband and broadband acoustic signal processing methods will be identified and their functional performances quantified.

3:20

2pSPb3. A physical simulation of sound propagation from unmanned aerial vehicles. Z. C. Zheng and Junjian Zhang (Aerosp. Eng., Univ. of Kansas, 2118C Learned Hall, 1530 W 15th St., Lawrence, KS 66045, zzheng@ku.edu)

The study is to investigate sound propagation from UAVs to the environment. The sound propagation is simulated based on the physical time and space, which is a time-domain simulation in a physical spatial domain. The investigation will be focused on the physics that is usually difficult to reveal using non-physical space simulation. The effects to be investigated include source characteristics and motion of UAV noise, influence of local environment such as atmospheric turbulence and wind shear,ground roughness and impedance, diffraction caused by local structures such as terrain, buildings, bushes, and trees. Because of proximity of UAVs to the ground, these effects are more significant for sound propagation from UAVs than from large aircraft that are at higher altitude from the ground. Examples of source recognition using the simulation results will also be discussed.

3:40

2pSPb4. Spectral broadening of acoustic tones generated by unmanned aerial vehicles in a turbulent atmosphere. Vladimir E. Ostashev, D. K. Wilson (U.S. Army Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@noaa.gov), Anthony Finn, Kevin Rogers (Defense and Systems Inst., Univ. of South Australia, Mawson Lakes, SA, Australia), and Emre Barlas (Wind Energy, Tech. Univ. of Denmark, Kongens Lyngby, Denmark)

The acoustic spectrum emitted by unmanned aerial vehicles (UAVs) and other aircraft can be distorted by propagation through atmospheric turbulence. Since most UAVs are propeller-based, they generate a series of acoustic tones and harmonics. In this paper, spectral broadening of these tones due to atmospheric turbulence is studied. The broadening results from the combined Doppler effect of multiply scattered acoustic signals propagating in a non-stationary turbulent atmosphere. It can be assessed as a Fourier transform of the temporal coherence function of a monochromatic signal propagating in an atmosphere with spatial-temporal fluctuations in temperature and wind velocity. This temporal coherence was recently investigated [V. E. Ostashev, D. K. Wilson, S. N. Vecherin, and S. L. Collier, J. Acoust. Soc. Am. **136** (5), 2414–2431 (2014)] for the model of locally frozen turbulence. Based on these results, spectral broadening is calculated and analyzed for typical meteorological regimes of the atmospheric boundary layer and different flight trajectories of UAVs. Experimental results are presented and compared with theoretical predictions. Spectral broadening might also provide a means for remotely sensing atmospheric turbulence.

4:00–4:20 Break

Contributed Papers

4:20

2pSPb5. Novel algorithm for acoustic classification of unmanned aircraft systems. Michael Brown, Cory Smith, and Amanda Hanford (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804-0030, mtb5264@psu.edu)

Advances in Unmanned Aircraft System (UAS) technologies have made them smaller and more affordable, causing a proliferation in UAS activity. This increase has led to a rise in UAS-related incidents, exposing the need for stricter UAS regulation. Detection, Localization, and Classification (DLC) of UAS are important techniques for UAS regulation. However, DLC can be challenging, since UAS span a variety of shapes, sizes, speeds, propulsion systems, and operational altitudes; they can be hard to detect with radar; and they can operate in radio silence. Research supporting DLC of UAS has led to the development of novel, passive acoustic signal processing algorithms for real-time classification. In particular, data fusion algorithms combining kinetic and acoustic data into features for classification are being developed. Kinetic data are collected using acoustic signal source tracking algorithms, and spectral content is extracted from frequency-domain analysis. Using free-body modeling and Doppler compensation, classification probabilities can then be calculated for a list of known UAS. Initial classification algorithms have been developed and tested in MATLAB using UAS data collected in operational environments. The test results will also be presented.

4:35

2pSPb6. Application of cross-correlation methods for passive acoustic unmannded aierial vehicle detection and tracking. Alexander Sedunov (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asedunov@stevens.edu), Alexander Sutin (Stevens Inst. of Technol., Summit, NJ), and Hady Salloum (Stevens Inst. of Technol., Glen Ridge, NJ)

Stevens Institute of Technology has applied the cross-correlation method for the detection of low flying aircraft using passive acoustics. A modified cross correlation method was tested for finding the time difference of arrival of acoustic signals from various unmanned aerial vehicles (UAV), collected by NASA's Langley Research Center. Cross-correlation was applied for UAV detection and for measurements of the time difference of arrival that is used for estimating the direction of arrival. The microphone separation combined with the high speed of the target had led to a significant differential Doppler effect leading to de-correlation of the received signals. For the compensation of those effects, Stevens applied a modified de-skewed short-time correlator (DSTC) approach. The detection ability of the cross-correlation method also depends on the cross correlation properties of ambient noise. The collection was performed in a relatively quiet area, to estimate the performance of the cross-correlation method in noisy areas the NASA UAV signatures were combined with ambient noise collected by Stevens. We demonstrate how the noise affects the detection distance. Special attention was paid to the influence of correlated components of the ambient noise. [The authors are grateful to Dr. Cabell (NASA's Langley Research Center) for providing the acoustic data. This work was supported by DHS's S&T Directorate.]

Invited Paper

4:50

2pSPb7. Acoustic detection of small unmanned aerial system using mills cross arrays. Seongil Kim (The 6th R&D Inst., Agency for Defense Development, PO Box 18, Jinhae, Changwon, Kyung-nam 645-600, South Korea, kim_seongil@yahoo.com)

Two mills cross microphone arrays are used to detect and track a small unmanned aerial system (UAS). Each array consists of two 64-channel line arrays which are perpendicular each other. The cross-shape array is to detect targets approaching from any direction and the two-array system is a minimum number for localizing the target. The microphones of the line array are equally spaced as a design frequency of 2 kHz. The received acoustic signal is digitized as a sampling rate of 8 kHz and transferred to a remote signal processor by a radio link in real time. Beam-formed acoustic data are displayed in various ways to detect and localize the moving acoustic target. In this paper, the proto-type acoustic system for UAS detection as well as the experimental results will be discussed. Methods for array beam-forming, tracking and classifying the target and the acoustic characteristics of the target will be briefly mentioned.

Contributed Papers

5:10

2pSPb8. Acoustic detection results for a small unmanned aircraft system extrapolated over range. Geoffrey H. Goldman (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil)

Small unmanned aircraft systems (SUAS) are becoming cheaper with more advanced reconnaissance and surveillance capabilities. Low-cost sensors and countermunitions are needed to defeat this asymmetric threat. The first step is to detect SUAS using low-cost sensors such as an array of microphones with robust signal processing algorithms. An analysis of six detection algorithms was performed using acoustic data generated by a class I SUAS at ranges of 50 to 650 m and measured with a small tetrahedral microphone array with 20-cm length arms. The detection algorithms were based upon the peak power in delay-and-sum, filtered delay-and-sum, and adaptive delay-and-sum beamforming algorithms. To test the performance of the algorithms at longer ranges, the measured signal, modeled as target plus additive noise, was modified. First, the signal from the target was attenuated using Bass's model and spherical attenuation, and then, the noise was adjusted to maintain an unbiased estimate of the measured power spectrum density function. Receiver operation characteristics (ROC) curves were generated and the performance of the algorithms as a function of range was evaluated.

5:25

2pSPb9. Stepped FM acoustic signal design. Edmund J. Sullivan (EJS Consultants, 46 Lawton Brook Ln., Portsmouth, RI 02871, bewegungslos@fastmail.fm)

The linear Frequency Modulated (LFM) signal is a means of providing a broadband signal with a short time duration. One of its main uses is in active sonar, where it provides enhanced target range resolution and improved performance against reverberation. Its generation can be simplified by approximating it using a so-called Stepped Frequency Modulated (SFM) signal, where the pulse is made up of a set of contiguous single frequency pulses. In its simplest form, the frequency of these pulses increases (or decreases) in a linear manner along the time history of the pulse. This paper presents a technique for designing a SFM pulse that best emulates its LFM counterpart in its detection performance. It is shown that if the time-bandwidth product of the LFM is BT, the number of pulses in the SFM that best emulates the LFM is equal to the nearest integer to the square root of BT. Values of N smaller than this reduce the detection performance and values larger than this provide no improvement in detection performance and indeed, can cause some reduction the performance, where this reduction varies erratically as N increases.

Session 2pUW

Underwater Acoustics, Acoustical Oceanography, and Physical Acoustics: Nonlinear Effects in Underwater Acoustics and Bubbles

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Chair's Introduction—1:00

Invited Papers

1:05

2pUW1. Bulk cavitation extent modeling: An energy-based approach. J. J. Esplin (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, jje166@psu.edu), Benjamin J. Kim, Michael P. Kinzel (Appl. Res. Lab., Penn State Univ., State College, PA), and R. L. Culver (Appl. Res. Lab., Penn State Univ., Reston, VA)

Bulk cavitation is a phenomenon that occurs when a negative-pressure or tension wave causes a liquid to rupture, or cavitate, over space. It is a process which causes resident microbubbles to grow to many times their original size, forming a bubble cloud. Such bubble clouds are observed in shallow underwater explosions, where negative-pressure waves are formed after shock waves reflect off the water surface; they are also observed in shock wave lithotripsy, shock wave histotripsy, ultrasonic cleaning, and other applications. Models had been developed for predicting the size and shape of such bulk cavitation regions. This work introduces a model that accounts for energy "lost" to bulk cavitation which in turn influences the extent that is dependent on the rate at which the passing negative-pressure wave dissipates. In-laboratory underwater experiments utilizing a spark source for high-amplitude pressure pulse generation, hydrophones and high-speed videography validate the energy transfer from tension wave to bubble cloud formation. These experiments are supplemented by computational fluid dynamics simulations. A cavitation absorption coefficient is introduced and parameterized for accurate prediction of cloud extent.

1:25

2pUW2. The effects of external acoustic fields on a free-running supercavitating projectile. Peter J. Cameron (School of Mech. Eng., Georgia Inst. of Technol., Cupertino, California), John W. Doane, and Peter H. Rogers (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, peter.rogers@gatech.edu)

Proliferation of supercavitating torpedoes has motivated research on countermeasures against them as well as on the fluid phenomenon which makes them possible. The goal of this research was to investigate an envisaged countermeasure, an acoustic field capable of slowing or diverting the weapon by disrupting the cavitation envelope. The research focused on the interactions between high pressure amplitude sound waves and a supercavity produced by a small free-flying projectile. The flight dynamics and cavity geometry measurements were compared to control experiments and theoretical considerations were made for evaluating the effects. Corrugations on the cavity/water interface caused by the pressure signal have been observed and characterized. Results also show that the accuracy of a supercavitating projectile can be adversely affected by the sound signal. This research concludes with results that indicate that it is acoustic cavitation in the medium surrounding the supercavity, caused by the high pressure amplitude sound, that is responsible for the reduced accuracy. A hypothesis has been presented addressing the means by which the acoustic cavitation could cause this effect. [Much of the content of this paper was published in J. Acoust. Soc. Am. **128,** 3381-3392 (2010).]

1:45

2pUW3. An introduction to the applications and bubble dynamics of the combustive sound source. Andrew R. McNeese (Appl. Res. Labs, UT: Austin, 10000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs, UT: Austin, Austin, TX), and Thomas G. Muir (Appl. Res. Labs, UT: Austin, Austin, TX)

The combustive sound source (CSS) is a versatile underwater sound source with an adjustable output amplitude used in underwater acoustics experiments. Unlike typical impulsive acoustic sources, CSS can maintain a wide bandwidth at low amplitude and can be tuned to operate in a band of interest. The CSS can be used as a source for low-frequency sediment characterization, array calibration, TL measurements, and when deployed on the bottom can produce seismic interface waves. In addition to stationary deployments in the water column, CSS can be deployed in a tow body. The CSS consists of a submersible combustion chamber, open to the water, which is filled with a combustive mixture that is ignited via spark. Upon ignition, the combustive mixture is converted into high temperature combustion byproducts which expand and ultimately collapse back to smaller volume than before ignition. Acoustic pulses are radiated by the bubble activity. Although the far-field acoustic propagation is generally assumed to be linear, the inherent nature of the bubble growth and collapse is a nonlinear phenomenon. Discussion will focus on the latest CSS design including functionality and acoustic output, as well as the nonlinearity of the bubble dynamics associated with a CSS event.

2:05

2pUW4. Two types of propagations of nonlinear sound beams in nonuniform bubbly liquids. Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba 305-8573, Japan, kanagawa@kz.tsukuba.ac.jp)

Weakly nonlinear propagation of diffracted sound beams in nonuniform bubbly liquids is theoretically examined. The spatial distribution of the number density of the bubbles, initially in a quiescent state, is assumed to be a slowly varying function of the spatial coordinates; the amplitude of variation is assumed to be small compared to the mean number density. Two types of nonlinear wave equations for progressive quasi-plane beams in weakly nonuniform bubbly liquids are then systematically derived via the method of multiple scales. The diffraction effect is incorporated by adding a relation that scales the circular sound source diameter to the wavelength into the original set of scaling relations composed of nondimensional physical parameters. A set of basic equations for bubbly liquids is composed of the averaged equations of mass and momentum in a two-fluid model, the Keller equation for bubble wall, the equation of state for gas and liquid, the mass conservation equation inside the bubble, and the balance equation of normal stresses across the gas-liquid interface. As a result, two types of evolution equations, a nonlinear Schrödinger (NLS) equation including dissipation, diffraction, and nonuniform effects for high-frequency short-wavelength case, and a Khokhlov-Zabolotskaya-Kuznetsov (KZK) equation including dispersion and nonuniform effects for low-frequency long-wavelength case, are derived from the basic set. Finally, numerical and analytical solutions of NLS and KZK equations toward some applications are presented.

Contributed Papers

2:25

2pUW5. Low-frequency approximations for the radiation damping of a bubble. Kyle S. Spratt and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX, sprattkyle@gmail.com)

The recent work by Ainslie and Leighton [J. Acoust. Soc. Am. **126**, 2163 (2009)] pointed out that there exist two different expressions in the literature for the scattering cross-section of a spherical gas bubble in liquid. The difference between the two expressions is contained in the term corresponding to the losses due to acoustic radiation. The more common expression, often attributed to R. Wildt ["Acoustic theory of bubbles," in *Physics of Sound in the Sea*], is based on a low-frequency approximation and is thus only accurate when the wavelength of the incident wave is large compared to the circumference of the bubble. The purpose of the present work is to investigate specifically the accuracy of this low-frequency approximation for various physical parameters and frequencies of interest. An alternative low-frequency approximation is also suggested, which is both simpler and more accurate than the present approximation. Finally the case of a non-spherical bubble is considered, and an expression for the scattering cross-section is given that is based on the same low-frequency assumptions made by Wildt. [Work supported by the IR&D program at ARL:UT.]

2:40

2pUW6. Dynamic equilibrium model for stable nano- and micro-bubbles partly covered with hydrophobic material. Kyuichi Yasui, Toru Tuziuti, and Wataru Kanematsu (National Inst. of Adv. Industrial Sci. and Technol. (AIST), 2266-98 Anagahora, Shimoshidami, Moriyama-ku, Nagoya 463-8560, Japan, k.yasui@aist.go.jp)

In the dynamic equilibrium model, a bubble is only partly covered with hydrophobic material in contrast to the skin model that a bubble is completely covered with organic material. On the surface of the hydrophobic material in water, gas is highly concentrated due to the presence of depletion layer. Then, gas diffuses into a bubble near the peripheral edge of the hydrophobic material, which may balance with the gas diffusion out of a bubble from the other part of the uncovered bubble surface due to the higher internal gas pressure than the partial pressure of gas dissolved in the liquid water by the Laplace pressure. In the present study, not only the condition of the mass balance but also that of stability is numerically calculated. The results have indicated that a nanobubble could be stable when the fraction of the surface coverage by the hydrophobic material is more than 0.5 in water both supersaturated and under-saturated with gas (air). In slightly degassed water, a microbubble could be stabilized when the fraction of the surface coverage is on the order of 10^{-4} or less. These bubbles could play an important role in underwater acoustics as well as in acoustic cavitation.

2:55

2pUW7. Phase speed measurements of a bubbly liquid in impedance tubes using a transfer function technique. Stanley A. Cheyne, H O. Thurman, and Cecil M. Tiblin (Dept. of Phys. & Astronomy, Hampden-Sydney College, Hampden-Sydney, VA 23943, scheyne@hsc.edu)

Two different impedance tube configurations were used to make phase speed measurements of a bubbly liquid. One setup consisted of a vertical water-filled tube with hypodermic needles near the top end to provide a bubbly termination. In this setup, the source was at the bottom of the tube. The second setup was a vertical tube partially filled with water with hypodermic needles at the bottom. In this setup, the source was at the top and the measurements were made in air with the bubbly termination at the bottom of the tube. In both cases, an SRS-785 spectrum analyzer was used to measure and calculate the transfer function. The transfer function technique (IS0 10534-2-1998) was used to determine the impedance of the bubbly liquid. Once the impedance was found, the phase speed was calculated. Results will be presented with a range of void fractions and bubble radii.

3:10–3:25 Break

Invited Papers

3:25

2pUW8. Modes of targets in water excited and identified using the radiation pressure of modulated focused ultrasound. Timothy D. Daniel, Philip L. Marston (Phys., WSU, Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ivars Kiersteins (NUWC, Newport, RI), and Ahmad T. Abawi (HLS, Res., La Jolla, CA)

Previously, we excited low frequency flexural modes of a circular plate in water using modulated radiation pressure generated by focused ultrasound [J. Acoust. Soc. Am. **137**, 2439 (2015)]. This effect is distinct from excitations associated with parametric array sonar in that static deformation can be produced by radiation pressure as has been measured and analyzed for bubbles [T. J. Asaki and P. L. Marston, J. Acoust. Soc. Am. **97**, 2138-2143 (1995)]. In our experiments, solid targets are suspended by lines or supported on sand and the modulated ultrasound is focused on the target's surface. Target sound emissions were recorded and a laser vibrometer was used to measure the surface velocity of the target to give the magnitude of the target response. An improved high-power focused transducer allows us to drive modes of larger more-realistic target models. By varying the modulation frequency and monitoring the target response, resonant frequencies of the target can be found. By scanning the point-like driving force, the target mode shapes can be measured and compared to finite element models. [Work supported by the Office of Naval Research.]

3:45

2pUW9. Improved object detection sonar using nonlinear acoustical effects in bubbly media. Kevin M. Lee, Grace A. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Clutter from air bubbles can significantly impact the performance of object detection sonar in shallow water environments where bubbles are present due to breaking waves, wakes, or organic matter. Nonlinear acoustical effects can be excited in bubbly liquids at sound pressures below that required in bubble-free media and hence can be used to discriminate between bubbles and solid objects or to suppress bubble-related clutter altogether. Whereas such effects are widely exploited for ultrasonic biomedical imaging enhancement and it is hypothesized that dolphins exploit them in their biosonar, relatively less attention has been given to their use in naval, commercial, or oceanographic manmade sonar. Here, we describe laboratory tank experiments and modeling efforts aimed at exploiting these effects to improve object detection sonar. A bubble dynamics model was employed to investigate parameter space for nonlinear effects such as subharmonic and higher-order harmonic generation from regions of bubbly liquid and sonar-induced bubble translation. Laboratory experiments were conducted to verify the presence of these effects. Finally, a laboratory demonstration of a nonlinear bubble/object discriminating sonar, in which nonlinear effects are used to place markers on returns from regions of bubbly liquid, will be presented. [Work supported by ONR.]

4:05

2pUW10. Measurement of the structural waves in a mine and low frequency tomographic imaging using a parametric sonar. Ron J. Wyber (Midspar Systems, 24 Farrer Pl., Oyster Bay, NSW 2225, Australia, rjwyber@ozemail.com.au) and Brian G. Feguson (DSTO, Pyrmont, NSW, Australia)

To investigate the structural response of mines at low frequencies a parametric sonar with a horizontal beamwidth of about 5 degrees and a bandwidth from 1 kHz to 10 kHz was used. Advantage was taken of the wide bandwidth to obtain a high resolution in the impulse response measurements of the targets by transmitting a linear period modulated pulse and applying pulse compression and a Kaiser Bessel shading to the reflected signal. Over the frequency response of the measurements structural waves are excited in the mines which clearly identify them as man-made targets. The structural waves observed include flexural waves propagating around the body of the mine which provide information on the mine construction. Results are presented showing the variation in the structural response as the targets are rotated through 360 degrees. Tomographic imaging is applied to the measured impulse response and it is shown that by gating out the structural waves clear images of the mines can be obtained even at these low frequencies. This provides the opportunity to image buried targets, particularly in muddy areas where burial is most likely. Profiling measurements in Sydney Harbour are presented showing the variation in the bottom penetration with the sediment type.

4:25

2pUW11. Parametric sources—From theory to commercial products. Johnny Dybedal (KONGSBERG, Strandveien 1, Stjørdal NO7502, Norway, johnnyd@broadpark.no)

The interest for underwater application of nonlinear acoustic began in the late 1950s and early 1960s with Westervelts idea of using nonlinear effects to generate low frequency sound beams with high directivity; the parametric source. This idea got a lot of attention during the 1960s and 1970s from various scientific institutions. An early experiment with a parametric source was performed by ELAB at the University of Trondheim and Simrad in 1977. In the early 1980s, Bentech started the design of a towed, combined mapping system for bathymetry and detection of buried pipelines. The idea was to do detection while moving along the pipeline, which required a sub-bottom profiler with steerable beams with high directivity. Due to limited space for transducer, a parametric source was the logical choice. A prototype of the parametric sub-bottom profiler was finished in the late 1980s. Following the prototype, the system was commercialized and additional versions were developed covering applications from shallow water, high resolution work to full ocean depth, high penetration work. Today KONGSBERG is a world leading supplier of parametric sub-bottom profilers.

4:45

2pUW12. Investigation of nonlinearity parameter *B/A* of saturated, unconsolidated marine sediments via a statistical approach. Hunki Lee, Eunghwy Noh, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, ssevi2@yonsei.ac.kr), and Oh-Cho Kwon (The 6th R&D Inst. Naval System Directorate Principal Researcher, Agency for Defense Development, Changwon, South Korea)

In this talk, we present a theory and the supporting experiment for the nonlinearity parameter *B/A* of saturated, unconsolidated marine sediments. The model is composed of the quadratic nonlinearity of the equivalent suspension of grains and the interstitial fluid and the nonlinearity resulting from the random grain contacts. To capture the random nature of grain contacts, the number of Hertzian contacts on each grain and the distribution of contact forces between grains are treated statistically. Predictions of *B/A* are compared with measurements performed with the finite amplitude insert-substitution (FAIS) method. Dependence of *B/A* on external compression and the incident sound pressure level is also discussed. [This work was conducted in the Unmanned Technology Research Center (UTRC) sponsored by the Defense Acquisition Program Administration (DAPA) and the Agency for Defense Development (ADD) in the Republic of Korea.]

5:00

2pUW13. Beam characteristics of the truncated parametric array generated by a panel piston radiator. Desen Yang, Hongdao Li, Jie Shi, Shengguo Shi, Haoyang Zhang, Di Li, and Shiyuan Jin (College of Underwater Acoust. Eng., Harbin Eng. Univ., College of Underwater Acoust. Eng., No. 145, Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, leehongdao@hrbeu.edu.cn)

A theoretical model is presented to describe the difference frequency beam characteristics of the truncated parametric array generated by a panel piston radiator. In this model, a low-pass acoustic filter is used to truncate the virtual-source. The truncated virtual-source can be viewed as a volume distributed cylindrical source and conic source within and beyond the Rayleigh distance, respectively. The difference frequency beam characteristics near the truncated position are significantly changed. Numerical simulation results show that the difference frequency beams near the truncated position are substantially sharpened with the distance between the observation point and the truncated position decreasing. Besides, the difference frequency beamwidth near the truncated position can be used to predict the farfield beamwidth of the truncated parametric array.

5:15

2pUW14. Research on the influence of dispersion in the shallow water waveguide on the parametric array. Desen Yang, Di Li, Jie Shi, Shengguo Shi, Haoyang Zhang, and Hongdao Li (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145, Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, donkeysoso@163.com)

In practice, the application of parametric array is limited due to the effect of dispersion in the shallow water waveguide which has a significant impact on the propagation of the difference frequency wave. This paper attempts to provide better understanding of the difference frequency field in the shallow water waveguide by discussing the virtue source of parametric array in such circumstance. In the paper, the Green's function of the shallow water waveguide is employed for calculating the difference frequency field with the help of virtue source in waveguide. It shows that the wavenumber mismatch caused by waveguide dispersion will restrain the accumulation of difference frequency wave and bring about fluctuations during the process of accumulation. Furthermore, this paper discusses the nonlinear interaction with different modes of pump wave based on the former conclusion and reveals pictures to describe the complicated mode structure of the difference frequency field.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, and Wednesday. See the list below for the exact schedule.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Tuesday, 29 November

Committee	Start Time	Room
Acoustical Oceanography	7:30 p.m.	Kahili
Animal Bioacoustics	7:30 p.m.	Coral 2
Architectural Acoustics	7:30 p.m.	Lehua
Engineering Acoustics	7:30 p.m.	South Pacific 4
Physical Acoustics	7:30 p.m.	South Pacific 2
Psychological and Physiological Acoustics	7:30 p.m.	South Pacific 3
Structural Acoustics and Vibration	7:30 p.m.	South Pacific 1

Committees meeting on Wednesday, 30 November

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Coral 1
Musical Acoustics	7:30 p.m.	Kahili
Noise	7:30 p.m.	South Pacific 4
Signal Processing in Acoustics	7:30 p.m.	South Pacific 1
Speech Communication	7:30 p.m.	Coral 4
Underwater Acoustics	7:30 p.m.	Nautilus