Session 4pAA

Architectural Acoustics and Psychological and Physiological Acoustics: Psychoacoustics in Rooms I

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Chair's Introduction—1:00

Invited Papers

1:05

4pAA1. Introduction to "Psychoacoustics in Rooms," and tutorial on architectural acoustics for psychoacousticians. Philip W. Robinson (Media Technol., Aalto Univ., PL 15500, Aalto 00076, Finland, philrob22@gmail.com)

This special session—"Psychoacoustics in Rooms"—was born from the observation that psychoacoustics and room acoustics are often highly interleaved topics. Those researching the former attempt to determine how the hearing system processes sound, including sound from within specific environmental conditions. Practitioners of the latter aim to produce architectural enclosures catered to the auditory system's needs, to create the best listening experience. However, these two groups do not necessarily utilize a common vocabulary or research approach. This session, a continuation of one with the same name held at Acoustics 2012 Hong Kong, is intended to appeal to both types of researchers and bring them towards a common understanding. As such, the first two presentations are basic surveys of each paradigm. This presentation will focus on common architectural acoustic methods that may be of interest or utility to psychoacousticians.

1:25

4pAA2. A tutorial on psychoacoustical approaches relevant to listening in rooms. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

From the year of its founding, members of the Acoustical Society of America have been interested in the question of how the acoustical effects of real-world environments influence the ability of human beings to process sound (Knudsen, "The hearing of speech in auditoriums," JASA **1**(1), 1929). While interest in this topic has been constant, the specialization of those focused on architectural acoustics and those focused on psychological and physiological acoustics has increased. Today, it is easily observed that we are likely to use methods and terminology that may be quite unfamiliar to those discussing a very similar question just down the hall. This presentation will survey a few of the most influential psychoacoustical approaches to the question of how the detection and identification of stimuli differs depending on whether the task is done in a real (or simulated) room as opposed to over headphones or in an anechoic chamber. The goal will be to set the stage for some of the talks to come and to begin a discussion about methods, terminology, and results that will help turn the diverse backgrounds of the participants into a shared resource rather than a barrier to understanding.

1:45

4pAA3. Speech intelligibility in rooms: An integrated model for temporal smearing, spatial unmasking, and binaural squelch. Thibaud Leclère, Mathieu Lavandier (LGCB, Université de Lyon - ENTPE, rue Maurice Audin, Vaulx-en-Velin, Rhônes 69518, France, thibaud.leclere@entpe.fr), and John F. Culling (School of Psych., Cardiff Univ., Cardiff, Wales, United Kingdom)

Speech intelligibility predictors based on room characteristics only consider the effects of temporal smearing of speech by room reflections and masking by diffuse ambient noise. In binaural listening conditions, a listener is able to separate target speech from interfering sounds. Lavandier and Culling (2010) proposed a model which incorporates this ability and its susceptibility to reverberation, but it neglects the temporal smearing of speech, so that prediction only holds for near-field targets. An extension of this model is presented here which accounts for both speech transmission and spatial unmasking, as well as binaural squelch in reverberant environments. The parameters of this integrated model were tested systematically by comparing the model predictions with speech reception thresholds measured in three experiments from the literature. The results showed a good correspondence between model predictions and experimental data for each experiment. The proposed model provides a unified interpretation of speech transmission, spatial unmasking, and binaural squelch.

2:05

4pAA4. Reverberation and noise pose challenges to speech recognition by cochlear implant users. Arlene C. Neuman (Dept. of Otolaryngol., New York Univ. School of Medicine, 550 First Ave., NBV 5E5, New York, NY 10016, arlene.neuman@nyumc.org)

The cochlear implant (CI) provides access to sound for a growing number of persons with hearing loss. Many CI users are quite successful in using the implant to understand speech in ideal listening conditions, but CI users also need to be able to communicate in noisy, reverberant environments. There is a growing body of research investigating how reverberation and noise affect speech recognition performance of children and adults who use cochlear implants. Findings from our own research and research from other groups will be reviewed and discussed.

2:25

4pAA5. Combined effects of amplitude compression and reverberation on speech modulations. Nirmal Kumar Srinivasan, Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, srinivan@ohsu.edu), Paul N. Reinhart, and Pamela E. Souza (Northwestern Univ. and Knowles Hearing Ctr., Evanston, IL)

It is well documented that reverberation in listening environments is common, and that reverberation reduces speech intelligibility for hearing impaired listeners. it has been proposed that multichannel wide-dynamic range compression (mWDRC) in hearing aids can overcome this difficulty. However, the combined effect of reverberation and mWDRC on speech intelligibility has not been examined quantitatively. In this study, 16 nonsense syllables (/aCa/ format) recorded in a double-walled sound booth were distorted using virtual acoustic methods to simulate eight reverberant listening environments. Each signal was then run through a hearing-aid simulation which applied four-channel WDRC similar to that which might be applied in a wearable aid. Compression release time was varied between 12 and 1500 ms. Consonant confusion matrices were predicted analytically by comparing the similarity in the modulation spectra for clean speech and compressed reverberant speech. Results of this acoustical analysis suggest that the consonant error patterns would be strongly influenced by the combination of compression and reverberation times. If confirmed behaviorally and extended to wearable hearing aids, this outcome could be used to determine the optimum compression time for improved speech intelligibility in reverberant environments. [Work supported by NIH R01 DC60014 and R01 DC011828.]

2:45–3:00 Break

3:00

4pAA6. Model of binaural speech intelligibility in rooms. Thomas Brand, Anna Warzybok (Medical Phys. and Acoust., Cluster of Excellence Hearing4All, Univ. of Oldenburg, Ammerländer Heerstr. 114-118, Oldenburg D-26129, Germany, thomas.brand@uni-oldenburg.de), Jan Rennies (Hearing, Speech and Audio Technol., Fraunhofer IDMT , Oldenburg, Germany), and Birger Kollmeier (Medical Phys. and Acoust., Cluster of Excellence Hearing4All, Univ. of Oldenburg, Oldenburg, Germany)

Many models of speech intelligibility in rooms are based on monaural measures. However, the effect of binaural unmasking improves speech intelligibility substantially. The binaural speech intelligibility model (BSIM) uses multi-frequency-band equalization-cancellation (EC), which models human binaural noise reduction, and the Speech-Intelligibility-Index (SII), which calculates the resulting speech intelligibility. The model analyzes the signal-to-noise ratios at the left and the right ear (modeling better-ear-listening) and the interaural cross correlation of target speech and binaural interferer(s). The effect of the hearing threshold is modeled by assuming two uncorrelated threshold simulation noises for both ears. BSIM describes the (binaural) aspects of useful and detrimental room reflections, reverb, and background noise. Especially the interaction of delay time and direction of speech reflections with binaural unmasking in different acoustical situations was modeled successfully. BSIM can use either the binaural room impulse responses of speech and interferers together with their frequency spectra or binaural recordings of speech and noise. A short-term version of BSIM can be applied to modulated maskers and predicts the consequence of dip listening. Aspects of informational masking are not taken into account yet. To model different degrees of informational masking. the SII threshold has to be re-calibrated.

Contributed Papers

3:20

4pAA7. Investigation of speech privacy in high-speed train cabins using a 1:10 scale model. Hansol Lim, Hyung Suk Jang, and Jin Yong Jeon (Architectural Eng., Hanyang Univ., 605-1, Sci. Technol. Bldg., Hang-dang dong, Seon-dong gu, Seoul KS013, South Korea, lim90128@gmail.com)

In this study, a 1:10 scale model was used to evaluate the acoustical parameters and speech transmission indices in high-speed train cabins when the interior design factors are changed to improve speech privacy. The 1:10 scale model materials were selected by considering real measured target factors, such as reverberation time (RT) and speech level (Lp,A,s). The characteristics of the background noise in a high-speed train depend on the train's speed; therefore, recordings of the background noise (LAeq) inside a train were considered in three situations: a stopped train, a train traveling at 100 km/h, and a train traveling at 300 km/h. The values of the STI were reproduced with the background noise levels at each speed using external array speakers with an equalizing filter in the scale model. The shapes and absorptions of chairs and interior surfaces were evaluated using scale modeling.

3:35

4pAA8. Laboratory experiments for speech intelligibility and speech privacy in passenger cars of high speed trains. Sung Min Oh, Joo Young Hong, and Jin Yong Jeon (Architectural Eng., Hanyang Univ., No. 605-1, Science&y Bldg., 222 Wangsimni-ro, Seongdong-gu, Seoul 133791, South Korea, pdpd5@naver.com)

This study explores the speech privacy criteria in passenger cars of high-speed trains. *In-situ* measurements were performed in running trains to analyze the acoustical characteristics of interior noises in train cabins, and laboratory experiments were conducted to determine the most appropriate single-number quantity for the assessment of speech privacy. In the listening tests, the participants were asked to rate (1) speech intelligibility, (2) speech privacy, and (3) annoyance with varying background noises and signal to noise ratio (SNR). From the results of the listening tests, the effects of background noise levels and SNR on the speech privacy and annoyance were examined and the optimum STI and background noise levels in the passenger car concerning both speech privacy and annoyance were derived.

3:50

4pAA9. Some effects of reflections and delayed sound arrivals on the perception of speech and corresponding measurements of the speech transmission index. Peter Mapp (Peter Mapp Assoc., Copford, Colchester CO6 1LG, United Kingdom, petermapp@petermapp.com)

Although the effects of reflections and later arriving sound repetitions (echoes) have been well researched and published over the past 60 years—ranging from Haas, Wallach, and more recently to Bradley & Sato and Toole, their effect on Speech Transmission Index measurements and assessments has only be cursorily studied. Over the past 20 years, the speech Transmission Index (STI) has become the most widely employed measure of potential speech intelligibility for both natural speech and more importantly of Public Address and emergency sound systems and Voice Alarms. There is a common perception that STI can fully account for echoes and late, discrete sound arrivals and reflections. The paper shows this not to be the case but that sound systems achieving high STI ratings can exhibit poor and unacceptable speech intelligibility due to the presence of late sound arrivals and echoes. The finding is based on the results of a series of listening tests and extensive sound system modeling, simulations and measurements. The results of the word score experiments were found to be highly dependent upon the nature of the test material and presentation.

4:05

4pAA10. Effects of room-acoustic exposure on localization and speech perception in cocktail-party listening situations. Renita Sudirga (Health and Rehabilitation Sci. Program, Western Univ., Elborn College, London, ON N6G 1H1, Canada, rsudirga@uwo.ca), Margaret F. Cheesman, and Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., London, ON, Canada)

Given previous findings suggesting perceptual mechanisms counteracting the effects of reverberation in a number of listening tasks, we asked whether listening experience in a particular room can enhance localization and speech perception abilities in cocktail-party situations. Utilizing the CRM stimuli we measured listeners' abilities in: (1) identifying the location of a speech target given a (-22.5^{o}, 0^{o}, $+$22.5^{o}) talker configuration, (2) identifying the target color/number under co-located (0^{o}, 0^{o}, 0^{o}) and spatially-separated (Ő22.5^{o}, 0^{o}, $+$22.5^{o}) configurations. Stimuli were presented in three types of artificial reverberation. All reverberation types had the same relative times-of-arrival and levels of the reflections (T_{60} = 400 ms, C_{50} = 14 dB; wideband) and varied only in the lateral spread of the reflections. Reverberated stimuli were presented via a circular loudspeaker array situated in an anechoic chamber. Listening exposure was varied by mixing or fixing the reverberation type within a block of trials. For the location identification task, exposure benefit decreased with increasing Target-to-Masker Ratio (TMR). No exposure effect was observed in the speech perception task at Ő4 to 10 dB TMRs, except in the separated, narrowest reverberation condition. Results will be discussed in relation to the different nature of the tasks and findings from other studies.

4:20

4pAA11. On the use of a real-time convolution system to study perception of and response to self-generated speech and music in variable acoustical environments. Jennifer K. Whiting, Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., C110 ESC, Brigham Young University, Provo, UT 84606, lundjenny@comcast.net), and Eric J. Hunter (College of Commnication Arts & Sci., Michigan State Univ., East Lansing, MI)

A real-time convolution system has been developed to quickly manipulate the auditory room-acoustical experiences of human subjects. This system is used to study the perception of self-generated speech and music and the responses of talkers and musicians to varying conditions. Simulated and measured oral-binaural room impulse responses are used within the convolution system. Subjects in an anechoic environment experience room responses excited by their own voices or instruments via the convolution system. Direct sound travels directly to the ear, but the convolved room response is heard specialized headphones spaced away from the head. The convolution system, a method for calibrating room level to be consistent across room impulse responses, and data from preliminary testing for vocal effort in various room environments are discussed.

4:35

4pAA12. Use of k-means clustering analysis to select representative head related transfer functions for use in subjective studies. Matthew Neal and Michelle C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mtn5048@psu.edu)

A head related transfer function (HRTF) must be applied when creating auralizations; however, the HRTFs of individual subjects are not typically known in advance. Often, an overall 'average' HRTF is used instead. The purpose of this study was to develop a listening test to identify a 'matched' (best) and 'unmatched' (worst) HRTF for specific subjects, which could be applied to customize auralizations for individual participants. The method of k-means clustering was used to identify eight representative HRTFs from the CIPIC database. HRTFs from 45 subjects' left and right ears in four directions were clustered, which resulted in 56 cluster centers (possible representative HRTFs). A comparative analysis was conducted to determine an appropriate set of HRTFs. These HRTFs were then convolved with pink noise bursts at 00 elevation and various azimuths to sound like the bursts were rotating around a subject's head. A paired comparison test was used where listeners selected the 'most natural' sounding HRTF signal. 'Most natural' was described as coming from the correct directions and located outside the head. The results from the clustering analysis and listening test will be presented, along with a subjective study that incorporated the HRTF listening test. [Work was supported by NSF Grant 1302741.]

Session 4pAB

Animal Bioacoustics: Acoustics as a Tool for Population Structure III

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Contributed Papers

1:15

4pAB1. Improving acoustic time-of-arrival location estimates by correcting for temperature drift in time base oscillators. Harold A. Cheyne, Peter M. Marchetto, Raymond C. Mack, Daniel P. Salisbury, and Janelle L. Morano (Lab of Ornithology, Cornell Univ., 95 Brown Rd., Rm. 201, Ithaca, NY 14850, haroldcheyne@gmail.com)

Using multiple acoustic sensors in an array for estimating sound source location relies on time synchrony among the devices. When independent time synchrony methods—such as GPS time stamps—are unavailable, the precision of the time base in individual sensors becomes one of the main sources of error in synchrony, and consequently increases the uncertainty of location estimates. Quartz crystal oscillators, on which many acoustic sensors base sampling rate timing, have a vibration frequency that varies with temperature f(T). Each oscillator exhibits a different frequency-temperature relationship, leading to sensor-dependent sample rate drift. Our Marine Autonomous Recording Units (MARUs) use such oscillators for their sample rate timing, and they experience variations in temperature of at least 20°C between preparation in air and deployment underwater, leading to sample rate drift over their deployments. By characterizing each MARU's oscillator f(T) function, and measuring the temperature of the MARU during the deployment, we developed a post-processing method of reducing the sample rate drift. When applied to acoustic data from an array of MARUs, this post-processing method resulted in a statistically significant decrease of the mean sample rate drift by a factor of two, and subsequent lower errors in acoustically derived location estimates.

1:30

4pAB2. Acoustic scene metrics for spatial planning. Kathleen J. Vigness-Raposa, Adam S. Frankel, Jennifer Giard, Kenneth T. Hunter, William T. Ellison (Marine Acoust., Inc., 809 Aquidneck Ave., Middletown, RI 02842, kathleen.vigness@marineacoustics.com)

Potential effects of anthropogenic underwater sounds on marine mammals are usually assessed on the basis of exposure to one sound source. Recently published research modeling underwater noise exposure and assessing its impact on marine life has extended the typical single source/single species absolute received level approach to defining exposure in a variety of ways including: relative levels of exposure, such as loudness, signal to noise ratio, and sensation level; metrics for evaluating chronic elevation in background noise; cumulative exposure to multiple and dissimilar sound sources, as well as the potential for animals to selectively avoid a particular source and other behavioral changes. New approaches to managing the overall acoustic scene that account for these issues requires a more holistic and multi-dimensional approach that addresses the relationships among the noise environment, animal hearing and behavior, and anthropogenic sound sources. We present a layered acoustic scene concept that considers each facet of the extended problem. Our exemplar is a seismic survey in the Gulf of Mexico with layers for ambient oceanographic and meteorological noise, shipping, and distant anthropogenic sources in which the exposure is filtered by the animal's hearing filter, sensation level, and nominal loudness of the signal.

1:45

4pAB3. Establishing baselines for cetaceans using passive acoustic monitoring off west Africa. Melinda Rekdahl, Salvatore Cerchio, and Howard Rosenbaum (WCS, 2300 Southern Blvd., The Bronx, New York, NY 10460, mlrekdahl@wcs.org)

Knowledge of cetacean presence in west African waters is sparse due to the remote and logistically challenging nature of working in these waters. Exploration and Production (E&P) activities are increasing in this region; therefore, collecting baseline information on species distribution is important. Previous research is limited although a number of species listed as vulnerable or data deficient by the IUCN red list have been documented. In 2012/2013, we deployed an array of eight Marine Autonomous Recording Units (MARUs) in a series of three deployments, off Northern Angola, targeting Mysticetes (2 kHz SR, continuous) during winter/spring and Odontocetes (32 kHz SR, 20% duty cycled) during summer/autumn. Preliminary results are presented on the temporal and spatial distribution of species identified from automated and manual detection methods. Humpback whales were frequently detected from August through December, with peaks during September/October. During the deployment period, sperm whales and Balaenopterid and Odontocete calls were also detected and possible species will be discussed. Species detections will be used to identify temporal hotspots for cetacean presence and any potential overlap with E&P activities. We recommend that future research efforts include visual and acoustic vessel surveys to increase the utility of passive acoustics for monitoring these populations.

2:00

4pAB4. Behavioral response of select reef fish and sea turtles to mid-frequency sonar. Stephanie L. Watwood, Joseph D. Iafrate (NUWC Newport, 1176 Howell St., Newport, RI 02841, stephanie.watwood@navy.mil), Eric A. Reyier (Kennedy Space Ctr. Ecological Program, Kennedy Space Ctr., FL), and William E. Redfoot (Marine Turtle Res. Group, Univ. of Central Florida, Orlando, FL)

There is growing concern over the potential effects of high-intensity sonar on wild marine species populations and commercial fisheries. Acoustic telemetry was employed to measure movements of free-ranging reef fish and sea turtles in Port Canaveral, Florida, in response to routine submarine sonar testing. Twenty-five sheepshead (*Archosargus probatocephalus*), 28 gray snapper (*Lutjanus griseus*), and 29 green sea turtles (*Chelonia mydas*) were tagged, with movements monitored for a period of up to four months using an array of passive acoustic receivers. Baseline residency was examined for fish and sea turtles before, during, and after the test event. No mortality of tagged fish or sea turtles was evident from the sonar test event. There was a significant increase in daily residency index for both sheepshead and gray snapper at the testing wharf subsequent to the event. No broad-scale movement from the study site was observed during or immediately after the test. One month after the sonar test, 56% of sheepshead, 71% of gray snappers, and 24% of green sea turtles were still detected on receivers located at the sonar testing wharf.

2:15

4pAB5. Quantifying the ocean soundscape at a very busy southern California location. John E. Joseph and Tetyana Margolina (Oceanogr., Naval Postgrad. School, 833 Dyer Rd, Monterey, CA 93943, jejoseph@nps.edu)

The underwater noise environment in the Southern California Bight is highly variable due to the presence of both episodic and persistent contributors to the soundscape. Short-term events have potential for inducing abrupt behavioral responses in marine life while long-term exposure may have chronic influences or cause more subtle responses. Here we identify and quantify various sources of sound over a wide frequency band using a passive acoustic receiver deployed at 30-mi Bank from December 2012 through March 2013. The site is in the eastern portion of the Navy's training range complex and is in close proximity to very active shipping routes. The region has diverse marine habitats and is known for frequent seismic activity. Acoustic data were scanned for anthropogenic, biologic and other natural noise sources up to 100 kHz. In addition, ancillary databases and data sets were used to verify, supplement and interpret results. Acoustic propagation models were used to explain ship-induced noise patterns. Results indicate that long-term trends in soundscapes over regional-scale areas can be accurately estimated using a combination of tuned acoustic modeling and recurrent *in-situ* data for validation. [Project funded by US Navy.]

2:30

4pAB6. Machine learning an audio taxonomy: Quantifying biodiversity and habitat recovery through rainforest audio recordings. Tim Treuer (Ecology and Evolutionary Biology, Princeton Univ., Princeton, NJ), Jaan Altosaar, Andrew Hartnett (Phys., Princeton Univ., 88 College Rd. West, Princeton, NJ 08544, altosaar@princeton.edu), Colin Twomey, Andy Dobson, David Wilcove, and Iain Couzin (Ecology and Evolutionary Biology, Princeton Univ., Princeton, NJ)

We present a set of tools for semi-supervised classification of ecosystem health in Meso-American tropical dry forest, one of the most highly endangered habitats on Earth. Audio recordings were collected from 15-year-old, 30-year-old and old growth tropical dry forest plots in the Guanacaste Conservation Area, Costa Rica, on both nutrient rich and nutrient poor soils. The goals of this project were to classify the overall health of the regenerating forests using markers of biodiversity. Semi-supervised machine learning and digital signal processing techniques were explored and tested for their ability to detect species and events in the audio recordings. Furthermore, multi-recorder setups within the same vicinity were able to improve detection rates and accuracy by enabling localization of audio events. Variations in species' and rainforest ambient noise detection rates over time were hypothesized to correlate to biodiversity and hence the health of the rainforest. By comparing levels of biodiversity measured in this manner between old growth and young dry forest plots, we hope to determine the effectiveness of reforestation techniques and identify key environmental factors shaping the recovery of forest ecosystems.

2:45–3:00 Break

3:00

4pAB7. Sound-based automatic neotropical sciaenid fishes identification: *Cynoscion jamaicensis*. Sebastian Ruiz-Blais (Res. Ctr. of Information and Commun. Technologies, Universidad de Costa Rica, Guadalupe, Goicoechea, San José 1385-2100, Costa Rica, ruizble@yahoo.com), Arturo Camacho (School of Comput. Sci. and Informatics, Universidad de Costa Rica, San José, Costa Rica), and Mario R. Rivera-Chavarria (Res. Ctr. of Information and Commun. Technologies, Universidad de Costa Rica, San José, Costa Rica)

Automatic software for sciaenid sound emissions identification are scarce. We present a method to automatically identify sound emissions produced by the sciaenid *Cynoscion jamaicensis*. The emissions of *C. jamaicensis* typically have a 24 Hz pulse repetition rate and a quasi-harmonic pattern in their spectra with a pitched quality in its sound. The proposed method is an adaptation of a previous method proposed to detect sounds of *Cynoscion squamipinnis* in recordings. It features long-term partial loudness, pulse repetition rate, pitch strength, and timbre statistics. The satisfactory results of 0.9 in the F-measure show that the method generalizes well over species, considering the different characteristics of *C. jamaicensis* and *C. squamipinnis*. Future research is required to test the method with other species recordings, in order to further evaluate its robustness.

3:15

4pAB8. Examining the impact of the ocean environment on cetacean classification using theocean acoustics and seismic exploration synthesis (OASES) propagation model. Carolyn M. Binder and Paul C. Hines (Defence R&D Canada , P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, carolyn.binder@drdc-rddc.gc.ca)

Passive acoustic monitoring (PAM) is now in wide use to study cetaceans in their natural habitats. Since cetaceans can be found in all ocean basins, their habitats cover diverse underwater environments. Properties of the ocean environment such as the sound speed profile, bathymetry, and sediment properties can be markedly different between these diverse environments. This leads to differences in how a cetacean vocalization is distorted by propagation effects and may impact the accuracy of PAM systems. To develop an automatic PAM system capable of operating effectively under numerous environmental conditions one must understand how propagation conditions affect these systems. Previous effort using a relatively limited data set has shown that a prototype aural classifier developed at Defence R&D Canada can be used to reduce false alarm rates and successfully discriminate cetacean vocalizations from several species. The aural classifier achieves accurate results by using perceptual signal features that model the features employed by the human auditory system. The current work uses the OASES pulse propagation model to examine the robustness of the classifier under various environmental conditions; preliminary results will be presented from cetacean vocalizations that were transmitted over several ranges through environments modeled using conditions measured during experimental trials.

3:30

4pAB9. Acoustic detection, localization, and tracking of vocalizing humpback whales on the U.S. Navy's Pacific Missile Range Facility. Tyler A. Helble (SSC-PAC, 2622 Lincoln Ave., San Diego, CA 92104, tyler.helble@gmail.com)

A subset of the 41 deep water broadband hydrophones on the U.S. Navy's Pacific Missile Range Facility (PMRF) to the northwest of Kauai, Hawaii was used to acoustically detect, localize, and track vocalizing humpback whales as they transited through this offshore range. The focus study area covers 960 square kilometers of water (water depths greater than 300 m and more than 20 km offshore). Because multiple animals vocalize simultaneously, novel techniques were developed for performing call association in order to localize and track individual animals. Several dozen whale track lines can be estimated over varying seasons and years from the hundreds of thousands of recorded vocalizations. An acoustic model was used to estimate the transmission loss between the animal and PMRF hydrophones so that source levels could be accurately estimated. Evidence suggests a Lombard effect: the average source level of humpback vocalizations changes with changes in background noise level. Additionally, song bout duration, cue (call) rates, swim speeds, and movement patterns of singing humpback whales can be readily extracted from the track estimates. [This work was supported by Commander U.S. Pacific Fleet, the Office of Naval Research, and Living Marine Resources.]

3:45

4pAB10. Determining the detection function of passive acoustic data loggers for porpoises using a large hydrophone array. Jens C. Koblitz, Katharina Brundiers, Mario Kost (German Oceanogr. Museum, Katharinenberg 14-20, Stralsund 18439, Germany, Jens.Koblitz@meeresmuseum.de), Louise Burt, Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom), Jamie MacAulay (Sea Mammal Res. Unit, Univ. of St. Andrews, St. Andrews, United Kingdom), Cinthia T. Ljungqvist (Kolmarden Wildlife Park, Kolmarden, Sweden), Lonnie Mikkelsen (Dept. of BioSci., Aarhus Univ., Roskilde, Denmark), Peter Stilz (Freelance Biologist, Hechingen, Germany), and Harald Benke (German Oceanogr. Museum, Stralsund, Germany)

Click loggers such as C-PODs are an important tool to monitor the spatial distribution and seasonal occurrence of small odontocetes. To determine absolute density, information on the detection function, the detection probability as a function of distance, and derived from this, the effective detection radius (EDR), is needed. In this study a 15 channel hydrophone array, deployed next to 12 C-PODs, was used to localize porpoises and determine their geo-referenced swim paths using the ship's GPS and motion sensors. The detection function of C-PODs was then computed using the distance between the animals and each C-POD. In addition to this, the acoustic detection function of C-PODs has been measured by playing back porpoise-like clicks using an omni-directional transducer. The EDR for these porpoise-like clicks with a source level of 168 dB re 1 μPa pp varied from 41 to 243 m. This variation seemed to be related to the sensitivity of the devices; however, season and water depth also seemed to have an influence on detectability.

4:00

4pAB11. Variations of soundscape in a shallow water marine environment for the Chinese white dolphin. Shane Guan (Dept. of Mech. Eng., The Catholic Univ. of America, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov), Tzu-Hao Lin, Lien-Siang Chou (Inst. of Ecology and Evolutionary Biology, National Taiwan Univ., Taipei, Taiwan), and Joseph F. Vignola (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, MD)

For acoustically oriented animals, sound field can either provide or mask critical information for their well-being and survival. In addition, understanding the variations of the soundscape in the Chinese white dolphin habitat is important to monitoring the relationship between human activities, calling fish, and dolphins, thus assist in coastal conservation and management. Here, we examined the soundscape of a critically endangered Chinese white dolphin population in two shallow water areas next to western coast of Taiwan. Two recording stations were established at Yunlin, which is close to an industrial harbor, and Waisanding, which is nearby a fishing village, in summer 2012. Site specific analyses were performed on variations of the temporal and spectral acoustic characteristics for both locations. The results show different soundscapes for the two sites from different recurring human activities. At Yunlin, high acoustic energy was usually dominated by cargo ships producing noise below 1 kHz. At Waisanding, much higher frequency noise, up to 16 kHz produced by passing fishing boats were detected. In addition, a diurnal cycle of the acoustic field between 1200 and 2600 Hz was observed. It is established that this sound was produced by fish chorus that were observed in both locations.

4:15

4pAB12. Anthropogenic noise has a knock-on effect on the behavior of a territorial species. Kirsty E. McLaughlin and Hansjoerg P. Kunc (School Biological Sci., Queens Univ. Belfast, 97 lisburn Rd., MBC, Belfast bt9 7gt, United Kingdom, kmclaughlin23@qub.ac.uk)

Noise pollution has been shown to induce overt behavioral changes such as avoidance of a noise source and changes to communication behavior. Few studies however have focused on the more subtle behaviors within an individual's repertoire such as foraging and territoriality. Many species are territorial making it unlikely they will leave a noisy area. The impact of noise on essential behaviors of such species must be examined. It has been suggested that a noise induced increase in sheltering behavior will decrease time available for other activities. To test for this potential knock-on effect, we exposed a territorial fish to noise of differing sound pressure levels (SPL). We found that exposure to noise increased sheltering behavior and decreased foraging activity. However, we found that these behavioral responses did not increase with SPL. Furthermore we demonstrate, for the first time experimentally, that noise has a negative knock-on effect on behavior as a noise induced increase in sheltering caused a decrease in foraging activity. This novel finding highlights the importance of examining less overt behavioral changes caused by noise, especially in those species unlikely to avoid a noisy area, and suggests the impacts of noise on animals may be greater than previously predicted.

4:30

4pAB13. Female North Atlantic right whales produce gunshot sounds. Edmund R. Gerstein (Psych., Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33486, gerstein2@aol.com), Vailis Trygonis (FAU / Harbor Branch Oceanogr. Inst. , Lesvos island, Greece), Steve McCulloch (FAU / Harbor Branch Oceanogr. Inst. , Fort Pierce, FL), Jim Moir (Marine Resources Council , Stuart, FL), and Scott Kraus (Edgerton Res. Lab., New England Aquarium, Bostopn, MA)

North Atlantic right whales (*Eubalaena glacialis*) produce loud, broadband, short duration sounds referred to as gunshots. The sounds have been hypothesized to function in a reproductive context, as sexual advertisement signals produced by solitary adult males to attract females and/or agonistic displays among males in surface active groups. This study provides evidence that gunshot sounds are also produced by adult females and examines the acoustics and behavioral contexts associated with these calls. Results from boat-based observational surveys investigating the early vocal ontogeny and behavior of right whales in the critical southeast calving habitat are presented for a subset of mothers who produced gunshots while in close proximity to their calves. Of 26 different isolated mother-calf pairs, gunshots were recorded from females of varied ages and maternal experience. The signals were recorded when calves separated from their mothers during curious approaches toward objects on the surface. While the spectral and temporal characteristics of female gunshots resemble those attributed to adult males, these calls were orders of magnitude quieter (Ő30 dB). Relatively quiet gunshots posed minimal risk of injury to nearby calves. The social and behavioral context suggests gunshots were associated with maternal communication and may also be indicators of stress and agitation.

4:45

4pAB14. Classifying humpback whale individuals from their nocturnal feeding-related calls. Wei Huang, Fan Wu (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., 302 Stearns, Boston, MA 02115, weihece@gmail.com), Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

A large number of humpback whale vocalizations, comprising of both songs and non-song calls, were passively recorded on a high-resolution towed horizontal receiver array during a field experiment in the Gulf of Maine near Georges Bank in the immediate vicinity of the Atlantic herring spawning ground from September to October 2006. The non-song calls were highly nocturnal and dominated by trains of "meows," which are downsweep chirps lasting roughly 1.4 s in the 300 to 600 Hz frequency range, related to night-time foraging activity. Statistical temporal-spectral analysis of the downsweep chirps from a localized whale group indicate that these "meows"can be classified into six or seven distinct types that occur repeatedly over the nighttime observation interval. These meows may be characteristic of different humpback individuals, similar to human vocalizations. Since the "meows" are feeding-related calls for night-time communication or prey echolocation, they may originate from both adults and juveniles of any gender; whereas songs are uttered primarily by adult males. The meows may then provide an approach for passive detection, localization and classification of humpback whale individuals regardless of sex and maturity, and be especially useful for night-time and/or long range monitoring and enumeration of this species.

Session 4pBAa

Biomedical Acoustics: Biomedical Applications of Low Intensity Ultrasound II

Thomas L. Szabo, Chair
Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215

Contributed Papers

1:00

4pBAa1. Investigation of effects of ultrasound on dermal wound healing in diabetic mice. Denise C. Hocking (Pharmacology and Physiol., Univ. of Rochester, 601 Elmwood Ave., Box 711, Rochester, NY 14642, denise_hocking@urmc.rochester.edu), Carol H. Raeman, and Diane Dalecki (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Chronic wounds, including diabetic, leg, and pressure ulcers, impose a significant health care burden worldwide. Currently, chronic wound therapy is primarily supportive. Ultrasound therapy is used clinically to promote bone healing and some evidence indicates that ultrasound can enhance soft tissue repair. Here, we investigated effects of ultrasound on dermal wound healing in a murine model of chronic wounds. An ultrasound exposure system and protocol were developed to provide daily ultrasound exposures to full-thickness, excisional wounds in genetically diabetic mice. Punch biopsy wounds were made on the dorsal skin and covered with acoustically transparent dressing. Mice were exposed to 1-MHz pulsed ultrasound (2 ms pulse, 100 Hz PRF, 0–0.4 MPa) for a duration of 8 min per day. Mice were exposed on 10 days over a 2-week period. No significant differences in the rate of re-epithelialization were observed in response to ultrasound exposure compared to sham-exposed controls. However, two weeks after injury, a statistically significant increase in granulation tissue thickness at the wound center was observed in mice exposed to 0.4 MPa (389 +/ 85 μm) compared to sham exposures (105 +/Ő 50 μm). Additionally, histological sections showed increased collagen deposition in wounds exposed to 0.4 MPa compared to shams.

1:15

4pBAa2. Evaluation of sub-micron, ultrasound-responsive particles as a drug delivery strategy. Rachel Myers, Susan Graham, James Kwan, Apurva Shah, Steven Mo, and Robert Carlisle (Inst. of Biomedical Eng., Univ. of Oxford, Dept. of Eng. Sci., ORCRB, Headington, Oxford OX3 7DQ, United Kingdom, rachel.myers@eng.ox.ac.uk)

Substantial portions of tumors are largely inaccessible to drugs due to their irregular vasculature and high intratumoral pressure. The enhanced permeability and retention effect causes drug carriers within the size range of 100–800 nm to passively accumulate within tumors; however, they remain localized close to the vasculature. Failure to penetrate into and throughout the tumor ultimately limits treatment efficacy. Ultrasound-induced cavitation events have been cited as a method of stimulating greater drug penetration. At present, this targeting strategy is limited by the difference in size between the nano-scale drug carriers used and the cavitation nuclei available, i.e., the micron-scale contrast agent SonoVue. *In vivo* this results in spatial separation of the two agents, limiting the capacity for one to impact upon the other. Our group has successfully formulated two different monodisperse suspensions of nanoparticles that are of a size that will permit better co-localization of cavitation nuclei and therapeutics. A mixture of these nanoparticles and a model drug carrier were passed through a tissue mimicking phantom to provide an *in vitro* simulation of flow through a tumor. The impact of ultrasound on the penetration of drug carrier from the flow channel was compared between both of our ultrasound-responsive particles and SonoVue.

1:30

4pBAa3. Temperature effects on the dynamics of contrast enhancing microbubbles. Faik C. Meral (Radiology, Brigham and Women's Hospital, 221 Longwood Ave., EBRC 521, Boston, MA 02115, fcmeral@bwh.harvard.edu)

Micron-sized, gas encapsulated bubbles are used as ultrasound contrast enhancing agents to improve diagnostic image quality. These microbubbles, which are vascular agents, undergo linear and non-linear oscillations when excited. It is this non-linear response of microbubbles, that helps to distinguish between signals from the tissue -mostly linear-, and signals from the bubbles,nonlinear, which represents vasculature. This opens up to numerous clinical applications such as echocardiography, focal lesion identification, perfusion imaging, etc. Characterization studies of microbubbles gained importance as the possible clinical applications increase. One aspect that these studies focused on is the temperature dependence of the microbubble dynamics. However, these studies were mostly comparing bubble dynamics at room temperature to their dynamics at the physiological temperatures. This study is focused on the changes in the bubble characteristics as a function of temperature. More specifically microbubble attenuation and scattering is measured as a function of temperature and time. Additionally, estimating the temperature changes from the changes in the bubble dynamics is considered as an inverse problem.

1:45

4pBAa4. Response to ultrasound of two types of lipid-coated microbubbles observed with a high-speed optical camera. Tom van Rooij, Ying Luan, Guillaume Renaud, Antonius F. W. van der Steen, Nico de Jong, and Klazina Kooiman (Dept. of Biomedical Eng., Erasmus MC, Postbus 2040, Rotterdam 3000 CA, Netherlands, t.vanrooij@erasmusmc.nl)

Microbubbles (MBs) can be coated with different lipids, but exact influences on acoustical responses remain unclear. The distribution of lipids in the coating of homemade MBs is heterogeneous for DSPC and homogeneous for DPPC-based MBs, as observed with 4Pi confocal microscopy. In this study, we investigated whether DSPC and DPPC MBs show a different vibrational response to ultrasound. MBs composed of main lipid DSPC or DPPC (2 C-atoms less) with a C_4F_{10} gas core, were made by sonication. Microbubble spectroscopy was performed by exciting single MBs with 10-cycle sine wave bursts having a frequency from 1 to 4 MHz and a peak negative pressure of 10, 20, and 50 kPa. The vibrational response to ultrasound was recorded with the Brandaris 128 high-speed camera at 15 Mfps. Larger acoustically induced deflation was observed for DPPC MBs. For a given resting diameter, the resonance frequency was higher for DSPC, resulting in higher shell elasticity of 0.26 N/m as compared to 0.06 N/m for DPPC MBs. Shell viscosity was similar (~10^{-8} kg/s) for both MB types. Non-linear behavior was characterized by the response at the subharmonic and second harmonic frequencies. More DPPC (71%) than DSPC MBs (27%) showed subharmonic response, while the behavior at the second harmonic frequency was comparable. The different acoustic responses of DSPC and DPPC MBs are likely due to the choice of the main lipid and the corresponding spatial distribution in the MB coating.

2:00

4pBAa5. Quantitative acoustic microscopy at 250 MHz for unstained *ex vivo* assessment of retinal layers. Daniel Rohrbach (Lizzi Ctr. for Biomedical Eng., Riverside Res. Inst., 156 William St., 9th Fl., New York City, NY 11215, drohrbach@RiversideResearch.org), Harriet O. Lloyd, Ronald H. Silverman (Dept. of Ophthalmology, Columbia Univ. Medical Ctr., New York City, NY), and Jonathan Mamou (Lizzi Ctr. for Biomedical Eng., Riverside Res. Inst., New York City, NY)

Few quantitative acoustic microscopy (QAM) investigations have been conducted on the vertebrate retina. However, quantitative assessment of acoustically-related material properties would provide valuable information for investigating several diseases. We imaged 12-μm sections of deparaffinized eyes of rdh4 knockout mice (N = 3) using a custom-built acoustic microscope with an F-1.16, 250-MHz transducer (Fraunhofer IBMT) with a 160-MHz bandwidth and 7-μm lateral beamwidth. 2D QAM maps of ultrasound attenuation (UA) and speed of sound (SOS) were generated from reflected signals. Scanned samples then were stained using hematoxylin and eosin and imaged by light microscopy for comparison with QAM maps. Spatial resolution and contrast of QAM maps of SOS and UA were sufficient to resolve anatomic layers within the 214 μm thick retina; anatomic features in QAM maps corresponded to those seen by light microscopy. UA was significantly higher in the outer plexiform layer (420$\pm$70 dB/mm) compared to the inner nuclear layer (343$\pm$22 dB/mm). SOS values ranged between 1696$\pm$56 m/s for the inner nuclear layer and 1583$\pm$42 m/s for the inner plexiform layer. To the authors' knowledge, this study is the first to assess the UA, and SOS of retina layers of vertebrate animals at high frequencies. [NIH Grant R21EB016117 and Core Grant P30EY019007.]

2:15

4pBAa6. Acoustic levitation of gels: A proof-of-concept for thromboelastography. Nate Gruver and R. Glynn Holt (Dept. of Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, Nate_Gruver@buacademy.org)

Current thromboelastography in the clinic requires contact between the measurement apparatus and the blood being studied. An alternative technique employs levitation of a small droplet to limit contact with the blood sample to air alone. As has been demonstrated for Newtonian liquid drops, the measurement of static spatial location and sample deformation can be used to infer sample surface tension. In the current study, ultrasonic acoustic levitation was used to levitate viscoelastic samples. Gelatin was used as a stand-in for blood to establish the validity of the ultrasonic levitation technique on viscoelastic materials. Liquid data was first taken to benchmark the apparatus, then deformation/location studies were performed on set and setting gelatin gels. Relationships between gelling time, gel concentration, and gel firmness were demonstrated. The elastic modulus of gels was inferred from the data using an idealized model.

2:30

4pBAa7. Numerical simulations of ultrasound-lung interaction. Brandon Patterson (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, awesome@umich.edu), Douglas L. Miller (Radiology, Univ. of Michigan, Ann Arbor, MI), David R. Dowling, and Eric Johnsen (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Lung hemorrhage (LH) remains the only bioeffect of non-contrast, diagnostic ultrasound (DUS) proven to occur in mammals. While DUS for lung imaging is routine in critical care situations, a fundamental understanding of DUS-induced LH remains lacking. The objective of this study is to numerically simulate DUS-lung interaction to identify potential damage mechanisms, with an emphasis on shear. Experimentally relevant ultrasound waveforms of different frequencies and amplitudes propagate in tissue (modeled as water) and interact with the lung (modeled as air). Different length scales ranging from single capillaries to lung surface sizes are investigated. For the simulations, a high-order accurate discontinuity-capturing scheme solves the two-dimensional, compressible Navier-Stokes equations to obtain velocities, pressures, stresses and interface displacements in the entire domain. In agreement with theoretical acoustic approximations, small interface displacements are observed. At the lung surface, shear stresses indicative of high strains rates develop and are shown to increase nonlinearly with decreasing ratio of interface curvature to ultrasonic wavelength.

THURSDAY AFTERNOON, 8 MAY 2014 — BALLROOM E, 3:00 P.M. TO 5:30 P.M.

Session 4pBAb

Biomedical Acoustics: Modeling and Characterization of Biomedical Systems

Diane Dalecki, Chair
Biomedical Eng., Univ. of Rochester, 310 Goergen Hall, P.O. Box 270168, Rochester, NY 14627

Contributed Papers

3:00

4pBAb1. Green's function-based simulations of shear waves generated by acoustic radiation force in elastic and viscoelastic soft tissue models. Yiqun Yang (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), Matthew W. Urban (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN), and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw, 2120 Eng. Bldg., East Lansing, MI 48824, mcgough@egr.msu.edu)

The Green's function approach describes propagating shear waves generated by an acoustic radiation force in elastic and viscoelastic soft tissue. Calculations with the Green's function approach are evaluated in elastic and viscoelastic soft tissue models for a line source and for a simulated focused beam. The results for the line source input are evaluated at 200 time samples in a 101 by 101 point grid that is perpendicular to the line source. For a shear wave speed of 1.4832 m/s and a compressional wave speed of 1500 m/s, shear wave simulations for a line source input in elastic and viscoelastic soft tissue models are completed in 431 and 2487 s with MATLAB scripts, respectively, where the shear viscosity is 0.1 Pa.s in the viscoelastic model. Simulations are evaluated at a single point for an acoustic radiation force generated by a 128 element linear array operating at 4.09 MHz, and these simulations require 228 s and 1327 s for elastic and viscoelastic soft tissue models, respectively. The results show that these are effective models for simulating shear wave propagation in soft tissue, and plans to accelerate these simulations will also be discussed. [Supported in part by NIH Grants R01 EB012079 and R01 DK092255.]

3:15

4pBAb2. Improved simulations of diagnostic ultrasound with the fast nearfield method and time-space. Pedro C. Nariyoshi and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw, 2120 Eng. Bldg., East Lansing, MI 48824, mcgough@egr.msu.edu)

Diagnostic ultrasound simulations are presently under development for FOCUS (http://www.egr.msu.edu/~fultras-web). To reduce the computation time without increasing the numerical error, each signal in FOCUS is calculated once, stored, and then the effects of different time delays are calculated with cubic spline interpolation. This is much more efficient than calculating the same transient signal at a scatterer repeatedly for different values of the time delay. Initially, the interpolation results were obtained from uniformly sampled signals, and now the signal start and end times are also considered. This step reduces the error in the pulse-echo calculation without significantly increasing the computation time. Simulated B-mode images were evaluated in a cyst phantom with 100 000 scatterers using this approach. Images with 50 A-lines are simulated for a linear array with 192 elements, where the translating subaperture contains 64 elements. The resulting simulated images are compared to images obtained with the same configuration in Field II (http://field-ii.dk/). An error of approximately 1% is achieved in FOCUS with a sampling frequency of 30 MHz, where Field II requires a sampling frequency of 180 MHz to reach the same error. FOCUS also reduces the simulation time by a factor of six. [Supported in part by NIH Grant R01 EB012079.]

3:30

4pBAb3. Simulations of ultrasound propagation in a spinal structure. Shan Qiao, Constantin-C Coussios, and Robin O. Cleveland (Dept. of Eng. Sci., University of Oxford, Biomedical Ultrason., Biotherapy & Biopharmaceuticals Lab. (BUBBL) Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Headington, OXFORD, Oxford OX3 7DQ, United Kingdom, shan.qiao@eng.ox.ac.uk)

Lower back pain is one of the most common health problems in developed countries, the main cause of which is the structure change of the intervertebral disks due to the degeneration. High intensity focused ultrasound (HIFU) can be used to remove the tissue of the degenerate discs through acoustic cavitation, after which injection of a replacement material can restore normal physiological function. The acoustic pressure distribution in and around the disc is important for both efficiency and safety. Ultrasound propagation from two 0.5 MHz focused transducers (placed confocally and oriented at 90 degrees) were simulated using a three-dimensional finite element model (PZFlex, Wiedlinger Associates) for both a homogeneous medium and a bovine spine. The size of computation domain was 64 mm*95 mm*95 mm, with a mesh size of 15 elements per wavelength of the fundamental waveform. Measurements of the pressure field from the two transducers in water were also performed. The simulations in a homogeneous medium agreed with the experimental results, in which a sharp ultrasound focus was observed. However, for the spine, the interference of the vertebral bodies lead to absorption in the bone and a smearing of the focus. [Work supported by EPSRC.]

3:45

4pBAb4. Can quantitative synthetic aperture vascular elastography predict the stress distribution within the fibrous cap non-invasively. Steven J. Huntzicker and Marvin M. Doyley (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, huntzick@ece.rochester.edu)

An imaging system that can detect and predict the propensity of an atherosclerotic plaque to rupture would reduce stroke. Radial and circumferential strain elastograms can reveal vulnerable regions within the fibrous cap. Circumferential stress imaging could predict the propensity of rupture. However, circumferential stress imaging demands either accurate knowledge of the geometric location of the fibrous cap or high quality strain information. We corroborated this hypothesis by performing studies on simulated vessel phantoms. More precisely, we computed stress elastograms with (1) precise knowledge of the fibrous cap, (2) no knowledge of the fibrous cap, (3) imprecise knowledge of the fibrous cap. We computed stress elastograms with accuracy of 8%, 15%, and 23% from high precision axial and lateral strain elastograms (i.e., 25 dB SNR) when precise, imprecise, and no geometric information was included the stress recovery method. The stress recovery method produced erroneous elastograms at lower noise level (i.e., 15 dB SNR), when no geometric information was included. Similarly, it produced elastograms with accuracy of 13% and 30% when precise and imprecise geometric information was included. The stress imaging method described in this paper performs well enough to warrant further studies with phantoms and *ex-vivo* samples.

4:00

4pBAb5. Super wideband quantitative ultrasound imaging for trabecular bone with novel wideband single crystal transducer and frequency sweep measurement. Liangjun Lin, Eesha Ambike (Biomedical Eng., Stony Brook Univ., Rm. 212 BioEng. Bldg., 100 Nicolls Rd., Stony Brook, NY 11794-3371, john85726@gmail.com), Raffi Sahul (TRS, Inc., State College, PA), and Yi-Xian Qin (Biomedical Eng., Stony Brook Univ., Stony Brook, NY)

Current quantitative ultrasound (QUS) imaging technology for bone provides a unique method for evaluating both bone strength and density. The broadband ultrasound attenuation (BUA) has been widely accepted as a strong indicator for bone health status. Researchers have reported BUA data between 0.3 and 0.7 MHz have strong correlation with the bone density. Recently, a novel spiral-wrapped wideband ultrasound transducer fabricated from piezoelectric PMN-PT single crystal is developed by TRS. This novel transducer combines the piezoelectric single crystal material and use of wide-band resonance transducer to provide a bandwidth superior to commercial devices with the capacity for a high sensitivity. To evaluate its application in bone imaging, a trabecular bone plate (6.5 mm thick) was prepared. The TRS transducer emits customized chirp pulses through the bone plate. The bandwidth of the ultrasound pulses is 0.2 MHz, ranging from 0.2 to 3 MHz. Based on the attenuation of the received pulses, the frequency spectrum is created to analyze the attenuation characteristics of the ultrasound attenuation across the super wide bandwidth. This new transducer technology provides more information across a wider bandwidth than the conventional ultrasound transducer and can therefore give rise to new QUS modality to evaluate bone health status.

4:15

4pBAb6. Spectrum analysis of photoacoustic signals for characterizing lymph nodes. Parag V. Chitnis, Jonathan Mamou, and Ernest J. Feleppa (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, pchitnis@riversideresearch.org)

Quantitative-ultrasound (QUS) estimates obtained from spectrum analysis of pulse-echo data are sensitive to tissue microstructure. We investigated the feasibility of obtaining quantitative photoacoustic (QPA) estimates for simultaneously providing sensitivity to microstructure and optical specificity, which could more robustly differentiate among tissue constituents. Experiments were conducted using four, gel-based phantoms ($1 \times 1 \times 2$ cm) containing black polyethylene spheres (1E5 particles/ml) that had nominal mean diameters of 23.5, 29.5, 42, or 58 μm. A pulsed, 532-nm laser excited the photoacoustic (PA) response. A 33-MHz transducer was raster scanned over the phantoms to acquire 3D PA data. PA signals were processed using rectangular-cuboidal regions-of-interests to yield three quantitative QPA estimates associated with tissue microstructure: spectral slope (SS), spectral intercept (SI), and effective-absorber size (EAS). SS and SI were computed using a linear-regression approximation to the normalized spectrum. EAS was computed by fitting the normalized spectrum to the multi-sphere analytical solution. The SS decreased and the SI increased with an increase in particle size. While EAS also was correlated with particle size, particle aggregation resulted in EAS estimates that were greater than the nominal particle size. Results indicated that QPA estimates potentially can be used for tissue classification. [Work supported by NIH grant EB015856.]

4:30

4pBAb7. Parametric assessment of acoustic output from laser-irradiated nanoparticle volumes. Michael D. Gray (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332-0405, michael.gray@me.gatech.edu), Aritra Sengupta, and Mark R. Prausnitz (School of Chemical and Biomolecular Eng., Georgia Inst. of Technol., Atlanta, GA)

A photoacoustic technique is being investigated for application to intracellular drug delivery. Previous work [Chakravarty *et al.,* Nat. Nanotechnol. **5**, 607–611 (2010)] has shown that cells immersed in nanoparticle-laden fluid underwent transient permeabilization when exposed to pulsed laser light. It was hypothesized that the stresses leading to cell membrane permeabilization were generated by impulsive pressures resulting from rapid nanoparticle thermal expansion. To assist in the study of the drug delivery technique, for which high uptake and viability rates have been demonstrated, an experimental method was developed for parametric assessment of photoacoustic output in the absence of field-perturbing elastic boundaries. This paper presents calibrated acoustic pressures from laser-irradiated streams, showing the impact of parameters including particle type, host liquid, and spatial distribution of laser energy.

4:45

4pBAb8. Modeling ultrasonic scattering from high-concentration cell pellet biophantoms using polydisperse structure functions. Aiguo Han and William D. O'Brien (Univ. of Illinois at Urbana-Champaign, 405 N. Mathews, Urbana, IL 61801, han51@uiuc.edu)

Backscattering coefficient (BSC) has been used extensively to characterize tissue. In most cases, sparse scatterer concentrations are assumed. However, many types of tissues have dense scattering media. This study models the scattering of dense media. Structure functions (defined herein as the total BSC divided by incoherent BSC) are used to take into account the correlation among scatterers for dense media. Structure function models are developed for polydisperse scatterers. The models are applied to cell pellet biophantoms that are constructed by placing live cells of known concentration in a mixture of bovine plasma and thrombin to form a clot. The BSCs of the biophantoms were measured using single-element transducers over 11–105 MHz. Experimental structure functions were derived by comparing the BSCs of two cell concentrations, a lower concentration (volume fraction: $<5\%$, incoherent scattering only) and a higher concentration (volume fraction: ~74%). The structure functions predicted by the models agreed with the experimental data. Fitting the models yielded cell radius estimates (Chinese hamster ovary cell: 6.9 microns, MAT cell: 7.1 microns, 4T1 cell: 8.3 microns) that were consistent with direct light microscope measures (Chinese hamster ovary: 6.7 microns, MAT: 7.3 microns, 4T1: 8.9 microns). [Work supported by NIH CA111289.]

5:00

4pBAb9. Characterizing collagen microstructure using high frequency ultrasound. Karla P. Mercado (Dept. of Biomedical Eng., Univ. of Rochester, 553 Richardson Rd., Rochester, NY 14623, karlapatricia.mercado@gmail.com), María Helguera (Ctr. for Imaging Sci., Rochester Inst. of Technol., Rochester, NY), Denise C. Hocking (Dept. of Pharmacology and Physiol., Univ. of Rochester, Rochester, NY), and Diane Dalecki (Dept. of Biomedical Eng., Univ. of Rochester, Rochester, NY)

Collagen is the most abundant extracellular matrix protein in mammals and is widely investigated as a scaffold material for tissue engineering. Collagen provides structural properties for scaffolds and, importantly, the microstructure of collagen can affect key cell behaviors such as cell migration and proliferation. This study investigated the feasibility of using high-frequency quantitative ultrasound to characterize collagen microstructure, namely, collagen fiber density and size, nondestructively. The integrated backscatter coefficient (IBC) was employed as a quantitative ultrasound parameter to characterize collagen microstructure in 3-D engineered hydrogels. To determine the relationship between the IBC and collagen fiber density, hydrogels were fabricated with different collagen concentrations (1–4 mg/mL). Further, collagen hydrogels polymerized at different temperatures (22–37°C) were investigated to determine the relationship between the IBC and collagen microfiber size. The IBC was computed from measurements of the backscattered radio-frequency data collected using a single-element transducer (38-MHz center frequency, 13–47 MHz bandwidth). Parallel studies using second harmonic generation microscopy verified changes in collagen microstructure. Results showed that the IBC increased with increasing collagen concentration and decreasing polymerization temperature. Further, we demonstrated that parametric images of the IBC were useful for assessing spatial variations in collagen microstructure within hydrogels.

5:15

4pBAb10. Surface roughness and air bubble effects on high-frequency ultrasonic measurements of tissue. Percy D. Segura, Caitlin Carter (Biology, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, pdsegura86@gmail.com), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

High-frequency (HF) ultrasound (10–100 MHz) has shown the ability to differentiate between healthy tissue, benign pathologies, and cancer in breast cancer surgical samples. It is hypothesized the sensitivity of HF ultrasound to breast cancer is due to changes in the microscopic structure of the tissue. The objective of this study was to determine the effects of surface roughness and air bubbles on ultrasound results. Since the testing is done with tissue inside a plastic bag, small air bubbles may form between the bag and tissue and interfere with test results. Data were collected on bovine and canine tissues to observe changes in HF readings in various organs and positions within specific tissues. Phantom samples were also created to mimic tissue with irregular surfaces and air bubbles. Samples were sealed into plastic bags, coupled to 50-MHz transducers using glycerin, and tested in pitch-catch and pulse-echo modes. The canine and bovine tissues produced similar results, with peak density trending with tissue heterogeneity. The surface grooves in bovine cardiac tissue also contributed to differences in peak densities. In phantom experiments, bubbles only affected peak density when they were isolated in the sample, but irregular surface structure had a strong effect on peak density.

Session 4pEA

Engineering Acoustics: Devices and Flow Noise

Roger T. Richards, Chair
US Navy, 169 Payer Ln., Mystic, CT 06355

Contributed Papers

1:30

4pEA1. Effect of fire and high temperatures on alarm signals. Mustafa Z. Abbasi, Preston S. Wilson, and Ofodike A. Ezekoye (Appl. Res. Lab. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78751, mustafa_abbasi@utexas.edu)

Firefighters use an acoustic alarm to recognize and locate other firefighters that need rescue. The alarm, codified under NFPA 1982 : Standard for Personal Alert Safety System (PASS), is typically implemented in firefighter's SCBA (self-contained breathing apparatus) and is carried by a majority of firefighter in the United States. In the past, the standard specified certain frequency tones and other parameters and left implementation up to manufacturers, leading to an infinite number of possibilities that could satisfy the standard. However, there is a move to converge the standard to a single alarm sound. The research presented provides science-based guidance for the next generation of PASS signal. In the two previous ASA meetings, a number of experimental and numerical studies were presented regarding the effect of temperature stratification on room acoustics. The present work uses models developed under those studies to quantify the effect of various signal parameters (frequency ranges, time delay between successive alarms, temporal envelope etc.) on the signal heard by a firefighter. Understanding the effect of these parameters will allow us to formulate a signal more resistant to distortion caused by the fire. [Work supported by U.S. Department of Homeland Security Assistance to Firefighters Grants Program.]

1:45

4pEA2. Acoustic impedance of large orifices in thin plates. Jongguen Lee, Tongxun Yi, Katsuo Maxted, Asif Syed, and Cameron Crippa (Aerosp. Eng., Univ. of Cincinnati, 539 Lowell Ave. Apt. #3, Cincinnati, OH 45220, maxtedkj@mail.uc.edu)

Acoustic impedance of large orifices (0.5–0.75 in. diameter) in thin plates (0.062 in. thickness) was investigated. This work extended the scope previously studied by Stinson and Shaw [Stinson and Shaw, Acoust. Soc. Am. **77**, 2039 (1985)] to orifice diameters that were 32 to 584 times greater than the boundary layer thickness. For a frequency range of 0.3–2.5 kHz, the resistive and reactive components were determined from an impedance tube with six fixed microphones. Sound pressure levels (SPL) were varied from 115 to 145 dB. The transition regime from constant to increasing resistances occurred at higher frequencies for larger diameters. Resistance measurements after the transition regime were in good agreement with Thurston's theory [Thurston, J. Acoust. Soc. Am. **24**, 653–656 (1952)] coupled with Morse and Ingard's resistance factor [Morse and Ingard, *Theoretical Acoustics* (McGraw-Hill, New York, 1969)]. Measured reactances remained constant at magnitudes predicted by Thurston's theory.

2:00

4pEA3. Temperature effect on ultrasonic monitoring during a filtration procedure. Lin Lin (Eng., Univ. of Southern Maine, 37 College Ave., 131 John Mitchell Ctr., Gorham, ME 04038, llin@usm.maine.edu)

Membranes are used extensively for a wide variety of commercial separation applications including those in the water purification, pharmaceutical, and food processing industries. Fouling is a major problem associated with membrane-based liquid separation processes because it can often severely limit process performance. The use of ultrasonic monitoring technique for the characterization of membranes and membrane processes has been widely used by university researchers and industrial groups for a variety of applications including membrane fouling, compaction, formation, defect detection, and morphology characterization. However, during the industrial application, such as desalination procedure, temperature of the feed liquid is not constant. This change of the temperature brings in the concern that whether the change of the ultrasonic signal is caused by the fouling or by the temperature change. This research is focus on to verify the degree of effect of temperature to ultrasonic signal, and provide a method that calibrate the temperature effect for real applications.

2:15

4pEA4. Acoustical level measuring device. Robert H. Cameron (Eng. Technol., NMSU (Retired), 714 Winter Dr., El Paso, TX 79902-2129, rcameron@elp.rr.com)

This abstract is for a poster session to describe a patent application made to the patent office in November 2012. The patent describes a system and method for determining the level of a substance in a container, based on measurement of resonance from an acoustic circuit that includes unfilled space within the container that changes size as substance is added or removed from the container. In particular, one application of this device is to measure the unfilled space in the fuel tanks of vehicles such as cars and trucks. For over 100 years, this measurement has been done by a simple float mechanism but, because of the development of tank design for vehicles that involve irregular shapes this method is increasingly less accurate. The proposed device will overcome these limitations and should provide a much more accurate reading of the unfilled space, and therefore, the amount of fuel in the tank since the total volume of the tank is known.

2:30

4pEA5. Noise induced hearing loss mitigation via planning and engineering. Raymond W. Fischer (Noise Control Eng. Inc., 799 Middlesex Turnpike, Ste. 4B, Billerica, MA 01821, rayf@noise-control.com), Kurt Yankaskas (Code 342, Office of Naval Res., Arlington, DC), and Chris Page (Noise Control Eng. Inc., Billerica, MA)

The US Navy, through an ONR lead effort, is investigating methods and techniques to mitigate hearing loss for the crews and warfighters. Hearing protection is a viable and increasingly popular method of reducing hearing exposure for many ship crew members; however, it has limitations on comfort and low frequency effectiveness, and is often used improperly. Proper naval vessel planning, programmatic changes, and advances in noise control engineering can also have significant impacts by inherently reducing noise exposure through ship design along with the use of passive noise control treatments. These impacts go beyond hearing loss mitigation since they can improve quality of life onboard vessels and provide enhanced warfighter performance. Such approaches also can be made to work in the lower frequency range where hearing protection is not as effective. This paper describes the programmatic and noise control methods being pursued to mitigate and control noise within the US Navy and US Marine Corps. Methodologies to assess the cost impact are also discussed.

2:45

4pEA6. Enhanced sound absorption of aluminum foam by the diffuse addition of elastomeric rubbers. Elizabeth Arroyo (Dept. of Mech. Eng., Univ. of Detroit Mercy, 547 N Gulley, Dearborn Heights, MI 48127, liz.arroyo12@gmail.com), Nassif Rayess, and Jonathan Weaver (Dept. of Mech. Eng., Univ. of Detroit Mercy, Detroit, MI)

The sound absorption properties of open cell aluminum foams are understood to be significant (Ashby *et al.Metal Foams: A Design Guide*, 2000) with theoretical models presented in the literature [J. Acoust. Soc. Am. **108**, 1697–1709 (2000)]. The pores that exist in metal foams, as artifacts of the manufacturing process, are left unfilled in the vast majority of cases. Work done by the US Navy (US patent 5895726 A) involved filling the voids with phthalonitrile prepolymer, resulting in a marked increase in sound absorption and vibration damping. The work presented here involves adding small amounts of elastomeric rubbers to the metal foam, thereby coating the ligaments of the foam with a thin layer of rubber. The goal is to achieve an increase in sound absorption without the addition of cost and weight. The work involves testing aluminum foam samples of various thicknesses and pore sizes in an impedance tube, with and without the added rubber. A design of experiment model was employed to gauge the effect of the various manufacturing parameters on the sound absorption and to set the stage for a physics-based predictive model.

3:00

4pEA7. Measures for noise reduction aboard ships in times of increasing comfort demands and new regulations. Robin D. Seiler and Gerd Holbach (EBMS, Technische Unversität Berlin, Salzufer 17-19, SG 6, Berlin 10967, Germany, r.seiler@tu-berlin.de)

Through the revision of the "Code of Noise Levels on Board Ships," the International Maritime Organization has tightened its recommendations from 1984 by lowering the allowed maximum noise exposure levels on board ships. Hereby, the most significant change can be observed for cabins. To consider the effects of noise on health and comfort their noise level limits were reduced by 5 dB to 55 dB(A) equivalent continuous SPL. Another important alteration is that parts of the new code will be integrated into the SOLAS-Convention, and therefore, some of its standards will become mandatory worldwide. In order to meet the increasing demands, the focus has to be put on noise reduction measures in receiving rooms and along the sound propagation paths since the opportunity to use noise reduced devices or machines is not always given. This study gives an overview of the current noise situation on board of different types of ships. The efficiency of measures for noise reduction is discussed with focus on cabins and cabin-like receiving rooms. Especially, the role of airborne sound radiation from ship windows induced by structure-borne sound is investigated.

3:15

4pEA8. Investigation of structural intensity applied to carbon composites. Mariam Jaber, Torsten Stoewer (Structural Dynam. and Anal., BMW Group, Knorrstr. 147, München 80788, Germany, mariam.jaber@bmw.de), Joachim Bös, and Tobias Melz (System Reliability and Machine Acoust. SzM, Technische Universität Darmstadt , Darmstadt, Germany)

Structures made from carbon composite materials are rapidly replacing metallic ones in the automotive industry because of their high strength to weight ratio. The goal of this study is to enhance acoustic comfort of cars made from carbon composites by comparing various carbon composites in order to find the most suitable composite in terms of mechanical and dynamic properties. In order to achieve this goal, the structural intensity method was implemented. This method can give information concerning the path of energy propagated through structures and the localization of vibration sources and sinks. The significance of the present research is that it takes into account the effect of the material damping on the dissipation of the energy in a structure. The damping of the composite is presented as a function of its micro and macro mechanical properties, frequency, geometry, and boundary conditions. The damping values were calculated by a 2D analytical multi-scale model based on the laminate theory. The benefit of this research for acoustics is that it demonstrates the effect of material properties on passive control. Consequently, structural energy propagated in carbon composite structures will be reduced and less noise will be radiated.

3:30

4pEA9. Experimental research on acoustic agglomeration of fine aerosol particles in the standing-wave tube with abrupt section. Zhao Yun, Zeng Xinwu, and Gong Changchao (Optical-Electron. Sci. and Eng., National University of Defense Technol., Changsha 410073, China, zhaoyun@nudt.edu.cn)

There is great concern about air pollution caused by fine aerosol particles, which are difficult to be removed by conventional removal system. Acoustic agglomeration is proved to be a promising method for particle control by coagulating the small particles into larger ones. Removal efficiency was grown rapidly as acoustic intensity increased. A standing-wave tube system with abrupt section was designed and built up to generate high intensity sound waves above 160 dB and avoid strong shock waves. Extensive tests were carried out to investigate the acoustic field and removal characteristics of coal-fired inhalation particles. For the development of industrial level system, a high power air-modulated speaker was applied and an insulation plate was used to separate flow induced sound. Separate experiments to determine the difference of plane standing-wave field and high order mode were conducted. The experimental study has demonstrated that agglomeration increases as sound pressure level, mass loading, and exposure time increase. The optimal frequency is around 2400 Hz for attaining integral removal effectiveness. The agglomeration rate is larger (above 86%) as much greater sound level is achieved for the pneumatic source and high order mode. The mechanism and testing system can be used effectively in industrial processes.

3:45–4:00 Break

4:00

4pEA10. Aerodynamic and acoustic analysis of an industrial fan. Jeremy Bain (Bain Aero LLC, Stockbridge, GA), Gang Wang (Ingersoll Rand, La Crosse, Wisconsin), Yi Liu (Ingersoll Rand, 800 Beaty St., Davidson, North Carolina 28036, yiliu@irco.com), and Percy Wang (Ingersoll Rand, Tyler, Texas)

The efforts to predict noise radiation for an industrial fan using direct computational fluid dynamics (CFD) simulation is presented in this paper. Industry has been using CFD tool to guide fan design in terms of efficiency prediction and improvement. However, the use of CFD tool for aerodynamic noise prediction is very limited in the past, partly due to the fact that research in aero-acoustics field was not practical for industry application. With the most recent technologies in CFD field and increasing computational power, the industry application of aero-acoustics becomes much more promising. It is demonstrated here that fan tonal noise and broadband noise at low frequencies can be directly predicted using an Overset grid system and high order finite difference schemes with acceptable fidelity.

4:15

4pEA11. On the acoustic and aerodynamic performance of serrated airfoils. Xiao Liu (Mech. Eng., Univ. of Bristol, Bristol, United Kingdom), Mahdi Azarpeyvand (Mech. Eng. Dept., Univ. of Bristol, Bristol BS8 1TR, United Kingdom, m.azarpeyvand@bristol.ac.uk), and Phillip Joseph (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom)

This paper is concerned with the aerodynamic and aeroacoustic performance of airfoils with serrated trailing edges. Although a great deal of research has been directed toward the application of serrations for reducing the trailing-edge noise, the aerodynamic performance of such airfoils has received very little research attention. Sawtooth and slitted-sawtooth trailing edges with specific geometrical characteristics have been shown to be effective in reducing the trailing edge noise over a wide range of frequencies. It has, however, also been shown that they can alter the flow characteristics near the trailing edge, namely the boundary layer thickness and surface-pressure fluctuations, and the wake formation. To better understand the effects of serrations, we shall carry out various acoustic and wind tunnel tests for a NACA6512-10 airfoil with various sawtooth, slitted and slitted-sawtooth trailing edge profiles. Flow measurements are carried out using PIV, LDV and hot-wire anemometry and the steady and unsteady forces on the airfoil are obtained using a three-component force balance system.

Results are presented for a wide range of Reynolds numbers and angles of attack. The results have shown that the use of sharp serrations can significantly change the aerodynamic performance and wake characteristics of the airfoil.

4:30

4pEA12. An experimental investigation on the near-field turbulence for an airfoil with trailing-edge serrations at different angles of attack. Kunbo Xu and Weiyang Qiao (School of Power and Energy, Northwestern PolyTech. Univ., No.127 Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

The ability to fly silently of most owl species has long been a source of inspiration for finding solutions for quieter aircraft and turbo machinery. This study concerns the mechanisms of the turbulent broadband noise reduction for an airfoil with the trailing edge serrations while the angles of attack varies from $+5°$ to Ő5°. The turbulence spatio-temporal information are measured with 3D hot-wire. The experiment is carried out in the Northwestern Polytechnical University low speed open jet wind tunnel on the SD2030 airfoil. $\lambda/h = 0.2$. It is showed the spreading rate of the wake and the decay rate of the wake centerline velocity deficit increased with serrated edge compared to the straight edge, and the three components of velocity changed differently with serrated trailing edge while the angle of attack was changed. It is also found that the turbulence peak occurs further from the airfoil surface in the presence of the serrations, and the serrations widened the mix area which allowed the flow mixed together ahead of the schedule.

4:45

4pEA13. An experimental investigation on the near-field turbulence and noise for an airfoil with trailing-edge serrations. Kunbo Xu (School of Power and Energy, Northwestern Polytechnical Univ., No.127 Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

This study concerns the mechanisms of the turbulent broadband noise reduction for an airfoil with the trailing edge serrations. The turbulence spatio-temporal information were measured with 3D hot-wire and the noise results were acquired with a line array. The experiment is carried out in the Northwestern Polytechnical University low speed open jet wind tunnel on the SD2030 airfoil. $\lambda/h = 0.2$. It is showed the spreading rate of the wake and the decay rate of the wake centerline velocity deficit increased with serrated edge compared to the straight edge, shedding vortex peaks appeared in the wake, and the three components of velocity changed differently with serrated trailing edge. Serrated trailing edge structure could reduce the radiated noise was proofed by noise results.

THURSDAY AFTERNOON, 8 MAY 2014 — BALLROOM C, 2:00 P.M. TO 4:30 P.M.

Session 4pMUa

Musical Acoustics: Automatic Musical Accompaniment Systems

Christopher Raphael, Cochair
Indiana Univ., School of Informatics and Computing, Bloomington, IN 47408

James W. Beauchamp, Cochair
Music and Electrical and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824

Invited Papers

2:00

4pMUa1. Human-computer music performance: A brief history and future prospects. Roger B. Dannenberg (School of Comput. Sci., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, rbd@cs.cmu.edu)

Computer accompaniment began in the eighties as a technology to synchronize computers to live musicians by sensing, following, and adapting to expressive musical performances. The technology has progressed from systems where performances were modeled as sequences of discrete symbols, i.e., pitches, to modern systems that use continuous probabilistic models. Although score following techniques have been a common focus, computer accompaniment research has addressed many other interesting topics, including the musical adjustment of tempo, the problem of following an ensemble of musicians, and making systems more robust to unexpected mistakes by performers. Looking toward the future, we find that score following is only one of many ways musicians use to synchronize. Score following is appropriate when scores exist and describe the performance accurately, and where timing deviations are to be followed rather than ignored. In many cases, however, especially in popular music forms, tempo is rather steady, and performers improvise many of their parts. Traditional computer accompaniment techniques do not solve these important music performance scenarios. The term Human-Computer Music Performance (HCMP) has been introduced to cover a broader spectrum of problems and technologies where humans and computers perform music together, adding interesting new problems and directions for future research.

2:25

4pMUa2. The cyber-physical system approach for automatic music accompaniment in Antescofo. Arshia Cont (STMS 9912-CNRS, UPMC, Inria MuTant Team-Project, IRCAM, 1 Pl. Igor Stravinsky, Paris 75004, France, arshia.cont@ircam.fr), José Echeveste (STMS 9912, IRCAM, CNRS, Inria MuTant Team-Project, Sorbonne Univ., UPMC Paris 06, Paris, France), and Jean-Louis Giavitto (IRCAM, UPMC, Inria MuTant team-project, CNRS STMS 9912, Paris, France)

A system capable of undertaking automatic musical accompaniment with human musicians should be minimally able to undertake real-time listening of incoming music signals from human musicians, and synchronize its own actions in real-time with that of musicians according to a music score. To this, one must also add the following requirements to assure correctness: Fault-tolerance to human or machine listening errors, and best-effort (in contrast to optimal) strategies for synchronizing heterogeneous flows of information. Our approach in Antescofo consists of a tight coupling of real-time Machine Listening and Reactive and Timed-Synchronous systems. The machine listening in Antescofo is in charge of encoding the dynamics of the outside environment (i.e., musicians) in terms of incoming events, tempo and other parameters from incoming polyphonic audio signal; whereas the synchronous timed and reactive component is in charge of assuring correctness of generated accompaniment. The novelty in Antescofo approach lies in its focus on Time as a semantic property tied to correctness rather than a performance metric. Creating automatic accompaniment out of symbolic (MIDI) or audio data follows the same procedure, with explicit attributes for synchronization and fault-tolerance strategies in the language that might vary between different styles of music. In this sense, Antescofo is a cyber-physical system featuring a tight integration of, and coordination between heterogeneous systems including human musicians in the loop of computing.

2:50

4pMUa3. Automatic music accompaniment allowing errors and arbitrary repeats and jumps. Shigeki Sagayama (Div. of Information Principles Res., National Inst. of Informatics, 2-1-2, Hitotsubashi, Choyoda-ku, Tokyo 101-8430, Japan, sagayama@nii.ac.jp), Tomohiko Nakamura (Graduate School of Information Sci. and Technol., Univ. of Tokyo, Tokyo, Japan), Eita Nakamura (Div. of Information Principles Res., National Inst. of Informatics, Japan, Tokyo, Japan), Yasuyuki Saito (Dept. of Information Eng., Kisarazu National College of Technol., Kisarazu, Japan), and Hirokazu Kameoka (Graduate School of Information Sci. and Technol., Univ. of Tokyo, Tokyo, Japan)

Automatic music accompaniment is considered to be particularly useful in exercises, rehearsals and personal enjoyment of concerto, chamber music, four-hand piano pieces, and left/right hand filled in to one-hand performances. As amateur musicians may make errors and want to correct them, or he/she may want to skip hard parts in the score, the system should allow errors as well as arbitrary repeats and jumps. Detecting such repeats/jumps, however, involves a large complexity of search for maximum likelihood transition from one onset timing to another in the entire score for every input event. We have developed several efficient algorithms to cope with this problem under practical assumptions used in an online automatic accompaniment system named "Eurydice." In Eurydice for MIDI piano, the score of music piece is modeled by Hidden Markov Model (HMM) as we proposed for rhythm modeling in 1999 and the maximum likelihood score following is done to the polyphonic MIDI input to yield the accompanying MIDI output (e.g., orchestra sound). Another version of Eurydice accepts monaural audio signal input and accompanies to it. Trills, grace notes, arpeggio, and other issues are also discussed. Our video examples include concertos with MIDI piano and piano accompanied sonatas for acoustic clarinet.

3:15

4pMUa4. The informatics philharmonic. Christopher Raphael (Comput. Sci., Indiana Univ., School of Informatics and Computing, Bloomington, IN 47408, craphael@indiana.edu)

I present ongoing work in developing a system that accompanies a live musician in a classical concerto-type setting, providing a flexible ensemble the follows the soloist in real-time and adapts to the soloist's interpretation through rehearsal. An accompanist must hear the soloist. The program models hearing through a hidden Markov model that can accurately and reliably parse highly complex audio in both offline and online fashion. The probabilistic formulation allows the program to navigate the latency/accuracy tradeoff in online following, so that onset detections occur with greater latency (and greater latency) when local ambiguities arise. For music with a sense of pulse, coordination between parts must be achieved by anticipating future evolution. The program develops a probabilistic model for musical timing, a Bayesian Belief Network, that allows the program to anticipate where future note onsets will occur, and to achieve better prediction using rehearsal data. The talk will include a live demonstration of the system on a staple from the violin concerto repertoire, as well as applications to more forward-looking interactions between soloist and computer controlled instruments.

3:40

4pMUa5. Interactive conducting systems overview and assessment. Teresa M. Nakra (Music, The College of New Jersey, P.O. Box 7718, Ewing, NJ 08628, nakra@tcnj.edu)

"Interactive Conducting" might be defined as the accompaniment of free gestures with sound—frequently, but not necessarily, the sounds of an orchestra. Such systems have been in development for many decades now, beginning with Max Mathews' "Daton" interface and "Conductor" program, evolving to more recent video games and amusement park experiences. The author will review historical developments in this area and present several of her own recent interactive conducting projects, including museum exhibits, simulation/training systems for music students, and data collection/analysis methods for the study of professional musical behavior and response. A framework for assessing and evaluating effective characteristics of these systems will be proposed, focusing on the reactions and experiences of users/subjects and audiences.

4:05

4pMUa6. The songsmith story, or how a small-town hidden Markov model dade it to the big time. Sumit Basu, Dan Morris, and Ian Simon (Microsoft Res., One Microsoft Way, Redmond, WA 98052, sumitb@microsoft.com)

It all started with a simple idea—that perhaps lead sheets could be predicted from melodies, at least within a few options for each bar. Early experiments with conventional models led to compelling results, and by designing some user interactions along with an augmented model, we were able to create a potent tool with a range of options, from an automated backing band for musical novices to a flexible musical scratchpad for songwriters. The academic papers on the method and tool led to an unexpected level of external interest, so we decided to make a product for consumers, thus was Songsmith born. What came next surprised us all—from internet parodies to stock market melodies to over 600 000 downloads and a second life in music education, Songsmith has been an amazing lesson in what happens when research and the real world collide, sometimes with unintended consequences. In this talk, I'll take you through our story, from the technical beginnings to the Internet-sized spectacle to the vast opportunities in future work, sharing with you the laughter, the heartbreak, the tears, and the joy of bringing Songsmith to the world.

THURSDAY AFTERNOON, 8 MAY 2014 BALLROOM C, 4:45 P.M. TO 6:00 P.M.

Session 4pMUb

Musical Acoustics: Automatic Accompaniment Demonstration Concert

Christopher Raphael, Cochair
Indiana Univ., School of Informatics and Computing, Bloomington, IN 47408

James W. Beauchamp, Cochair
Music and Electrical and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824

Music performed by Christopher Raphael (oboe), Roger Dannenberg (trumpet), accompanied by their automatic systems.

THURSDAY AFTERNOON, 8 MAY 2014 557, 1:30 P.M. TO 5:10 P.M.

Session 4pNS

Noise: Out on a Limb and Other Topics in Noise

Eric L. Reuter, Chair
Reuter Associates, LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801

Invited Papers

1:30

4pNS1. Necessity as the mother of innovation: Adapting noise control practice to very different set of mechanical system design approaches in an age of low energy designs. Scott D. Pfeiffer (Threshold Acoust. LLC, 53 West Jackson Blvd., Ste. 815, Chicago, IL 60604, spfeiffer@thresholdacoustics.com)

The shift in Mechanical Systems design to natural ventilation, dedicated outside air systems, variable refrigerant flow, and the return to radiant systems all present new challenges in low-noise systems. Case studies of current projects explore the sound isolation impact of natural ventilation, the benefits of reduced air quantity in dedicated outside air, the distributed noise issues in variable refrigerant flow, and the limitations of radiant systems as they apply in performing arts and noise critical spaces.

1:50

4pNS2. Readily available noise control for residences in Boston. Nancy S. Timmerman (Nancy S. Timmerman, P.E., 25 Upton St., Boston, MA 02118, nstpe@hotmail.com)

Urban residential noise control may involve high-end interior finishes, insufficient noise reduction between neighbors (in a same building), or interior/exterior noise reduction for mechanical equipment or transportation where the distances are small or non-existent. Three residences in Boston's South End, where the author is a consultant (and resident), will be discussed. The area consists of brownstones built in the mid-nineteenth century, with granite foundations, masonry facades, and common brick walls. Treatments were used which were acceptable to the "users"—neighbors on both sides of the fence.

2:10

4pNS3. Singing in the wind; noise from railings on coastal and high-rise residential construction. Kenneth Cunefare (Arpeggio Acoust. Consulting, LLC, Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

Beach-front and high-rise residential buildings are commonly exposed to sustained high winds. Balcony railings with long spans and identical pickets on uniform spacing may be driven into extremely high amplitude synchronous motion due to phase and frequency locked vortex shedding. The railing motion can excite structural vibration in floor slabs which can propagate into units and produce undesirable tone-rich noise within the units, noise that stands out well above the wind noise that also propagates into the units. Solution of this problem requires breaking the physical phenomena that induce the railing motion, including blanking off the railings; stiffening the railings; and breaking the symmetry of the individual pickets. The problem may be further complicated by questions of who should pay for the remediation of the problem, and the costs associated with remediating numerous units, particularly on high-rise developments. Increased awareness during the design phase of the potential for this problem may reduce the need for post-construction controls.

2:25

4pNS4. What do teachers think about noise in the classroom? Ana M. Jaramillo (Ahnert Feistel Media Group, 3711 Lake Dr., 55422, Robbinsdale, MN 55422, ana.jaramillo@afmg.eu), Michael G. Ermann, and Patrick Miller (School of Architecture + Design, Virginia Tech, Blacksburg, VA)

Surveys were sent to 396 Orlando-area elementary school teachers to gauge their subjective evaluation of noise in their classroom, and their general attitudes toward classroom noise. The 87 responses were correlated with the types of mechanical systems in their respective schools: (1) fan and compressor in room, (2) fan in room and remote compressor, or (3) remote fan and remote compressor. Results were also compared to the results of a previous study of the same 73 schools that linked school mechanical system type with student achievement. While teachers were more likely to be annoyed by noise in the schools with the noisiest types of mechanical systems, they were still less likely to be annoyed than the research might suggest—and when teachers did express annoyance, it was more likely to be centered around the kind of distracting noise generated by other children in adjacent corridors than by mechanical system noise.

2:40

4pNS5. Sound classification of dwellings—A comparison between national schemes in Europe and United States. Umberto Berardi (Civil and Environ. Eng. Dept., Worcester Polytechnic Inst., via Orabona 4, Bari 70125, Italy, u.berardi@poliba.it)

Schemes for the classification of dwellings related to different performances have been proposed in the last years worldwide. The general idea behind previous schemes relates to the increase in the real estate value that should follow a label corresponding to a better performance. In particular, focusing on sound insulation, national schemes for acoustic classification of dwellings have been developed in more than ten European countries. These schemes define classification classes according to different levels of sound insulation. The considered criteria are the airborne and impact sound insulation between dwellings, the facade sound insulation, and the equipment noise. Originally, due to the lack of coordination among European countries, a significant diversity among the schemes occurred; the descriptors, number of classes, and class intervals varied among schemes. However, in the last year, an "acoustic classification scheme for dwellings" has been proposed within a ISO technical committee. This paper compares existing classification schemes with the current situation in the United States. The hope is that by increasing cross-country comparisons of sound classification schemes, it may be easier to exchange experiences about constructions fulfilling different classes and by doing this, reduce trade barriers, and increase the sound insulation of dwellings.

2:55

4pNS6. Sound insulation analysis of residential building at China. Zhu Xiangdong, Wang Jianghua, Xue Xiaoyan, and Wang Xuguang (The Bldg. Acoust. Lab of Tsinghua Univ., No. 104 Main Academic Bldg. Architectural Physical Lab., Tsinghua Univ., Beijing, Beijing 100084, China, zxd@abcd.edu.cn)

Residential acoustic environment is one of the living environments that are most closely related to the daily life. The high-quality residential acoustic environment depends not only on the urban planning, building design, construction, and supervision, but also on the related regulations. In some developed countries, the residential acoustic regulations have been built up and evolved into a relatively complete system with high quality standards required. This thesis (1) conducted a questionnaire survey for resident building which be constructed at different period; (2) investigate the Technical level, the legal system, and the quality of residents to analysis the sound environment satisfaction of resident and compare it with developed countries.

3:10–3:25 Break

3:25

4pNS7. Relationship between air infiltration and acoustic leakage of building enclosures. Ralph T. Muehleisen, Eric Tatara, and Brett Bethke (Decision and Information Sci., Argonne National Lab., 9700 S. Cass Ave., Bldg. 221, Lemont, IL 60439, rmuehleisen@anl.gov)

Air infiltration, the uncontrolled leakage of air into buildings through the enclosure from pressure differences across it, accounts for a significant fraction of the heating energy in cold weather climates. Measurement and control of this infiltration is a necessary part of reducing the energy and carbon footprint of both current and newly constructed buildings. The most popular method of measuring infiltration, whole building pressurization, is limited to small buildings with fully constructed enclosures, which makes it an impractical method for measuring infiltration on medium to large buildings or small buildings still under construction. Acoustic methods, which allow for the measurement of infiltration of building sections and incomplete enclosures, have been proposed as an alternative to whole building pressurization. These new methods show great promise in extending infiltration measurement to many more buildings, but links between the acoustic leakage characteristics and the infiltration characteristics of typical enclosures are required. In this paper, the relationship between the acoustic leakage and the air infiltration through typical building envelope cracks is investigated. [This work was supported by the U.S. Department of Energy under Contract No. DE-AC02-06CH11357.]

3:40

4pNS8. Hemi-anechoic chamber qualification and comparison of room qualification standards. Madeline A. Davidson (Acoust. and Mech., Trane Lab., 700 College Dr. SPO 542, Luther College, Decorah, Iowa 52101, davima07@luther.edu)

The hemi-anechoic chamber at the Trane Laboratory in La Crosse, Wisconsin, is commonly used for acoustic testing of machinery and equipment. As required by standards, it must periodically be qualified. Sound measurements taken in a hemi-anechoic facility often depend on the assumption that the chamber is essentially free-field. To verify that the room is sufficiently anechoic, the procedures in ANSI/ASA Standard S12.55-2012/ISO 3745:2012 and ISO Standard 26101-2012 are followed. One challenge of a room qualification is finding adequate sound sources. Sources used in the qualification procedure must be Omni-directional, so directionality measurements must be taken to prove that a source is suitable for the room qualification procedure. The specific qualification procedure described in this paper involved two sound sources—a compression driver and a 6 in. $\times$ 9 in. speaker. In addition, the particular method described in this paper involves a temporary plywood floor and six microphone traverse paths extending out from the center of the chamber. This approach to qualifying a facility is

expected to define what part of the room is adequately anechoic. This paper will describe the results obtained when following each of these standards.

3:55

4pNS9. Improvement of the measurement of the sound absorption using the reverberation chamber method. Martijn Vercammen (Peutz, Lindenlaan 41, Mook 6585 ZH, Netherlands, m.vercammen@peutz.nl) and Margriet Lautenbach (Peutz, Zoetermeer, Netherlands)

The random incidence absorption coefficient is measured in a reverberation room according to ISO 354 or ASTM C423-09a. It is known that the inter laboratory accuracy under Reproducibility conditions of these results is still not very well. It is generally assumed that the limited diffusion properties of reverberation rooms, especially with a strongly sound absorbing sample, are the main reason for the bad reproducibility values for the sound absorption between laboratories. Reverberation rooms should be made much more diffuse to reduce the interlaboratory differences. However there are practical limitations in quantifying and improving the diffuse field conditions. The measured sound absorption still seems to be the most sensitive descriptor of the diffuse field conditions. A way to further reduce the interlaboratory differences is the use of a reference absorber to qualify a room and to calibrate the results of a sound absorption measurement. In the presentation an overview will be given of the research performed and some suggestions for the new version of ISO 354 will be given.

4:10

4pNS10. When acoustically rated doors fail to perform as rated, who is responsible—Manufacturer or installer? Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

Acoustical doors are designed, manufactured, and sold by several companies in the United States. They are available in multiple styles and acoustical performance ratings. The doors are specified, selected, and purchased based on the published performance ratings provided by the manufacturers, which often have had their doors tested by NAVLAP accredited acoustical testing laboratories. Of course, it should be understood by the acoustical door specifier that lab-rated doors will rarely, if ever, perform as rated after field installation. This paper presents field performance test results for numerous acoustical doors that significantly failed even the lower expected field performance criteria. The acoustical doors were all tested in-situ after they were installed in several different venues by the manufacturer's or vendor's trained and/or certified acoustical door installers. Reasons for certain field-performance failures are discussed and specific remedies are recommended.

4:25

4pNS11. Sound absorption of parallel arrangement of multiple microperforated panel absorbers at oblique incidence. Chunqi Wang, Lixi Huang, Yumin Zhang (Lab of AeroDynam, and Acoust., Zhejiang Inst. of Res. and Innovation and Dept of Mech. Eng., The Univ. of Hong Kong , Pokfulam Rd., Hong Kong, lixi@hku.hk)

Many efforts have been made to enhance the sound absorption performance of micro-perforated panel (MPP) absorbers. Among them, one straightforward approach is to arrange multiple MPP absorbers of different frequency characteristics in parallel so as to combine different frequency bands together, hence an MPP absorber array. In previous study, the parallel absorption mechanism is identified to be contributed by three factors: (i) the strong local resonance absorption, (ii) the supplementary absorption by nonresonating absorbers, and (iii) the change of environmental impedance conditions; and the local resonance absorption mechanism accounts for the increased equivalent acoustic resistance of the MPP. This study seeks to examine how the MPP absorber array performs at oblique incidence and in diffuse field. One major concern here is how the incidence angle of the sound waves affects the parallel absorption mechanism. In this study, a finite element model is developed to simulate the acoustic performance of an infinitely large MPP absorber array. Numerical results show that the sound absorption coefficients of the MPP absorber array may change noticeably as the incidence angle varies. The diffuse field sound absorption coefficients of a prototype specimen are measured in a reverberation room and compared with the numerical predictions.

4:40

4pNS12. Reverberation time in ordinary rooms of typical residences in Southern Brazil. Michael Ã. Klein, Andriele da Silva Panosso, and Stephan Paul (DECC-CT-UFSM, UFSM, Av. Roraima 1000, Camobi, Santa Maria 97105-900, Brazil, michaelklein92@hotmail.com)

In order to develop a subjective evaluation to assess the annoyance related to impact noise, it is necessary to record samples of sounds in an impact chamber that is acoustically representative for ordinary rooms, especially with respect to reverberation time. To define the target reverberation time measurements were carried out in 30 typical residences in Southern Brazil. This study presents the characteristic reverberation times of 30 furnished living rooms and 30 furnished bedrooms in buildings and houses with an average age of 34 years, 40% of them with wooden floor coverings, not as usual in modern constructions. The median T30 at 1 kHz for living rooms with an average volume of 63.60m^3 (std dev: 18.27m^3) was 0.68 s (std dev: 0.14 s), thus higher than the reference TR = 0.5 s according to EN ISO 140 parts 4, 5, and 7. The median T30 at 1 kHz for bedrooms with average volume of 33.76m^3 (std dev: 8.38m^3) was 0.49 s (std dev: 0.13 s), nearly exact the reference TR according to EN ISO 140 parts 4, 5, and 7. Data will also be compared to studies from other countries.

4:55

4pNS13. Research on the flow resistance of acoustic materials—Takes Concert Hall at Gulangyu Music School in Xiamen as an Example. Peng Wang, Xiang Yan, Lu W. Shuai, Gang Song, and Yan Liang (Acoust. Lab., School of Architecture, Tsinghua Univ., Beijng, China, 29580150@qq.com)

Different kinds of acoustic materials are used in a concert hall design, which has different functions such as diffusing, reflecting, or absorbing. The cushion of chairs in concert halls usually uses porous sound-absorbing material, whose absorbing attributes are mainly determined by its flow resistance. In the design of Concert Hall at Gulangyu Music School in Xiamen, we measured the flow resistance of materials, trying to acquire the best sound-absorbing attributes by adjusting the flow resistance, and also tested the material samples' absorbing coefficients in reverberation room. In a nutshell, measuring and analyzing flow resistance is an advanced method in acoustic design, which could help acousticians decide the most suitable absorbing attributes of chairs, and acquire the best sound quality.

Session 4pPA

Physical Acoustics: Topics in Wave Propagation and Noise

Richard Raspet, Chair
NCPA, Univ. of Mississippi, University, MS 38677

Contributed Papers

1:00

4pPA1. Mechanisms for wind noise reduction by a spherical wind screen. Richard Raspet, Jeremy Webster, and Vahid Naderyan (National Ctr. for Physical Acoust., Univ. of MS, 1 Coliseum Dr., University, MS 38606, raspet@olemiss.edu)

Spherical wind screens provide wind noise reduction at frequencies which correspond to turbulence scales much larger than the wind screen. A popular theory is that reduction corresponds to averaging the steady flow pressure distribution over the surface. Since the steady flow pressure distribution is positive on the front of the sphere and negative on the back of the sphere, the averaging results in a reduction in measured wind noise in comparison to an unscreened microphone. A specially constructed 180 mm diameter foam sphere allows the placement of an array of probe microphone tubes just under the surface of the foam sphere. The longitudinal and transverse correlation lengths as a function of frequency and the rms pressure fluctuation distribution over the sphere surface can be determined from these measurements. The measurements show that the wind noise correlation lengths are much shorter than the correlations measured in the free stream. The correlation length weighted pressure squared average over the surface is a good predictor of the wind noise measured at the center of the wind screen. [This work was supported by the Army Research Laboratory under Cooperative Agreement W911NF-13-2-0021.]

1:15

4pPA2. Infrasonic wind noise in a pine forest; convection velocity. Richard Raspet and Jeremy Webster (National Ctr. for Physical Acoust., Univ. of MS, 1 Coliseum Dr., University, MS 38606, raspet@olemiss.edu)

Simultaneous measurements of the infrasonic wind noise, the wind velocity profile in and above the canopy, and the wind turbulence spectrum in a pine forest have been completed. The wind noise spectrum can be computed from the meteorological measurements with the assumption that the lowest frequency wind noise is generated by the turbulence field above the canopy and that the higher frequencies are generated by the turbulence within the tree layer [JASA **134**(5), 4160 (2013)]. To confirm the source region identification, an array of infrasound sensors is deployed along the approximate flow direction so that the convection velocity as a function of frequency band can be determined. This paper reports on the results of this experiment. [Work supported by the U. S. Army Research Office under grant W911NF-12-0547.]

1:30

4pPA3. The effective sound speed approximation and its implications for en-route propagation. Victor Sparrow, Kieran Poulain, and Rachel Romond (Grad. Prog. Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

The effective sound speed approximation is widely used in underwater and outdoor sound propagation using common models such as ray tracing, the parabolic equation, and wavenumber integration methods such as the fast field program. It is also used in popular specialized propagation methods such as NORD2000 and the Hybrid Propagation Model (HPM). Long ago when the effective sound speed approximation was first introduced, its shortcomings were understood. But over the years, a common knowledge of those shortcomings has waned. The purpose of this talk is to remind everyone that for certain situations the effective sound speed approximation is not appropriate. One of those instances is for the propagation of sound from aircraft cruising at en-route altitudes when wind is present. This is one situation where the effective sound speed approximation can lead to substantially incorrect sound level predictions on the ground. [Work supported by the FAA. The opinions, conclusions, and recommendations in this material are those of the authors and do not necessarily reflect the views of FAA Center of Excellence sponsoring organizations.]

1:45

4pPA4. Nonlinearity spectral analysis of high-power military jet aircraft waveforms. Kent L. Gee, Tracianne B. Neilsen, Brent O. Reichman, Derek C. Thomas (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

One of the methods for analyzing noise waveforms for nonlinear propagation effects is a spectrally-based nonlinearity indicator that involves the cross spectrum between the pressure waveform and square of the pressure. This quantity, which stems directly from ensemble averaging the generalized Burgers equation, is proportional to the local rate of change of the power spectrum due to nonlinearity [Morfey and Howell, AIAA J. **19**, 986–992 (1981)], i.e., it quantifies the parametric sum and difference-frequency generation during propagation. In jet noise investigations, the quadspectral indicator has been used to complement power spectral analysis to interpret mid-field propagation effects [Gee *et al.*, AIP Conf. Proc. **1474**, 307–310 (2012)]. In this paper, various normalizations of the quadspectral indicator are applied to F-22A Raptor data at different engine powers. Particular attention is paid to the broadband spectral energy transfer around the spatial region of maximum overall sound pressure level. [Work supported by ONR.]

2:00

4pPA5. Evolution of the derivative skewness for high-amplitude sound propagation. Brent O. Reichman (Brigham Young Univ., 453 E 1980 N, #B, Provo, UT 84604, brent.reichman@byu.edu), Michael B. Muhlestein (Brigham Young Univ., Austin, Texas), Kent L. Gee, Tracianne B. Neilsen, and Derek C. Thomas (Brigham Young Univ., Provo, UT)

The skewness of the first time derivative of a pressure waveform has been used as an indicator of shocks and nonlinearity in both rocket and jet noise data [e.g., Gee *et al.*, J. Acoust. Soc. Am. **133**, EL88–EL93 (2013)]. The skewness is the third central moment of the probability density function and demonstrates asymmetry of the distribution, e.g., a positive skewness may indicate large, infrequently occurring values in the data. In the case of nonlinearly propagating noise, a positive derivative skewness signifies occasional instances of large positive slope and more instances of negative slope as shocks form [Shepherd *et al.*, J. Acoust. Soc. Am. **130**, EL8–EL13 (2011)]. In this paper, the evolution of the derivative skewness, and its interpretation, is considered analytically using key solutions of the Burgers equation. This paper complements a study by Muhlestein *et al.* [J. Acoust. Soc. Am. **134**, 3981 (2013)] that used similar methods but with a different metric.

An analysis is performed to investigate the effect of a finite sampling frequency and additive noise. Plane-wave tube experiments and numerical simulations are used to verify the analytic solutions and investigate derivative skewness in random noise waveforms. [Work supported by ONR.]

2:15

4pPA6. Application of time reversal analysis to military jet aircraft noise. Blaine M. Harker (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@byu.net), Brian E. Anderson (Geophys. Group (EES-17), Los Alamos National Lab., Los Alamos, NM), Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

The source mechanisms of jet noise are not fully understood and different analysis methods can provide insight. Time reversal (TR) is a robust data processing method that has been used in myriad contexts to localize and characterize sources from measured data, but has not extensively been applied to jet noise. It is applied here in the context of an installed full-scale military jet engine. Recently, measurements of an F-22A were taken using linear and planar microphone arrays at various engine conditions near the jet plume [Wall *et al.*, Noise Control Eng. J. **60**, 421–434 (2012)]. TR provides source imaging information as broadband and narrowband jet noise recordings are reversed and back propagated to the source region. These reconstruction estimates provide information on dominant source regions as a function of frequency and highlight directional features attributed to large-scale structures in the downstream jet direction. They also highlight the utility of TR analysis as being complementary to beamforming and other array methods. [Work supported by ONR.]

2:30–2:45 Break

2:45

4pPA7. Spectral variations near a high-performance military aircraft. Tracianne B. Neilsen, Kent L. Gee (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Spectral characteristics of jet noise depend upon location relative to the nozzle axis. Studies of the spectral variation in the far field led to a two-source model of jet noise, in which fine-scale turbulent structures are primarily responsible for noise radiation to the nozzle sideline and large-scale turbulent structures produce the broad, dominant radiation lobe farther aft. Detailed noise measurements near an F-22A Raptor shed additional insights into this variation. An initial study [Neilsen *et al.*, J. Acoust. Soc. Am. **133**, 2116–2125] was performed with ground-based microphones in the mid-field. The similarity spectra associated with the large and fine-scale turbulent structures [Tam *et al.*, AIAA paper 96–1716 (1996)] provide a reasonable representation of measured spectra at many locations. However, there are additional features that need further investigation. This paper explores the presence of a double peak in the spectra in the maximum radiation direction and a significant change in spectral shape at the farthest aft angles using data from large measurement planes (2 m $\times$ 23 m) located 4–6 jet nozzle diameters from the shear layer. The spatial variation of the spectra provides additional insight into ties between the similarity spectra and full-scale jet noise. [Work supported by ONR.]

3:00

4pPA8. Large eddy simulation of surface pressure fluctuations generated by elevated gusts. Jericho Cain (National Ctr. for Physical Acoust., Univ. of MS, 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, Maryland 20783, jericho.cain.ctr@mail.mil), Richard Raspet (National Ctr. for Physical Acoust., Univ. of MS, University, MS), and Martin Otte (Environ. Protection Agency, Atmospheric Modeling and Anal. Div., Res. Triangle Park, NC)

A surface monitoring system that can detect turbulence aloft would benefit wind turbine damage prevention, aircraft safety, and would be a new probe to study the atmospheric boundary layer. Previous research indicated that elevated velocity events may trigger pressure fluctuations on the ground. If that is true, it should be possible to monitor elevated wind gusts by measuring these pressure fluctuations. The goal of this project was to develop a ground based detection method that monitors pressure fluctuations on the ground for indicators that a gust event may be taking place at higher altitudes. Using gust data generated with a convective boundary layer large eddy simulation, cross-correlation analysis between the time evolution of the frequency content corresponding to elevated wind gusts and the pressure on the ground below were investigated. Several common features of the pressures caused by elevated gusts were identified. These features were used to develop a tracking program that monitors fast moving high amplitude pressure fluctuations and to design a ground based pressure sensing array. The array design and tracking software was used to identify several new gust events within the simulated atmosphere.

3:15

4pPA9. Response of a channel in a semi-infinite stratified medium. Ambika Bhatta, Hui Zhou, Nita Nagdewate, Charles Thompson, and Kavitha Chandra (ECE, UMass, 1 University Ave., Lowell, MA 01854, ambika_bhatta@student.uml.edu)

The presented work focuses in the exact response of two globally reacting surfaces separating a semi-infinite channel from two mediums to a point source when the speed of sound of the host medium is greater than that of the other two mediums. Analytical and numerical image based response will also be discussed in detail for different medium profiles. The modal solution of the 2-D semi-infinite channel of the stratified mediums will be obtained. The Green's function evaluated from the image based reflection coefficient will numerically be compared with the modal solution. The solution approach will be extended for three-dimensional channel. The 3-D response will be discussed in relation with the case of locally reacting surfaces of the channel.

3:30

4pPA10. Spatial coherence function for a wideband acoustic signal. Jericho Cain, Sandra Collier (US Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783, jericho.cain.ctr@mail.mil), Vladimir Ostashev, and D. Keith Wilson (U.S Army Engineer Res. and Development Ctr., Hanover, NH)

Atmospheric turbulence has a significant impact on acoustic propagation. It is necessary to account for this impact in order to study noise propagation, sound localization, and for the development of new remote sensing methods. A solution to a set of recently derived closed form equations for the spatial coherence function of a broadband acoustic pulse propagating in a turbulent atmosphere without refraction and with spatial fluctuations in the wind and temperature fields is presented. Typical regimes of the atmospheric boundary layer are explored.

3:45

4pPA11. Acoustic propagation over a complex site: A parametric study using a time-domain approach. Didier Dragna and Philippe Blanc-Benon (LMFA, Ecole Centrale de Lyon, 36 Ave. Guy de Collongue, Ecully, France, philippe.blanc-benon@ec-lyon.fr)

The influence of the ground characteristics and the meteorological conditions on the acoustic propagation of impulse signals above a complex site is studied. For that, numerical simulations using a finite-difference time-domain solver in curvilinear coordinates [Dragna *et al.*, JASA **133**(6), 3751–3763 (2013)] are performed. The reference site is a railway site in la Veuve near Reims, France, with a non-flat terrain and a mixed impedance ground, where outdoor measurements were performed in May 2010. Comparisons between the experimental data and the numerical results will be reported both in frequency domain and time domain. First, it will be shown that the numerical predictions are in a good agreement with the measured energy spectral densities and waveforms of the acoustic pressure. Second, the impacts of the variations of the ground surface impedances, of the topography and the wind direction will be analyzed.

4:00

4pPA12. The high-order parabolic equation to solve propagation problems in aeroacoustics. Patrice Malbéqui (CFD and aeroAcoust., ONERA, 29, Ave. de la Div. Leclerc, Châtillon 92350, France, patrice.malbequi@onera.fr)

The parabolic equation (PE) has proved its capability to deal with the long range sound propagation in the atmosphere. It also represents an attractive alternative to the ray model to handle duct propagation in high frequencies, for the noise radiated by the nacelle of aero-engines. It was recently shown that the High-Order Parabolic Equation (HOPE), based on a Padé expansion with an order of 5, significantly increases the aperture angle of propagation compared to the standard and the Wide-Angle PEs, allowing prediction close to cut-off frequency of the duct. This paper concerns the propagation using the HOPE in heterogeneous flows, including boundary layers above a wall and in shear layers. The thickness of the boundary layer is about dozens of centimeters while outside it, the Mach number reaches 0.5. The boundary layer effects are investigated showing the refraction effects on a range propagation of 30 m, up to 4 kHz. In the shear layer, discontinuities in the directivity patterns occur significant differences of the directivity patterns occur. Comparisons with the Euler solutions are considered, establishing the domain of application of the HOPE on a set of flow configurations, including beyond its theoretical limits. [Work supported by Airbus-France.]

4:15

4pPA13. Noise and flow measurement of serrated cascade. Kunbo Xu and Qiao Weiyang (School of Power and Energy, Northwestern Polytechnical Univ., No.127 Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

This study concerns the mechanisms of the turbulent broadband noise reduction for cascade with the trailing edge serrations. The turbulence spatio-temporal information were measured with 3D hot-wire and the noise results were acquired with a line array. The experiment is carried out in the Northwestern Polytechnical University low speed open jet wind tunnel. It is showed the spreading rate of the wake and the decay rate of the wake centerline velocity deficit increased with serrated edge compared to the straight edge, shedding vortex peaks appeared in the wake, and the three components of velocity changed differently with serrated trailing edge. Serrated trailing edge structure could reduce the radiated noise was proofed by noise results, and some peaks appeared in downstream of the cascade.

THURSDAY AFTERNOON, 8 MAY 2014 555 A/B, 1:30 P.M. TO 5:00 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Role of Medial Olivicochlear Efferents in Auditory Function

Magdalena Wojtczak, Cochair
Psychology, Univ. of Minnesota, 1237 Imperial Ln., New Brighton, MN 55112

Enrique A. Lopez-Poveda, Cochair
Inst. of Neurosci. of Castilla y Leon, Univ. of Salamanca, Calle Pintor Fernando Gallego 1, Salamanca 37007, Spain

Chair's Introduction—1:30

Invited Papers

1:35

4pPP1. Medial olivocochlear efferent effects on auditory responses. John J. Guinan (Eaton Peabody Lab, Mass. Eye & Ear Infirmary, Harvard Med. School, 243 Charles St., Boston, MA 02114, jjg@epl.meei.harvard.edu)

Medial Olivocochlear (MOC) inhibition in one ear can be elicited by sound in either ear. Curiously, the ratio of ipsilateral/contralateral inhibition depends on sound bandwidth; the ratio is ~2 for narrow-band sounds but ~1 for wide-band sounds. Reflex amplitude also depends on elicitor bandwidth and increases as bandwidth is increased, even when elicitor-sound energy is held constant. After elicitor onset (or offset), nothing changes for 20–30 ms and then MOC inhibition builds up (or decays) over 100–300 ms. MOC inhibition has typically been measured in humans by its effects on otoacoustic emissions (OAEs). Problems in such OAE studies include inadequate signal-to-noise ratios (SNRs) and inadequate separation of MOC effects from middle-ear-muscle effects. MOC inhibition reduces basilar-membrane responses more at low levels than high levels, which increases the response SNRs of higher-level signals relative to lower-level background noises, and reduces noise-induced adaptation. The net effect is expected to be increased intelligibility of sounds such as speech. Numerous studies have looked for such perceptual benefits of MOC activity with mixed results. More work is needed to determine whether the differing results are due to experimental conditions (e.g., the speech and noise levels used) or to methodological weaknesses. [Work supported by NIH-RO1DC005977.]

1:55

4pPP2. Shelter from the Glutamate storm: Loss of olivocochlear efferents increases cochlear nerve degeneration during aging.
M. Charles Liberman and Stephane F. Maison (Eaton Peabody Labs., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA, MA 02114, charles_liberman@meei.harvard.edu)

The olivocochlear (OC) feedback pathways include one population, the medial (M)OC projections to outer hair cells, which forms a sound-evoked inhibitory reflex that can reduce sound-induced cochlear vibrations, and a second population, the lateral (L)OC projections to the synaptic zone underneath the inner hair cells, that can modulate the excitability of the cochlear nerve terminals. Although there is ample evidence of OC-mediated protective effects from both of these systems when the ear is exposed to intense noise, the functional significance of this protection is questionable in a pre-industrial environment where intense noise was not so commonplace. We have re-evaluated the phenomenon of OC-mediated protection in light of recent work showing that acoustic exposure destroys cochlear neurons at sound pressure levels previously considered atraumatic, because they cause no permanent hair cell loss or threshold shift. We have shown that loss of OC innervation at a young age causes the cochlea to age a greatly accelerated rate, even without purposeful noise exposure, when aging is measured by the loss of synaptic connections between cochlear nerve fibers and hair cells. Possible relevance to hearing-in-noise problems of the elderly will be discussed.

2:15

4pPP3. Peripheral effects of the cortico-olivocochlear efferent system.
Paul H. Delano (Otolaryngol. Dept., Universidad de Chile, Independencia 1027, Santiago 8380453, Chile, phdelano@gmail.com), Gonzalo Terreros, and Luis Robles (Physiol. and Biophys., ICBM, Universidad de Chile, Santiago, Chile)

The auditory efferent system comprises descending pathways from the auditory cortex to the cochlea, allowing modulation of sensory processing even at the most peripheral level. Although the presence of descending circuits that connect the cerebral cortex with olivocochlear neurons have been reported in several species, the functional role of the cortico-olivocochlear efferent system remains largely unknown. We have been studying the influence of cortical descending pathways on cochlear responses in chinchillas. Here, we recorded cochlear microphonics and auditory-nerve compound action potentials in response to tones (1–8 kHz; 30–90 dB SPL) before, during, and after auditory-cortex lidocaine or cooling inactivation ($n = 20$). In addition, we recorded cochlear potentials in the presence and absence of contralateral noise, before, during, and after auditory-cortex micro-stimulation (2-50 μA, 32 Hz rate) ($n = 15$). Both types of auditory-cortex inactivation produced changes in the amplitude of cochlear potentials. In addition, in the microstimulation experiments, we found an increase of the suppressive effects of contralateral noise in neural responses to 2–4 kHz tones. In conclusion, we demonstrated that auditory-cortex basal activity exerts tonic influences on the olivocochlear system and that auditory-cortex electrical micro-stimulation enhances the suppressive effects of the acoustic evoked olivocochlear reflex. [Work supported by FONDECYT 1120256; FONDECYT 3130635 and Fundacion Puelma.]

2:35

4pPP4. Does the efferent system aid with selective attention?
Dennis McFadden (Psych., Univ. of Texas, 108 E. Dean Keeton A8000, Austin, TX 78712-1043, mcfadden@psy.utexas.edu), Kyle P. Walsh (Psych., Univ. of Minnesota, Minneapolis, MN), and Edward G. Pasanen (Psych., Univ. of Texas, Austin, TX)

To study whether attention and inattention lead to differential activation of the olivocochlear (OC) efferent system, a cochlear measure of efferent activity was collected while human subjects performed behaviorally under the two conditions. Listeners heard two independent, simultaneous strings of seven digits, one spoken by a male and the other by a female, and at the end of some trials (known in advance), they were required to recognize the middle five digits spoken by the female. Interleaved with the digits were one stimulus that evokes a stimulus-frequency otoacoustic emission (SFOAE) and another that activates the OC system—a 4-kHz tone (60 dB SPL, 300 ms in duration) and a wideband noise (1.0–6.0 kHz, 25 dB spectrum level, 250 ms in duration, beginning 50 ms after tone onset). These interleaved sounds, used with a double-evoked procedure, permitted the collection of a nonlinear measure called the nSFOAE. When selective attention was required behaviorally, the magnitude of the nSFOAE to tone-plus-noise differed by 1.3–4.0 dB compared to inattention. Our interpretation is that the OC efferent system was more active during attention than during relative inattention. Whether or how this efferent activity actually aided behavioral performance under attention is not known.

2:55

4pPP5. Behavioral explorations of cochlear gain reduction.
Elizabeth A. Strickland, Elin Roverud, and Kristina DeRoy Milvae (Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, estrick@purdue.edu)

Physiological measures have shown that the medial olivocochlear reflex (MOCR) decreases the gain of the cochlear active process in response to ipsilateral or contralateral sound. As a first step to determining its role in human hearing in different environments, our lab has used psychoacoustical techniques to look for evidence of the MOCR in behavioral results. Well-known forward masking techniques that are thought to measure frequency selectivity and the input/output function at the level of the cochlea have been modified so that the stimuli (masker and signal) are short enough that they should not evoke the MOCR. With this paradigm, a longer sound (a precursor) can be presented before these stimuli to evoke the MOCR. The amount of threshold shift caused by the precursor depends on its duration and its frequency relative to the signal in a way that supports the hypothesis that the precursor has reduced the gain of the cochlear active process. The magnitude and time course of gain reduction measured across our studies will be discussed. The results support the hypothesis that one role of the MOCR may be to adjust the dynamic range of hearing in noise. [Work supported by NIH(NIDCD)R01 DC008327, T32 DC000030-21, and Purdue Research Foundation.]

3:15–3:30 Break

3:30

4pPP6. Challenges in exploring the role of medial olivocochlear efferents in auditory tasks via otoacoustic emissions. Magdalena Wojtczak (Psych., Univ. of Minnesota, 1237 Imperial Ln., New Brighton, MN 55112, wojtc001@umn.edu), Jordan A. Beim, and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

A number of recent psychophysical studies have hypothesized that the activation of medial olivocochlear (MOC) efferents plays a significant role in forward masking. These hypotheses are based on general similarities between spectral and temporal characteristics exhibited by some psychophysical forward-masking results and by effects of efferent activation measured using physiological methods. In humans, noninvasive physiological measurements of otoacoustic emissions have been used to probe changes in cochlear responses due to MOC efferent activation. The aim of this study was to verify our earlier efferent-based hypothesis regarding the dependence of psychophysical forward masking of a 6-kHz probe on the phase curvature of harmonic-complex maskers. The ear-canal pressure for a continuous 6-kHz probe was measured in the presence and absence of Schroeder-phase complexes used as forward maskers in our previous psychophysical study. Changes in the ear-canal pressure were analyzed using methods for estimating the effects of efferent activation on stimulus frequency otoacoustic emissions under the assumption that changes in cochlear gain due to efferent activation will be reflected in changes in the magnitude and phase of the emission. Limitations and challenges in relating effects of feedback-based reflexes to psychophysical effects will be discussed. [Work supported by NIH grant R01DC010374.]

3:50

4pPP7. The function of the basilar membrane and medial olivocochlear (MOC) reflex mimicked in a hearing aid algorithm. Tim JÃrgens (Dept. of Medical Phys. and Acoust., Cluster of Excellence Hearing4all, UniversitÃt Oldenburg, Carl-von-Ossietzky Str. 9-11, Oldenburg 26121, Germany, tim.juergens@uni-oldenburg.de), Nicholas R. Clark, Wendy Lecluyse (Dept. of Psych., Univ. of Essex, Colchester, United Kingdom), and Meddis Ray (Dept. of Psych., Univ. of Essex, Colchester, Germany)

The hearing aid algorithm “BioAid” mimics two basic principles of normal hearing: the instantaneous compression of the basilar membrane and the efferent feedback of the medial olivocochlear (MOC) reflex. The design of this algorithm aims at restoring those parts of the auditory system, which are hypothesized to dysfunction in the individual listener. In the initial stage of this study individual computer models of three hearing-impaired listeners were constructed. These computer models reproduce the listeners’ performance in psychoacoustic measures of (1) absolute thresholds, (2) compression, and (3) frequency selectivity. Subsequently, these computer models were used as “artificial listeners.” Using BioAid as a front-end to the models, parameters of the algorithm were individually adjusted with the aim to ‘normalize’ the model performance on these psychoacoustic measures. In the final stage of the study, the optimized hearing aid fittings were evaluated with the three hearing-impaired listeners. The aided listeners showed the same qualitative characteristics of the psychoacoustic measures as the aided computer models: near-normal absolute thresholds, steeper compression estimates and sharper frequency selectivity curves. A systematic investigation of the effect of compression and the MOC feedback in the algorithm revealed that both are necessary to restore performance. [Work supported by DFG.]

4:10

4pPP8. Mimicking the unmasking effects of the medial olivo-cochlear efferent reflex with cochlear implants. Enrique A. Lopez-Poveda and Almudena Eustaquio-Martin (Inst. of Neurosci. of Castilla y Leon, Univ. of Salamanca, Calle Pintor Fernando Gallego 1, Salamanca, Salamanca 37007, Spain, ealopezpoveda@usal.es)

In healthy ears, cochlear sensitivity and tuning are not fixed; they vary depending on the state of activation of medial olivo-cochlear (MOC) efferent fibers, which act upon outer hair cells modulating the gain of the cochlear amplifier. MOC efferents may be activated in a reflexive manner by ipsilateral and contralateral sounds. Activation of the MOC reflex (MOCR) is thought to unmask sounds by reducing the adaptation of auditory nerve afferent fibers response to noise. This effect almost certainly improves speech recognition in noise. Furthermore, there is evidence that contralateral stimulation can improve the detection of pure tones embedded in noise as well as speech intelligibility in noise probably by activation of the contralateral MOCR. The unmasking effects of the MOCR are unavailable to current cochlear implant (CI) users and this might explain part of their difficulty at understanding speech in noise compared to normal hearing subjects. Here, we present preliminary results of a bilateral CI sound-coding strategy that mimics the unmasking benefits of the ipsilateral and contralateral MOCR. [Work supported by the Spanish MINECO and MED-EL GmbH.]

Contributed Papers

4:30

4pPP9. Mice with chronic medial olivocochlear dysfunction do not perform as predicted by common hypotheses about the role of efferent cochlear feedback in hearing. Amanda Lauer (Otolaryngology-HNS, Johns Hopkins Univ. School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205, alauer2@jhmi.edu)

Mice missing the alpha9 nicotinic acetylcholine receptor subunit (A9KO) show a lack of classic efferent effects on cochlear activity; however, behavioral and physiological studies in these mice have failed to support common hypotheses about the role of efferent feedback in auditory function. A9KO mice do not show deficits detecting or discriminating tones in noise. These mice also do not appear to be more susceptible to age-related hearing loss, and they do not show increased auditory brainstem response thresholds when chronically exposed to moderate-level noise. A9KO mice do show increased susceptibility to temporal processing deficits, especially when exposed to environmental noise. Furthermore, A9KO mice show extremely variable, and sometimes poor, performance when discriminating changes in the location of broadband sounds in the horizontal plane. Temporal and spatial processing deficits may be attributable to abnormal or poorly optimized representation of acoustic cues in the central auditory pathways. These results are consistent with experiments in humans that suggest artificial stimulation of medial olivocochlear efferents overestimates the actual activation of these pathways. Thus, the primary role of medial olivocochlear efferent feedback may be to regulate input from the cochlea to the brain (and within the brain) to maintain an optimal, calibrated representation of sounds.

4:45

4pPP10. Time-course of recovery from the effects of a notched-noise on the ear-canal pressure at different frequencies. Kyle P. Walsh and Magdalena Wojtczak (Psych., Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, kpwalsh@umn.edu)

Different methods for estimating the effect of the medial olivocochlear reflex (MOCR) on stimulus-frequency otoacoustic emissions (SFOAEs) in humans appear to yield different estimates of the time-course of recovery from the effect. However, it is uncertain whether the observed differences in recovery times were due to differences in the methods used to extract the changes in SFOAEs, due to the fact that different feedback-based reflexes—MOCR or the middle ear muscle reflex (MEMR)—were activated, or due to the dependence of recovery from the activated reflex on the probe frequency. In this study, the ear-canal pressure was measured for continuous probes with frequencies of 1, 2, 4, and 6 kHz, in the presence and absence of an ipsilateral notched-noise elicitor. Changes in the magnitude and phase of the ear-canal pressure were extracted to estimate recovery times from the effects of the elicitor. The results showed that the recovery time increased with increasing probe frequency—from about 380 ms at 1 kHz to about 1500 ms at 6 kHz, on average. The measurements also were repeated for each of the probe frequencies paired with a simultaneous 500-Hz tone to examine the role of the MEMR. [Work supported by NIH grant R01DC010374.]

THURSDAY AFTERNOON, 8 MAY 2014 — 553 A/B, 1:30 P.M. TO 4:55 P.M.

Session 4pSA

Structural Acoustics and Vibration and Physical Acoustics: Acoustics of Cylindrical Shells II

Sabih I. Hayek, Cochair
Eng. Sci., Penn State, 953 McCormick Ave., State College, PA 16801-6530

Robert M. Koch, Cochair
Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Invited Papers

1:30

4pSA1. A study of multi-element/multi-path concentric shell structures to reduce noise and vibration. Donald B. Bliss, David Raudales, and Linda P. Franzoni (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708, dbb@duke.edu)

Vibration transmission and noise can be reduced by dividing a structural barrier into several constituent subsystems with separate, elastically coupled, wave transmission paths. Multi-element/multi-path (MEMP) structures utilize the inherent dynamics of the system, rather than damping, to achieve substantial wide-band reduction in the low frequency range, while satisfying constraints on static strength and weight. The increased complexity of MEMP structures provides a wealth of opportunities for reduction, but the approach requires rethinking the structural design process. Prior analytical and experimental work, reviewed briefly, focused on simple beam systems. The current work extends the method to elastically coupled concentric shells, and is the first multi-dimensional study of the concept. Subsystems are modeled using a modal decomposition of the thin shell equations. Axially discrete azimuthally continuous elastic connections occur at regular intervals along the concentric shells. Simulations show the existence of robust solutions that provide large wide-band reductions. Vibratory force and sound attenuation are achieved through several processes acting in concert: different subsystem wave speeds, mixed boundary conditions at end points, interaction through elastic couplings, and stop band behavior. The results show the concept may have application in automotive and aerospace vehicles, and low vibration environments such as sensor mounts.

1:50

4pSA2. Scattering from a cylindrical shell with an internal mass. Andrew Norris and Alexey S. Titovich (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

Perhaps the simplest approach to modeling acoustic scattering from objects with internal substructure is to consider a cylindrical shell with an internal mass attached by springs. The earliest analyses, published in JASA in 1992, by Achenbach *et al.* and by Guo assumed one and two springs, respectively. Subsequent studies examined the effects of internal plates and more sophisticated models of substructure. In this talk we reconsider the Achenbach—Guo model but for an arbitrary number, say J, of axisymmetrically distributed stiffeners. The presence of a springs-mass substructure breaks the cylindrical symmetry, coupling all azimuthal modes. Our main result provides a surprisingly simple form for the scattering solution for time harmonic incidence. We show that the scattering, or T-matrix, decouples into the sum of the T-matrix for the bare shell plus J matrices each defined by an infinite vector. In addition, an approximate expression is derived for the frequencies of the quasi-flexural resonances induced by the discontinuities on the shell, which excite subsonic shell flexural waves. Some applications of the model to shells with specified long wavelength effective bulk modulus and density will be discussed. [Work supported by ONR.]

2:10

4pSA3. Active noise control for cylindrical shells using a sum of weighted spatial gradients (WSSG) control metric. Pegah Aslani, Scott D. Sommerfeldt, Yin Cao (Dept. of Phys. and Astronomy, N203 ESC Brigham Young Univ., Provo, UT 84602-4673, pegah.aslani@gmail.com), and Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

There are a number of applications involving cylindrical shells where it is desired to attenuate the acoustic power radiated from the shell, such as from an aircraft fuselage or a submarine. In this paper, a new active control approach is outlined for reducing radiated sound power from structures using a weighted sum of spatial gradients (WSSG) control metric. The structural response field associated with the WSSG has been shown to be relatively uniform over the surface of both plates and cylindrical shells, which makes the control method relatively insensitive to error sensor location. It has also been shown that minimizing WSSG is closely related to minimizing the radiated sound power. This results in global control being achieved using a local control approach. This paper will outline these properties of the WSSG control approach and present control results for a simply supported cylindrical shell showing the attenuation of radiated sound power that can be achieved.

2:30

4pSA4. Causality and scattering from cylindrical shells. James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, jgm@bu.edu)

Acoustic scattering from a cylindrical shell is required to be causal, so that the incident wave must precede the scattered wave that it creates. In the frequency domain, this statement may be explored by forming a frequency-dependent complex-valued reflection coefficient that relates the scattered wave to the incident wave. The real and imaginary parts of the reflection coefficient must therefore satisfy Hilbert Transform relations that involve integrals over frequency. As a result, one may find the real part of the reflection coefficient given only its imaginary part over a frequency range, and vice-versa. The reflection coefficient is not required to be minimum phase and rarely is minimum phase, so the causality condition cannot be used directly to estimate the phase of the reflection coefficient from its magnitude. However, the effective impedance associated with the reflection coefficient is required to be minimum phase. An approach is presented for using these relations to estimate the phase of a reflection coefficient given only its magnitude. Examples are presented that illustrate these relationships for cylindrical shells.

2:50

4pSA5. Frequency domain comparisons of different analytical and computational radiated noise solutions for point-excited cylindrical shells. Robert M. Koch (Chief Technol. Office, Naval Undersea Warfare Ctr., Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708, Robert.M.Koch@navy.mil)

Among a multitude of diverse applications, the acoustics of cylindrical shells is also an important area of study for its applicability to and representation of many US Navy undersea vehicles and systems. Examination of structural acoustic predictions of cylindrical-shell-based system designs are frequently made using a variety of analytical and computational approaches including closed-form 3D elasticity, numerous kinematic plate/shell theories, Finite Element Analysis (FEA), Energy-based FEA (EFEA) coupled with Energy Boundary Element Analysis (EBEA), and Statistical Energy Analysis (SEA). Each of these approaches has its own set of assumptions, advantages, and applicable frequency range which can make for confusion. This paper presents radiated noise solutions in the area of cylindrical shell structural acoustics from the above list of methodologies for the canonical problem of a point-excited, finite cylindrical shell with/without fluid loading. Specifically, far-field radiated sound power predictions for cylindrical shells using many different classical analytical and modern day numerical approaches (i.e., 3D elasticity, closed form plate and shell theory solutions FEA, EFEA/EBEA, SEA) are made and compared. Of particular interest for this comparison is the applicable frequency regimes for each solution and also how the solution approaches compare/transition from one to the other over a wide frequency range.

3:10–3:30 Break

3:30

4pSA6. Applications of interior fluid-loaded orthotropic shell theory for noise control and cochlear mechanics. Karl Grosh, Suyi Li, and Kon-Well Wang (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109-2125, grosh@umich.edu)

The vibration of shells with heavy interior fluid loading is a classical theory, as analyzed nearly 30 years ago by Fuller and Fahy in a series of seminal papers. Wave propagation for interiorly filled hydraulic lines, biological blood vessels, and pipelines represent classes of well-studied problems. In this paper we consider the application of this theory to two specific and seemingly disparate problems. The theory for interiorly fluid-loaded finite orthotropic shells with heavy interior fluid loading subject to end loading and with stiff end-cap terminations will be presented and compared to detailed experimental results. Application of this theory to the development of transfer matrices for developing networks of interconnected units of these systems (including the possibility of fluid flow between vessels) will be presented along with a discussion of the effects of fluid compressibility for the mechanics of outer hair cells of the mammalian cochlea.

3:50

4pSA7. Acoustic scattering from finite bilaminar cylindrical shells-directivity functions. Sabih I. Hayek (Eng. Sci., Penn State, 953 McCormick Ave., State College, PA 16801-6530, sihesm@engr.psu.edu) and Jeffrey E. Boisvert (NAVSEA Div. Newport, NUWC, Newport, RI)

The spectra of the acoustic scattered field from a normally insonified finite bilaminar cylindrical shell has been previously analyzed using the exact theory of three-dimensional elasticity (J. Acoust. Soc. Am. **134**, 4013 (2013)). The two shell laminates, having the same lateral dimensions but different radii and material properties, are perfectly bonded. The finite bilaminar shell is submerged in an infinite

acoustic medium and is terminated by two semi-infinite rigid cylindrical baffles. The shell has shear-diaphragm supports at the ends $z=0$, L and is internally filled with another acoustic medium. The bilaminar shell is insonified by an incident plane wave at an oblique incidence angle. The scattered acoustic farfield directivity function is evaluated for various incident wave frequencies and for a range of shell thicknesses, lengths, radii, and material properties. A uniform steel and a bilaminar shell made up of an outer elastomeric material bonded to an inner steel shell are analyzed to study the influence of elastomeric properties on the directivity functions. [Work supported by NAVSEA Division Newport under ONR Summer Faculty Program.]

Contributed Papers

4:10

4pSA8. Coupled vibrations in hollow cylindrical shells of arbitrary aspect ratio. Boris Aronov (BTech Acoust. LLC, Fall River, MA) and David A. Brown (Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

Vibrations of hollow cylinders have been the subject of considerable interest for many years. Piezoelectric cylinders offer a convenient system to study the vibration mode shapes, resonance frequencies and their mode coupling do to the ability to strongly and symmetrically excite extensional circumferential and axial vibration modes as well as flexural bending axial modes. While the mode repulsion of coupled circumferential and axial modes is generally widely known, their interaction gives rise to tubular flexural resonances in cylinders of finite thickness. Junger *et al*. [JASA **26**, 709–713 (1954)] appears to have been first to discredit the notion of a forbidden zone, a frequency band free of resonant modes, as being an artifact of treating thin cylinders in the membrane limit. Aronov [JASA **125**(2), 803–818 (2009)] showed experimental and theoretical proof of the presence of resonant modes throughout the spectrum as a result of the extensional mode coupling induced symmetric tubular bending modes in cylinders and their relationships as a function of different piezoelectric polarizations. That analysis used the energy method and the Euler-Lagrange equations based on the coupling of assumed modes of vibration and the synthesis of results using equivalent electromechanical circuits. This paper aims to both summarize and generalize those results for the applicability of passive cylindrical shells.

4:25

4pSA9. Attenuation of noise from impact pile driving in water using an acoustic shield. Per G. Reinhall, Peter H. Dahl, and John T. Dardis (Mech. Eng., Univ. of Washington, Stevens Way, Box 352600, Seattle, WA 98195, tdardis@u.washington.edu)

Offshore impact pile driving produces extremely high sound levels in water. Peak acoustic pressures from the pile driving operation of ~10^3 Pa at a range of 3000 m, ~10^4 Pa at a range of 60 m, and ~10^5 Pa at a range of 10 m have been measured. Pressures of these magnitudes can have negative effects on both fish and marine mammals. Previously, it was shown that the primary source of sound originates from radial expansion of the pile as a compression wave propagates down the pile after each strike. As the compression wave travels it produces an acoustic field in the shape of an axisymmetric cone, or Mach cone. The field associated with this cone clearly dominates the peak pressures. In this paper, we present an evaluation of the effectiveness of attenuating pile driving noise using an acoustic shield. In order to fully evaluate the acoustic shield, we provide results from finite element modeling and simple plane wave analysis of impact pile driving events with and without a noise shield. This effort is supported by the findings from a full-scale pile driving experiment designed to evaluate the effectiveness of the noise shield. Finally, we will discuss methods for improving the effectiveness of the acoustic shield.

4:40

4pSA10. Free and forced vibrations of hollow elastic cylinders of finite length. D. D. Ebenezer, K. Ravichandran (Naval Physical and Oceanogr. Lab, Thrikkakara, Kochi, Kerala 682021, India, d.d.ebenezer@gmail.com), and Chandramouli Padmanabhan (Indian Inst. of Technol., Madras, Chennai, Tamil Nadu, India)

An analytical model of axisymmetric vibrations of hollow elastic circular cylinders with arbitrary boundary conditions is presented. Free vibrations of cylinders with free or fixed boundaries and forced vibrations of cylinders with specified non-uniform displacement or stress on the boundaries are considered. Three series solutions are used and each term in each series is an exact solution to the exact governing equations of motion. The terms in the expressions for components of displacement and stress are products of Bessel and sinusoidal functions and are orthogonal to each other. Complete sets of functions in the radial and axial directions are formed by terms in the first series and the other two, respectively. It is therefore possible to satisfy arbitrary boundary conditions. It is shown that two terms in each series are sufficient to determine several resonance frequencies of cylinders with certain specified boundary conditions. The error is less than 1% for free cylinders. Numerical results are also presented for forced vibration of hollow steel cylinders of length 10 mm and outer diameter 10 mm with specified normal displacement or stress. Excellent agreement with finite element results is obtained at all frequencies up to 1 MHz. Convergence of the series is also discussed.

Session 4pSC

Speech Communication: Special Populations and Clinical Considerations

Sarah H. Ferguson, Chair
Commun. Sci. and Disorders, Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112

Contributed Papers

1:30

4pSC1. Internal three dimensional tongue motion during "s" and "sh" from tagged magnetic resonance imaging; control and glossectomy motion. Joseph K. Ziemba, Maureen Stone, Andrew D. Pedersen, Jonghye Woo (Neural and Pain Sci., Univ. of Maryland Dental School, 650 W. Baltimore St., Rm. 8207, Orthodontics, Baltimore, MD 21201, mstone@umaryland.edu), Fangxu Xing, and Jerry L. Prince (Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

This study aims to ascertain the effects of tongue cancer surgery (glossectomy) on tongue motion during the speech sounds "s" and "sh." Subjects were one control and three glossectomies. The first patient had surgery closed with sutures. The second had sutures plus radiation, which produces fibrosis and stiffness. The third was closed with an external free flap, and is of particular interest since he has no direct motor control of the flap. Cine and tagged-MRI data were recorded in axial, coronal and sagittal orientations at 26 fps. 3D tissue point motion was tracked at every time-frame in the word. 3D displacement fields were calculated at each time-frame to show tissue motion during speech. A previous pilot study showed differences in "s" production [Pedersen *et al.*, JASA (2013)]. Specifically, subjects differed in internal tongue motion pattern, and the flap patient had unusual genioglossus lengthening patterns. The "s" requires a midline tongue groove, which is challenging for the patients. This study continues that effort by adding the motion of "sh," because "sh" does not require a midline groove and may be easier for the patients to pronounce. We also add more muscles, to determine how they interact to produce successful motion. [This study was supported by NIH R01CA133015.]

1:45

4pSC2. An acoustic threshold for third formant in American English /r/. Sarah M. Hamilton, Suzanne E. Boyce, Leah Scholl, and Kelsey Douglas (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Mail Location 379Cincinnati, OH 45267, Suzanne.Boyce@uc.edu)

It is well known that a low F3 is the most salient acoustic feature of American English /r/, and that the degree of F3 lowering is correlated with the degree to which /r/ is perceptually acceptable to native listeners as a "good" vs. "misarticulated" /r/. Identifying the point at which F3 lowering produces a "good" /r/ would be helpful in remediation of /r/-production difficulties in children and second language learners. Such a measure would require normalization across speakers. Hagiwara (1995) observed that F3 for /r/ in competent adult speakers was at or below 80% of the average vowel frequencies for a given speaker. In this study, we investigate whether children's productions start to sound "good" when they lower F3 to the 80% demarcation level or below. Words with /r/ and vowel targets from 20 children with a history of /r/ misarticulation were extracted from acoustic records of speech therapy sessions. Three experienced clinicians judged correctness of /r/ productions. Measured F3's at the midpoint of /r/ and a range of vowels were compared for these productions. Preliminary findings suggest that the 80% level is a viable demarcation point for good vs. misarticulated articulation of /r/.

2:00

4pSC3. Prosodic variability in the speech of children who stutter. Timothy Arbisi-Kelm, Julia Hollister, Patricia Zebrowski, and Julia Gupta (Commun. Sci. and Disord., Univ. of Iowa, Wendell Johnson Speech and Hearing Ctr., Iowa City, IA 52242, timothy-arbisi-kelm@uiowa.edu)

Developmental stuttering is a heterogeneous language disorder characterized by persistent speech disruptions, which are generally realized as repetitions, blocks, or prolongations of sounds and syllables (DSM-IV-R, 1994). While previous studies have uncovered ample evidence of deficits in both "higher-level" linguistic planning and "lower-level" motor plan assembly, identifying the relative contribution of the specific factors underlying these deficits has proved difficult. Phrasal prosody represents a point of intersection between linguistic and motoric planning, and therefore a promising direction for stuttering research. In the present study, 12 children who stutter (CWS) and 12 age-matched controls (CWNS) produced sentences varying in length and syntactic complexity. Quantitative measures (F0, duration, and intensity) were calculated for each word, juncture, and utterance. Overall, CWS produced a narrower F0 range across utterance types than did CWNS, while utterance duration did not differ significantly between groups. Within utterances, CWS (but not CWNS) produced a greater degree of preboundary lengthening preceding relative clauses in syntactically complex sentences, as well as higher F0 variability at these juncture points. Such differences suggest that for CWS utterance planning is sensitive to syntactic complexity, possibly reflecting either a deficit in syntactic processing or the relative effects of syntactic processing on a strained processing system.

2:15

4pSC4. Tongue shape complexity for liquids in Parkinsonian speech. Doug H. Whalen (Haskins Labs., 300 George St. Ste. 900, New Haven, CT 06511, whalen@haskins.yale.edu), Katherine M. Dawson, Micalle Carl (Speech-Language-Hearing Sci., City Univ. of New York, New York, NY), and Khalil Iskarous (Dept. of Linguist, Univ. of Southern California, Los Angeles, CA)

Parkinson's disease (PD) is a neurological disorder characterized by the degeneration of dopaminergic neurons. Speech impairments in PD are characterized by slowed muscle activation, muscle rigidity, variable rate, and imprecise consonant articulation. Complex muscular synergies are necessary to coordinate tongue motion for linguistic purposes. Our previous work showed that people with PD had an altered rate of change in tongue shape during vowel to consonant transitions, but differences were small, perhaps due to the simplicity of the speech task. In order to test sentences, four PD participants and three older controls were imaged using ultrasound. They repeated sentences from the Rainbow Passage. Tongue shape complexity in liquids and adjacent vowels was assessed by their bending energy [Young *et al.*, Info. Control **25**(4), 357–370 (1974)]. Preliminary results show that bending energy was higher in liquids than in vowels, and higher in controls than PD speakers. Production of liquids typical requires a flexible tongue shape; these PD speakers show reduced flexibility that is nonetheless compensated sufficiently for the production of intelligible speech. Implications for speech motor control and for PD evaluation will be discussed.

2:30

4pSC5. VocaliD: Personal voices for augmented communicators. H Timothy Bunnell (Ctr. for Pediatric Auditory and Speech Sci., Alfred I. duPont Hospital for Children, 1701 Rockland Rd., Wilmington, DE 19807, bunnell@asel.udel.edu) and Rupal Patel (Dept. of Speech Lang. Pathol. and Audiol., Northeastern Univ., Boston, MA)

The goal of the VocaliD project (for vocal identity) is to develop personalized synthetic voices for children and adults who rely on speech generating devices (SGDs) for verbal communication. Our approach extracts acoustic properties related to source, vocal tract, or both from a target talker's disordered speech (whatever sounds they can still produce) and applies these features to a synthetic voice that was created from a surrogate voice donor who (ideally) is similar in age, size, gender, etc. The result is a synthetic voice that contains as much of the vocal identity of the target talker as possible yet the speech clarity of the surrogate talker's synthetic voice. To date, we have deployed several synthetic voices using this technology. Three case studies will be presented to illustrate the methods used in voice generation and the results from three pediatric SGD users. We will also describe plans to greatly extend our database of surrogate voice donor speech, allowing us to better match regional/dialectical features to the needs of the target SGD users.

2:45

4pSC6. Perceptual learning in the laboratory versus real-world conversational interaction. Elizabeth D. Casserly (Dept. of Psych., Trinity College, 300 Summit St., Hartford, CT 06106, elizabeth.casserly@trincoll.edu) and David B. Pisoni (Dept. of Psychol. & Brain Sci., Indiana Univ., Bloomington, IN)

Understanding perceptual learning effects under novel acoustic circumstances, e.g., situations of hearing loss or cochlear implantation, constitutes a critical goal for research in the hearing sciences and for basic perceptual research surrounding spoken language use. These effects have primarily been studied in traditional laboratory settings using stationary subjects, prerecorded materials, and a restricted set of potential subject responses. In the present series of experiments, we extended this paradigm to investigate perceptual learning in a situated, interactive, real-world context for spoken language use. Experiments 1 and 2 compared the learning achieved by normal-hearing subjects experiencing real-time cochlear implant acoustic simulation in either conversation or traditional feedback-based computer training. In experiment 1, we found that interactive conversational subjects achieved perceptual learning equal to that of laboratory-trained subjects for speech recognition in the quiet, but neither group generalized this learning to other domains. Experiment 2 replicated the learning findings for speech recognition in quiet and further demonstrated that subjects given active perceptual exposure were able to transfer their perceptual learning to a novel task, gaining significantly more benefit from the availability of semantic context in an isolated word recognition task than subjects who completed conventional laboratory-based training.

3:00

4pSC7. Spectrotemporal alterations and syllable stereotypy in the vocalizations of mouse genetic models of speech-language disorders. Gregg A. Castellucci (Linguist, Yale Univ., 333 Cedar St., Rm. I-407, New Haven, CT 06511, gregg.castellucci@yale.edu), Matthew J. McGinley, and David A. McCormick (Neurobiology, Yale School of Medicine, New Haven, CT)

Specific language impairment (SLI) and developmental dyslexia (DD) are common speech-language disorders exhibiting a range of phonological and speech motor deficits. Recently, mouse genetic models of SLI (Foxp2) and DD (Dcdc2) have been developed and promise to be powerful tools in understanding the biological basis of these diseases. Surprisingly, no studies of the adult vocalizations—which exhibit the most elaborate and complex call structure—have been performed in these mouse strains. Here, we analyze the male ultrasonic courtship song of Dcdc2 knockout mice and Foxp2 heterozygous knockout mice and compare it to the song of their C57BL/6J background littermates. Preliminary analysis indicates considerable difference between the three groups. For example, Foxp2 heterozygous knockout song contains less frequency modulation and has a reduced syllable inventory in comparison to that of wildtype littermates. The call production and phonological deficits exhibited by these mouse models are reminiscent of the symptoms observed in humans with these disorders.

3:15

4pSC8. Listening effort in bilateral cochlear implants and bimodal hearing. Matthew Fitzgerald, Katelyn Glassman (Otolaryngol., New York Univ. School of Medicine, 550 1st Ave., NBV-5E5, New York, NY 10016, fitz.mb@gmail.com), Sapna Mehta (City Univ. of New York, New York, NY), Keena Seward, and Arlene Neuman (Otolaryngol., New York Univ. School of Medicine, New York, NY)

Many users of bilateral cochlear implants, or of bimodal hearing, report, reduced listening effort when both devices are active relative to a single device. To quantify listening effort in these individuals, we used a dual-task paradigm. In such paradigms, the participant divides attention between a primary and secondary task. As the primary task becomes more difficult, fewer cognitive resources are available for the secondary task, resulting in poorer performance. The primary task was to repeat AzBio sentences in quiet, and in noise. The secondary task was to recall a digit string presented visually before a set of two sentences. As a control, both the primary and secondary tasks were tested alone in a single-task paradigm. Participants were tested unilaterally and bilaterally / bimodally. Relative to the single-task control, scores obtained in the dual-task paradigm were not affected in the primary sentence-recognition task, but were lower on the secondary digit-recall task. This suggests that a dual-task paradigm has potential to quantify listening effort. Some listeners who showed bilateral benefits to speech understanding had higher bilateral than unilateral digit-recall scores. However, there was considerable variability on the digit-recall task, which hinders our ability to draw clear conclusions.

3:30–3:45 Break

3:45

4pSC9. Measurement of spectral resolution and listening effort in people with cochlear implants. Matthew Winn (Dept. of Surgery, Univ. of Wisconsin-Madison, 1500 Highland Ave., Rm. 565, Madison, WI 53705, mwinn83@gmail.com), Ruth Y. Litovsky, and Jan R. Edwards (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI)

Cochlear implants (CIs) provide notably poor spectral resolution, which poses significant challenges for speech understanding, and places greater demands on listening effort. We evaluated a CI stimulation strategy designed to improve spectral resolution by measuring its impact on listening effort (as quantified by pupil dilation, which is considered to be a reliable index of cognitive load). Specifically, we investigated dichotic interleaved processing channels (where odd channels are active in one ear, and even channels are active in the contralateral ear). We used a sentence listening and repetition task where listeners alternated between their everyday clinical CI configurations and the interleaved channel strategy, to test which offered better resolution and demanded less effort. Methods and analyses stemmed from previous experiments confirming that spectral resolution has a systematic impact on listening effort in individuals with normal hearing. Pupil dilation measures were generally consistent with speech perception ($r^2 = 0.48$, $p < 0.001$), suggesting that spectral resolution plays an important role in listening effort for listeners with CIs. When using interleaved channels, both speech perception performance and pupillary responses were variable across individuals, underscoring the need for individualized measurement for CI listeners rather than group analysis, in the pursuit of better clinical fitting.

4:00

4pSC10. Automatic speech recognition of naturalistic recordings in families with children who are hard of hearing. Mark VanDam (Speech & Hearing Sci., Washington State Univ., PO BOX 1495, Spokane, WA 99202, mark.vandam@wsu.edu) and Noah H. Silbert (Commun. Sci. & Disord., Univ. Cincinnati, Cincinnati, OH)

Performance of an automatic speech recognition (ASR) system [LENA Research Foundation, Boulder, CO] has been reported for naturalistic, whole day recordings collected in families with typically developing (TD) children. This report examines ASR performance of the LENA system in families with children who are hard-of-hearing (HH). Machine-labeled segments were compared with human judges' assessment of talker identity (*child*, *mother*, or *father*), and recordings from families with TD children were compared with families with HH children. Classification models were fit to several acoustic variables to assess decision process differences between machine and human labels and between TD and HH groups. Accuracy and error of both machine and human performance is reported. Results may be useful to improve implementation and interpretation of ASR techniques in terms of special populations such as children with hearing loss. Findings also have implications for very large database applications of unsupervised ASR, especially its application to naturalistic acoustic data.

4:15

4pSC11. Assessing functional auditory performance in hearing-impaired listeners with an updated version of the Modified Rhyme Test. Douglas Brungart, Matthew J. Makashay, Van Summers, Benjamin M. Sheffield, and Thomas A. Heil (Audiol. and Speech Pathol. Ctr., Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil)

Pure-tone audiometric thresholds are the gold standard for assessing hearing loss, but most clinicians agree that the audiogram must be paired with a speech-in-noise test to make accurate predictions about how listeners will perform in difficult auditory environments. This study evaluated the effectiveness of a six-alternative closed-set speech-in-noise test based on the Modified Rhyme Test (House, 1965). This 104-word test was carefully constructed to present stimuli with and without ITD-based spatial cues at two different levels and two different SNR values. This allows the results to be analyzed not only in terms of overall performance, but also in terms of the impact of audibility, the slope of the psychometric function, and the amount of spatial release from masking for each individual listener. Preliminary results from normal and hearing-impaired listeners show that the increase in overall level from 70 dB to 78 dB that was implemented in half of the trials had little impact on performance. This suggests that the test is relatively effective at isolating speech-in-noise distortion from the effects of reduced audibility at high frequencies. Data collection is currently underway to compare performance in the MRT test to performance in a matrix sentence task in a variety of realistic operational listening environments. [The views expressed in this abstract are those of the authors and do not necessarily reflect the official policy or position of the DoD or the US Government.]

4:30

4pSC12. The contribution of speech motor function to the cognitive testing. Emily Wang, Stanley Sheft, Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 1611 West Harrison St., Ste. 530, Chicago, IL 60612, emily_wang@rush.edu), and Raj Shah (The Rush Alzheimer's Disease Core Ctr., Rush Univ. Medical Ctr., Chicago, IL)

This pilot study was to explore speech function as a possible confounding factor in the assessment of persons with Mild Cognitive Impairment (MCI) due to Alzheimer's disease (AD). In the United States, over 30 million people are 65 and older with 10 to 20% of them suffering from MCI due to AD. Episodic memory is tested in diagnosis of MCI due to AD using recall of a story or a list of words. Such tasks involve both speech and hearing. Normal aging also impacts one's speech and hearing. In this study, we designed a test battery to investigate the contribution of speech and hearing on testing of episodic memory. Sixty community-dwelling Black and 60 demographically matched White, all over 74 years, non-demented persons participated in the study. They each produced a story-retell and named animals in one minute. All subjects were tested with hearing and speech measures (maximum-sustained vowel phonation and diadochokinetic rates). Preliminary results showed that small but consistent differences were seen between the two racial groups in the diadochokinetic rates ($p < 0.05$). There were negative correlations between the Story-retell and diadochokinetic rates, which may suggest that speech motor control may indeed be a confounding factor in episodic memory testing.

4:45

4pSC13. The effect of background noise on intelligibility of adults and children with dysphonia. Keiko Ishikawa (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, 5371 Farmridge Way, Mason, OH 45040, ishikak@mail.uc.edu), Maria Powell (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Amelia, OH), Heidi Phero (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Alessandro de Alarcon (Pediatric Otolaryngol. Head & Neck Surgery, Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), Sid M. Khosla (Dept. of Otolaryngol., Univ. of Cincinnati, College of Medicine, Cincinnati, OH), Suzanne Boyce (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and Lisa Kelchner (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., OH)

A majority of patients with dysphonia report reduced intelligibility in their daily communication environments. Laryngeal pathology often causes abnormal vibration and incomplete closure of the vocal folds, resulting in increased noise and decreased harmonic power in the speech signal. These acoustic consequences likely make dysphonic speech more difficult to understand, particularly in the presence of background noise. The study tested two hypotheses: (1) intelligibility of dysphonic speech is more negatively affected by background noise than that of normal speech, and (2) listener ratings of intelligibility will correlate with clinical measures of dysphonia. One hundred twenty speech samples were collected from 6 adults and 4 children with normal voice and 6 adults and 4 children with varying degrees of dysphonia. Each sample consisted of a short phrase or sentence and was characterized by two acoustic measures commonly associated with degree of dysphonia: cepstral peak prominence (CPP) and harmonic to noise ratio (HNR). Samples were combined with three levels of "cafeteria" noise (+0 dB SNR, +5 dB SNR, and no noise) and then subjected to a speech perception experiment with 60 normal listeners. This project is ongoing. Preliminary results support hypothesis 1; additional findings related to hypothesis 2 will also be discussed.

Session 4pSP

Signal Processing in Acoustics and Underwater Acoustics: Sensor Array Signal Processing II

Mingsian R. Bai, Chair
Power Mech. Eng., Tsing Hua Univ., 101 Sec.2, Kuang_Fu Rd., Hsinchu 30013, Taiwan

Contributed Papers

1:30

4pSP1. Processing methods for coprime arrays in complex shallow water environments. Andrew T. Pyzdek (Graduate Program in Acoust., The Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu) and R. Lee Culver (Appl. Res. Lab., The Penn State Univ., State College, PA)

Utilizing the concept of the coarray, coprime arrays can be used to generate fully populated cross-correlation matrices with a greatly reduced number of sensors by imaging sensors to fill in gaps in the physical array. Developed under free space far-field assumptions, such image sensors may not give accurate results in complicated propagation environments, such as shallow water. Taking shallow water acoustic models under consideration, it will be shown that image sensors can still be used, but to a more limited extent based on spatial variability. Performance of a coprime array with limited image sensors and full image sensors will be compared with that of a fully populated array. [This research was supported by the Applied Research Laboratory, at the Pennsylvania State University through the Eric Walker Graduate Assistantship Program.]

1:45

4pSP2. Compressive beamforming in noisy environments. Geoffrey F. Edelmann, Charles F. Gaumond, and Jeffrey S. Rogers (Acoust. (Code 7160), U.S. Naval Res. Lab., 4555 Overlook Ave. SW (Code 7162), Code 7145, Washington, DC 20375, edelmann@nrl.navy.mil)

The application of compressive sensing to detect targets of interest could greatly impact future beamforming systems. Inevitably, at-sea data are contaminated with measured noise. When the ocean is stationary enough to form multiple snap-shots, a covariance matrix may be formed to mitigate noise. Results of compressive beamforming on a covariance matrix will be shown on at-sea measurements. Results will be compared with a robust adaptive beamformer and compressive beamformer. It will be shown that the dictionary of a compressive covariance beamformer goes as the number of measurements squared leading to a compromise between processor and array gain. [This work was supported by ONR.]

2:00

4pSP3. Passive ranging in underwater acoustic environment subject to spatial coherence loss. Hongya Ge (ECE, New Jersey Inst. of Technol., New Jersey Inst. of Technol., University Heights, Newark, NJ 07102, ge@njit.edu) and Ivars P. Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

In this work, a two-stage multi-rank solution for passive ranging is presented for acoustic sensing systems using multi-module towed hydrophone arrays operating in underwater environments subject to spatial coherence loss. The first stage of processing consists of adaptive beam-forming on the individual modular array level to improve the signal-to-noise ratio and at the same time to adaptively reduce the data dimensionality. The second stage of multi-rank filtering exploits the possible spatial coherence existing across the spatially distributed modular arrays to further improve the accuracy of passive ranging. The proposed solution reduces to either the well-known non-coherent solution under no spatial coherence, or the fully coherent solution under perfect spatial coherence. For large distributed arrays, the asymptotic approximation of the proposed solution has a simple beam-space interpretation. We conclude with a discussion of the estimator when the spatial coherence is unknown and its implications for the passive ranging system performance.

2:15

4pSP4. Eigenvector-based test for local stationarity applied to beamforming. Jorge E. Quijano (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd. (Ring Road), Victoria, BC V8P 5C2, Canada, jorgeq@uvic.ca) and Lisa M. Zurk (Elec. and Comput. Eng. Dept., Portland State Univ. , Portland, OR)

Sonar experiments with large-aperture horizontal arrays often include a combination of targets moving at various speeds, resulting in non-stationary statistics of the data snapshots recorded at the array. Accurate estimation of the sample covariance (prior to beamforming and other array processing procedures) is achieved by including a large number of snapshots. In practice, this accuracy is affected by the requirement to limit the observation interval to snapshots with local stationarity. Data-driven statistical tests for stationarity are then relevant as they allow determining the maximum number of snapshots (i.e., the best case scenario) for sample covariance estimation. This work presents an eigenvector-based test for local stationarity. It can be applied to the improvement of beamforming when targets must be detected in the presence of loud-slow interferers in the water column. Given a set of (possibly) non-stationary snapshots, the proposed approach forms subsets of a few snapshots, which are used to estimate a sequence of sample covariances. Based on the structure of sample eigenvectors, the proposed test gives a probability measure of whether such consecutive sample covariances have been drawn from the same underlying statistics. The approach is demonstrated with simulated data using parameters from the Shallow Water Array Processing (SWAP) project.

2:30

4pSP5. Wind turbine blade health monitoring using acoustic beamforming techniques. Kai Aizawa (Dept. of Precision Mech., Chuo Univ., Lowell, Massachusetts) and Christopher Niezrecki (Dept. of Mech. Eng., Univ. of Massachusetts Lowell, One University Ave., Lowell, MA 01854, Christopher_Niezrecki@uml.edu)

Wind turbines operate autonomously and can possess reliability issues attributed to manufacturing defects, fatigue failure, or extreme weather events. In particular, wind turbine blades can suffer from leading and trailing edge splits, holes, or cracks that can lead to blade failure and loss of energy revenue generation. In order to help identify damage, several approaches have been used to detect cracks in wind turbine blades; however, most of these methods require transducers to be mounted on the turbine blades, are not effective, or require visual inspection. This paper will propose a new methodology of the wind turbine non-contact health monitoring using the acoustic beamforming techniques. By mounting an audio speaker inside of the wind turbine blade, it may be possible to detect cracks or damage within the structure by observing the sound radiated from the blade. Within this work, a phased array beamforming technique is used to process acoustic data for the purpose of damage detection. Several algorithms are

evaluated including the CLEAN-based Subtraction of Point spread function from a Reference (CLSPR) on a composite panel and a section of a wind turbine blade in the laboratory.

2:45

4pSP6. Compressive beamforming with co-prime arrays. Jeffrey S. Rogers, Geoffrey F. Edelmann, and Charles F. Gaumnd (Acoust. Div., Naval Res. Lab, 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375, jeff.rogers@nrl.navy.mil)

The results of compressive beamforming using arrays formed by Nyquist, co-prime samplers, Wichmann rulers, and Golomb rulers are shown along with forms of array gain, resolution and latency as measures of performance. Results will be shown for the idea case of few sources with Gaussian amplitudes in spatially white Gaussian white noise. Results will also be shown for data taken on the Five Octave Research Array (FORA). [This work was supported by ONR.]

3:00

4pSP7. How round is the human head? Buye Xu, Ivo Merks, and Tao Zhang (Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344, buye_xu@starkey.com)

Binaural microphone arrays are becoming more popular for hearing aids due to their potential to improve speech understanding in noise for hearing impaired listeners. However, such algorithms are often developed using three-dimensional head-related transfer function measurements which are expensive and often limited to a manikin head such as KEMAR. As a result, it is highly desired to use a parametric model for binaural microphone array design on a human head. Human heads have been often modeled using a rigid sphere when diffraction of sound needs to be considered. Although the spherical model may be a reasonable model for first order binaural microphone arrays, recent study has shown that it may not be accurate enough for designing high order binaural microphone arrays for hearing aids on a KEMAR (Merks *et al*., 2014). In this study, main sources of these errors are further investigated based on numerical simulations as well as three-dimensional measurement data on KEMAR. The implications for further improvement will be discussed.

3:15–3:30 Break

3:30

4pSP8. Data fusion applied to beamforming measurement. William D. Fonseca (Civil Eng., Federal Univ. of Santa Maria, Rua Lauro Linhares, 657, Apto 203B, FlorianÃpolis, Santa Catarina 88036-001, Brazil, will.fonseca@eac.ufsm.br) and JoÃo P. Ristow (Mech. Eng., Federal Univ. of Santa Catarina, FlorianÃpolis, Santa Catarina, Brazil)

The aim of this work is use data fusion in a set of data obtained from measurements done with a microphone array in different times to improve beamforming results. Beamforming is a technique that basically samples the sound field with an array of sensors. The correct summation of these signals will render a reinforcement of the recorded sound for a chosen direction in space. In addition, processing a set of possible incoming directions enables the creation of sound maps. The spatial resolution in beamforming is directly related to array's constructive factors and frequency of analysis. One way to improve resolution is increasing array's size and number of sensors. Considering the measured source statistically stationary, it is possible to use signals obtained in different times to evaluate it. In this way, the array can be placed in different positions, and the data acquired can be processed and fused in order to create a single set of data corresponding to a virtual array composed by all aforementioned positions.

3:45

4pSP9. Passive multi-target localization by cross-correlating beams of a compact volumetric array. John Gebbie, Martin Siderius (Northwest Electromagnetics and Acoust. Res. Lab., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, jgebbie@ece.pdx.edu), Peter L. Nielsen, and James Miller (Res. Dept., STO-CMRE, La Spezia, Italy)

A technique is presented for passively localizing multiple noise-producing targets by cross-correlating the elevation beams of a compact volumetric array on separate bearings. A target's multipath structure inherently contains information about its range, however unknown, random noise waveforms make time separation of individual arrivals difficult. Ocean ambient noise has previously been used to measure multipath delays to the seabed by cross-correlating the beams of a vertical line array [Siderius *et al*., J. Acoust. Soc. Am. **127**, 2193–2200 (2010)], but this methodology has not been applied to distant noise sources having non-vertical arrivals. In this paper, methods are presented for using a compact volumetric array mounted to an autonomous underwater vehicle to measure the directionality and time delays of multipath arrivals, while simultaneously rejecting clutter and interference. This is validated with results from the GLASS'12 experiment in which a small workboat maneuvered in shallow water. Short ranges could be estimated reliably using straight ray paths, but longer ranges required accounting for ray refraction effects. Further, this is related to striation patterns observed in spectrograms, and it is shown that measured multipath time delays are used to predict this pattern, as well as the waveguide invariant parameter, β.

4:00

4pSP10. Near- and far-field beam forming using a linear array in deep and shallow water. Richard L. Culver, Brian E. Fowler, and D. Chris Barber (Appl. Res. Lab., Penn State Univ., Po Box 30, 16804, State College, PA 16801, r.lee.culver@gmail.com)

Underwater sources are typically characterized in terms of a source level based on measurements made in the free-field. Measurements made in a harbor environment, where multiple reflections, high background noise and short propagation paths are typical, violates these conditions. The subject of this paper is estimation of source location and source level from such measurements. Data from a test conducted at the US Navy Acoustic Research Detachment in Bayview, Idaho during the summers of 2010 and 2011 are analyzed. A line array of omnidirectional hydrophones was deployed from a barge in both deep and shallow water using calibrated acoustic sources to evaluate the effectiveness of post-processing techniques, as well as line array beamforming, in minimizing reflected path contributions and improving signal-to-noise ratio. A method of estimating the location of the sources while taking into account a real, non-linear array based on these measurements is presented. [Work supported by the Applied Research Laboratory under an Eric Walker Scholarship.]

4:15

4pSP11. Two-dimensional slant filters for beam steering. Dean J. Schmidlin (El Roi Analytical Services, 2629 US 70 East, Unit E-2, Valdese, NC 28690-9005, djschmidlin@charter.net)

The concept of a two-dimensional digital "slant" filter is introduced. If the input and output of the slant filter are represented by matrices whose row and column indices denote discrete time and discrete space, respectively, then each diagonal of the output matrix is equal to the linear convolution of the corresponding diagonal of the input matrix with a common one-dimensional sequence. This sequence may be considered as the impulse response of a one-dimensional shift-invariant filter. The transfer function of the slant filter has the form $H(z_1,z_2) = G(z_1z_2)$ where $G(z)$ is the transfer

4p THU. PM

function of the one-dimensional filter. It is shown that the slant filter is capable of forming and steering a beam using pressure samples from a linear array. The output of the beamformer is equal to the last column of the output matrix of the slant filter. One interesting feature is the possibility that two beamformers can have the same beamwidth but steer the beam to different angles. Another is that though the slant filter is two-dimensional, it can be designed by utilizing well-developed one-dimensional techniques. An example is presented to illustrate the theoretical concepts.

4:30

4pSP12. Compressive acoustic imaging with metamaterials. Yangbo Xie, Tsung-Han Tsai, David J. Brady, and Steven A. Cummer (Elec. and Comput. Eng., Duke Univ., 3417 CIEMAS, Durham, NC 27705, yx35@duke.edu)

Compressive imaging has brought revolutionary design methodologies to imaging systems. By shuffling and multiplexing the object information space, the imaging system compresses data on the physical layer and enables employing fewer sensors and acquiring less data than traditional isomorphic mapping imaging systems. Recently metamaterials have been investigated for designing compressive imager. Metamaterials are engineered materials with properties that are usually unattainable in nature. Acoustic metamaterials can possess highly anisotropy, strongly dispersion, negative dynamic density, or bulk modulus, and they open up new possibilities of wave-matter interaction and signal modulation. In this work, we designed, fabricated, and tested a metamaterial-based single detector, 360 degree field of view compressive acoustic imager. Local resonator arrays are design to resonate randomly in both spatial and spectrum dimensions to favor compressive imaging task. The presented experimental results show that with only about 60 measured values, the imager is able to reconstruct a scene of more than 1000 sampling points in space, achieving a compression ratio of about 20:1. Multiple static and moving target imaging task were performed with this low cost, single detector, non-mechanical scanning compressive imager. Our work paves the way for designing metamaterials based compressive acoustic imaging system.

4:45

4pSP13. Frequency-difference matched field processing in the presence of random scatterers. Brian Worthmann (Appl. Phys., Univ. of Michigan, 2010 W.E.Lay Automotive Lab., 1231 Beal Ave., Ann Arbor, MI 48109, bworthma@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Matched field processing (MFP) is an established technique for locating remote acoustic sources in known environments. Unfortunately, unknown random scattering and environment-to-propagation model mismatch prevents successful application of MFP in many circumstances, especially those involving high frequency signals. Recently a novel nonlinear array-signal-processing technique, frequency difference beamforming, was found to be successful in combating the detrimental effects of random scattering for 10 kHz to 20 kHz underwater signals that propagated 2.2 km in a shallow ocean sound channel and were recorded by a 16-element vertical array. This presentation covers the extension of the frequency-difference concept to MFP using sound propagation simulations in a nominally range-independent shallow ocean sound channel that includes point scatterers. Here again, 10 kHz to 20 kHz signals are broadcast to a vertical 16-element array, but the frequency difference approach allows Bartlett and adaptive MFP ambiguity surfaces to be calculated at frequencies that are an order of magnitude (or more) below the signal bandwidth where the detrimental effects of environmental mismatch and random scattering are much reduced. Comparison of these results with equivalent simulations of conventional Bartlett and adaptive MFP for different of source-array ranges are provided. [Sponsored by the Office of Naval Research.]

5:00

4pSP14. Evaluation of a high-order Ambisonics decoder for irregular loudspeaker arrays through reproduced field measurements. Jorge A. Trevino Lopez (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 9808577, Japan, jorge@ais.riec.tohoku.ac.jp), Takuma Okamoto (National Inst. of Information and Communications Technol., Kyoto, Japan), Yukio Iwaya (Faculty of Eng., Tohoku Gakuin Univ., Tagajo, Miyagi, Japan), Shuichi Sakamoto, and Yo-iti Suzuki (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., Sendai, Japan)

High-order Ambisonics (HOA) is a sound field reproduction technique that defines a scalable and system-independent encoding of spatial sound information. Decoding of HOA signals for reproduction using loudspeaker arrays can be a difficult task if the angular spacing between adjacent loudspeakers, as observed from the listening position, is not uniform. In this research, one of such systems is considered: a 157-channel irregular loudspeaker array. The array is used to reproduce simple HOA-encoded sound fields. Three HOA decoding methods are evaluated: two conventional ones and a recently proposed decoder designed for irregular loudspeaker arrays. Reproduction accuracy is compared by directly measuring the sound pressure around the listening position, the so-called sweet spot. Coarse-resolution sound field measurements give an approximate size for the listening region generated by the different methods. In addition, dummy head recordings are used to evaluate interaural level and phase differences. The results are used to estimate the accuracy of the system when presenting spatial sound. This study shows the importance of selecting a proper decoding method to reproduce HOA with irregular loudspeaker arrays. This is emphasized by the use of an actual loudspeakers system instead of a computer simulation, a common shortcoming of previous studies.

Session 4pUW

Underwater Acoustics: Acoustic Vector Sensor Measurements: Basic Properties of the Intensity Vector Field and Applications II

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Invited Papers

1:30

4pUW1. Development of a uniaxial pressure-acceleration probe for diagnostic measurements performed on a spherical sound projector. James A. McConnell (Appl. Physical Sci. Corp. , 4301 North Fairfax Dr., Ste. 640, Arlington, VA 22203, jmcconnell@aphysci.com)

Historically speaking, underwater acoustic vector sensors have seen widespread use in direction finding applications. However, given that this class of sensor typically measures both the acoustic pressure and at least one component of the particle velocity at a single point in space, they can be used effectively to measure the acoustic intensity and/or the acoustic impedance. These metrics can be useful in understanding the acoustic field associated with simple and complex sound radiators. The focus of this paper concerns the development of a uniaxial pressure-acceleration (p-a) probe to measure the specific acoustic impedance of a spherical sound projector (i.e., International Transducers Corporation ITC1001 transducer) over the frequency range from 2.5 to 10 kHz. The design, fabrication, and calibration of the probe are covered along with the results of the aforementioned experiment. Results show that reasonable agreement was obtained between the measured data and an analytical prediction, which models the sound projector as a point source positioned in a free-field.

1:50

4pUW2. An adaptive beamformer algorithm using a quadratic norm of the Poynting vector for vector sensor arrays. Arthur B. Baggeroer (Mech. and Elec. Eng., Massachusetts Inst. of Technol., Rm. 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu)

An adaptive beamformer for vector sensor arrays (VSA's), which uses a quadratic norm of the acoustic Poynting vector (PV) and linear constraint on the PV itself, is introduced. The paradigm follows minimum variance distortionless response (MVDR) but now the metric to be minimized is a quartic function of the filter weights and the constraint is quadratic. This leads to numerical approaches for the optimization instead of a matrix inversion for MVDR. This exploration is motivated by the observation that many nonlinear processing methods lead to "better" performance when a signal is above some threshold SNR. Examples of these include split beam arrays, DIFAR's and monopulse systems. This presentation discusses the optimization method and compares the results for ABF with linear processing for VSA's. The use of linear and quadratic refer to the clairvoyant processing where the ABF uses ensemble covariances and leaves open the problem of sample covariance estimation. [Work supported by ONR Code 321, Undersea Signal Processing.]

Contributed Papers

2:10

4pUW3. The modal noise covariance matrix for an array of vector sensors. Richard B. Evans (Terrafore. Inc., 99F Hugo Rd., North Stonington, CT 06359, rbevans@99main.com)

A modal noise covariance matrix for an array of vector sensors is presented. It is assumed that the sensors measure pressure and gradients or velocities on three axes. The noise covariance matrix is obtained as a discrete modal sum. The derivation relies on the differentiation of the complex pressure field and the application of a set of Bessel function integrals. The modal representation is restricted to a horizontally stratified environment and assumes that the noise sources form a layer of uncorrelated monopoles. The resulting noise field is horizontally isotropic, but vertically non-isotropic. Particular attention is paid to the effect of the noise source intensity on the normalization of the covariance matrix and, consequently, to the effect of noise on the output of the array of vector sensors.

2:25

4pUW4. Bearing estimation from vector sensor intensity processing for autonomous underwater gliders. Kevin B. Smith, Timothy Kubisak, James M. Upshaw (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Rm. 114, Monterey, CA 93943, kbsmith@nps.edu), James S. Martin, David Trivett (Woodruff School of Mech. Eng., Georgia Inst. of Technol. , Atlanta, GA), and C. Michael Traweek (Office of Naval Res., Arlington, VA)

Data have been collected on acoustic vector sensors mounted on autonomous underwater gliders in the Monterey Bay during 2012–2013. In this work, we show results of intensity processing to estimate bearing to impulsive sources of interest. These sources included small explosive shots deployed by local fisherman, and humpback whale vocalizations. While the highly impulsive shot data produced unambiguous bearing estimations, the longer duration whale vocalizations showed a fairly wide spread in bearing. The causes of the ambiguity in bearing estimation are investigated in the context of the highly variable bathymetry of the Monterey Bay Canyon, as well as the coherent multipath interference in the longer duration calls.

2:40

4pUW5. Detection and tracking of quiet signals in noisy environments with vector sensors. Donald DelBalzo (Marine Information Resources Corp., 18139 Bellezza Dr., Orlando, Florida 32820, delbalzo@earthlink.net), James Leclere, Dennis Lindwall, Edward Yoerger, Dimitrios Charalampidis, and George Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

We analyze the utility of vector sensors to detect and track underwater acoustic signals in noisy environments. High ambient noise levels below 300 Hz are often dominated by a few loud discrete ships that produce a complicated and dynamic noise covariance structure. Horizontal arrays of omni-directional hydrophones improve detection by forming (planewave) beams that "listen" between loud azimuthal directions with little regard to changing noise fields. The inherent 3-D directionality of vector sensors offers the opportunity to exploit detailed noise covariance structure at the element level. We present simulation performance results for vector sensors in simple and realistic environments using particle filters that can adapt to changing acoustic field structures. We demonstrate the ability of vector sensors to characterize and mitigate the deleterious effects of noise sources. We also demonstrate the relative value of vector vs. omni-directional sensing (and processing) for single sensors and compact arrays.

2:55

4pUW6. Coherent vector sensor processing for autonomous underwater glider networks. Brendan Nichols, James Martin, Karim Sabra, David Trivett (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr. NW, Atlanta, GA 30309, bnichols8@gatech.edu), and Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA)

A distributed array of autonomous underwater gliders, each fitted with a vector sensor measuring acoustic pressure and velocity, form an autonomous sensor network theoretically capable of detecting and tracking objects in an ocean environment. However, uncertainties in sensor positions impede the ability of this glider network to perform optimally. Our work aims to compare the performance of coherent and incoherent processing for acoustic source localization using an array of underwater gliders. Data used in the study were obtained from numerical simulations as well as experimental data collected using the research vessel as a source for localization purposes. By estimating the vessel position with a single glider's data (incoherent) and comparing to the location estimated with both gliders' data (coherent), it was determined that location estimation accuracy could be improved using coherent processing, provided the gliders' positions could be measured with sufficient precision. The results of this study could potentially aid the design and navigation strategies of future glider networks with a large number of elements.

3:10–3:30 Break

3:30

4pUW7. Development of vector sensors for flexible towed array. Vladimir Korenbaum and Alexandr Tagiltcev (Pacific Oceanologic Inst. FEB RAS, 43, Baltiiskaya Str., Vladivostok 690041, Russian Federation, v-kor@poi.dvo.ru)

Main problems of application of vector sensors (VSs) for flexible towed arrays are providing high performance under small dimensions as well as necessary flow noise immunity. The objective is to develop VSs met these demands. A simulation of performance of VS embedded in a flexible towed array body formed with sound transparent compound is performed. The developed one-dimensional model, predicts existence of a suspension resonance, dividing frequency band of VS into two parts. The lower part of the band is more applicable for VS of inertial type while the upper one is more preferred for VS of gradient type. A possibility to control the suspension resonance frequency in limits of 500–2000 Hz is shown for experimental model. The flow noise immunity problem is analyzed for different frequency bands and types of VSs. Various methods of flow noise cancellation are developed for different frequency bands and types of VSs, which include power flux processing, compensation of vibration response, convolution processing. Examples of design of one- and two-component VSs are represented. [The study was supported by the grant 13-NTP-II-08 of Far Eastern Branch of Russian Academy of Sciences.]

3:45

4pUW8. Acoustic particle velocity amplification and flow noise reduction with acoustic velocity horns. Dimitri Donskoy (Ocean Eng., Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, ddonskoy@stevens.edu) and Scott E. Hassan (Naval Undersea Warfare Ctr., Newport, RI)

Small wavelength size acoustic velocity horns (AVH) were recently introduced [J. Acoust. Soc. Am. **131**(5), 3883–3890 (2012)] as particle velocity amplifiers having flat amplitude and phase frequency responses below their first resonance. AVH predicted amplification characteristics have been experimentally verified demonstrating interesting opportunities for vector sensors (VS) sensitivity enhancement. Present work provides enhanced analysis of amplification and characteristics of complex shape horns. Additionally, we address another AVH feature: turbulence flow noise reduction due to turbulence field spatial averaging across horn's mouth area. Numerical analysis demonstrated up to 25 dB convective turbulent pressure and velocity reduction at the horn throat.

4:00

4pUW9. Development of a standing-wave calibration apparatus for acoustic vector sensors. Richard D. Lenhart, Jason D. Sagers (Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, lenhart@arlut.utexas.edu), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Lab., The Univ. of Texas at Austin, Austin, TX)

An apparatus was developed to calibrate acoustic hydrophones and vector sensors between 25 and 2000 Hz. A standing-wave field is established inside a vertically oriented, water-filled, elastic-walled waveguide by a piston velocity source at the bottom and a pressure-release boundary condition at the air/water interface. A computer-controlled linear positioning system allows reference hydrophones and/or the device under test to be scanned through the water column while their acoustic response is measured. Some of the challenges of calibrating vector sensors in such an apparatus are discussed, including designing the waveguide to mitigate dispersion, mechanically isolating the apparatus from floor vibrations, understanding the impact of waveguide structural resonances on the acoustic field, and developing processing algorithms to calibrate vector sensors in a standing-wave field. Data from waveguide characterization experiments and calibration measurements will be presented. [Work supported by ARL IR&D.]

4:15

4pUW10. Very low frequency acoustic vector sensor calibration. Dimitri Donskoy (Ocean Eng., Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, ddonskoy@stevens.edu)

In-water calibration of Acoustic Vector Sensors (AVS) operating at very low frequencies (fraction of Hz to hundreds of Hz) presents a set of unique challenges as the acoustic wavelengths are much longer than any existing laboratory calibration facilities. The developed calibration approach utilizes existing Naval Undersea Warfare Center's pressurized horizontal calibrating steel tube equipped with two independently controlled sound sources located at the opposite ends of the tube. Controlling the phase and amplitude of these sources allows for creating of pressure or velocity fields inside the tube. Respective pressure and particle velocity complex amplitudes are measured and calculated, respectively, with two reference hydrophones. Experimental results of this calibration approach is presented for a newly developed very low frequency AVS comprising of pressure and non-inertial velocity sensors built into an acoustic velocity horn.

4:30

4pUW11. Spatial correlation of the acoustic vector field of the surface noise in three-dimensional ocean environments. Yiwang Huang and Junyuan Guo (College of Underwater Acoust. Eng., Harbin Eng. Univ., Nantong St. No.145, Nangang District, Heilongjiang,Harbin 150001, China, guojunyuan89@163.com)

Spatial correlation of ocean ambient noise is a classical and attractive topic in ocean acoustics. Usually acoustic particle velocity can be formulated by the gradient of sound pressure. But due to the complexity of the sound pressure in range-dependent environments, the velocities of the surface noise are too difficult to be solved by this way. Fortunately, by taking advantage of the exchangeability of partial derivative and integral operation, a new derivation was proposed and a vector model for the surface-generated noise in three-dimensional ocean environments was developed directly from the correlation function of sound pressure. As a model verification, spatial correlation of the acoustic vector field of the surface noise in a range-independent environment was derived, and the identical correlation functions were given compared with the literature. After then, the surface noise in a range-dependent environment was considered with a rigid bottom hypothesis. The effects on the correlation taken by the bottom sloping and medium absorption were analyzed numerically.

THURSDAY EVENING, 8 MAY 2014 — 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Thursday are as follows:

7:30 p.m.	Animal Bioacoustics	554AB
7:30 p.m.	Biomedical Acoustics	Ballroom E
7:30 p.m.	Musical Acoustics	Ballroom C
7:30 p.m.	Noise	557
7:30 p.m.	Speech Communication	Ballroom D
7:30 p.m.	Underwater Acoustics	556AB