Session 4aAA

Architectural Acoustics: Green Building Acoustics Design and Challenges

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Chair's Introduction—8:10

Invited Papers

8:15

4aAA1. Noise prediction of vehicle sources on freeways and arterials using measured sound data. John J. LoVerde, David W. Dong, and Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Evaluation and mitigation of noise from vehicular sources is common as a building design criterion and has been part of the California State Building Code and HUD multi-family building design requirements since the 1970s. It is now included in Green Building Design Standards and school and healthcare facility design guidelines and is expanding to all types of buildings with the addition of the new acoustics credit in LEED version 4. These criteria require that the noise level be quantified precisely, but do not provide a method for defining the noise level given the normal variations in noise level. This paper examines the factors that should be considered when defining the exterior noise from vehicular sources. Methods for predicting the noise level using data from relatively short measurement periods are evaluated, and minimum survey requirements to determine specific exterior noise parameters are suggested.

8:40

4aAA2. Acoustics design associated with natural ventilation. Weihwa Chiang, Huiping Wu, and Haohsiang Hsu (Architecture, National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd., Taipei 106, Taiwan, edchiang1224@hotmail.com)

Increased effort on building sustainability has caused a revolution in building acoustic design regarding savings in resources and energy. Sustainable design strategies such as natural ventilation, on the other hand, challenged the common practice in acoustic design. Case studies on building acoustic solutions associated with natural ventilation have been conducted and reviewed with multi-facet design concerns. Issues addressed included noise insulation of building facade that allows natural ventilation, abatement of noise from urban sub-stations that generated heat, replacing mechanical ventilation system by stack ventilation for parking garages to decrease both power consumption and mechanical noise, and noise suppression of ceiling fan to prevent from overuse of PA system in classrooms that may consequently cause further problems. Discussions were also made about degradation of absorbent materials caused by increased humidity due to natural ventilation.

9:05

4aAA3. Meeting "green" acoustical requirements in flexible classrooms. Rose Mary Su and Benjamin Markham (Acentech Inc., 33 Moulton St., Cambridge, MA 02139, rsu@acentech.com)

Acoustical design in classrooms has evolved significantly since the ANSI S12.60-2002 standard was first established. Since then, institutions such as Leadership in Energy & Environmental Design (LEED) and the Collaborative for High Performance Schools (CHPS) have adopted aspects of the ANSI standard for school projects. Simultaneously, architects are creating increasingly flexible classroom designs. The push for a more flexible learning space sometimes clashes with acoustical design requirements stipulated by LEED and CHPS. This paper will discuss some of the acoustic design challenges of creating flexible, 21st century learning spaces while at the same time meeting the acoustic requirements driven by LEED and CHPS compliance. Discussion will include movable partitions in a learning space that work, sound absorptive finishes implemented beyond the standard suspended acoustical ceiling, and non-conventional mechanical systems in classroom settings. Some case studies will illustrate the discussion.

9:30

4aAA4. Top opportunities and challenges in meeting acoustics criteria in green buildings—Specific case studies. Joseph F. Bridger (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 101, Raleigh, NC 27607, joe@sacnc.com)

Acoustics criteria found in for example LEED, DODEA, and ANSI S12.60 (classroom acoustics in schools) often must be met either in addition to or as part of the sustainable design project requirements. The challenge is that building designs are changing to meet the goals of sustainable design, often with unanticipated effects on building acoustics. The criteria themselves are evolving as experience is gained. Acoustics materials are also adapting to meet sustainable design goals. We will discuss the top opportunities and challenges we are finding in meeting acoustics criteria in green buildings.

9:55–10:10 Break

10:10

4aAA5. Leadership in energy and environmental design acoustics compliance: Simulation versus field measurements. Lucky S. Tsaih, Lee-Su Huang, and Lisa Huang (Dept. of Architecture, Natl. Taiwan Univ. of Sci. and Technol., Taipei, Taiwan, akustx@mail.ntust.edu.tw)

In classrooms, acoustics and lighting are equally critical qualities that shape the learning environment. LEED IEQ Prerequisite 3: Minimum Acoustic Performance only addresses a prescriptive requirement for compliance. LEED IEQ Credit 8.1: Daylight and Views allows several options for demonstrating achievement of minimum illumination levels: simulation, prescriptive, measurement, and combination. In examining PK Yonge Elementary School in Gainesville, Florida, field measurements of daylighting and acoustics were performed. Measurements demonstrate that 3 pm illumination levels are better than at 9 am, but model simulations show both to be the same. Model simulation results are at least 10 times better than field measurements. This discrepancy suggests that measurements are critical for more accurate results. Acoustical model simulations for reverberation time were also conducted and the results showed inconsistencies from field measurements. Both daylighting and acoustical simulations provide preliminary results but a range of unpredictable factors affect the final precision of the simulation results. These factors include as-built material finishes, furniture layouts, reflectance, and absorption coefficients. Therefore, LEED should require that upon a building's completion, sample acoustic field measurements are necessary to verify compliance.

10:35

4aAA6. Do elementary school children eat less in noisy cafeterias? Michael Ermann (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu), Elena L. Serrano (Dept. of Human Nutrition, Foods, & Exercise, Virginia Tech, Blacksburg, VA), Carman Byker (Health and Human Development, Montana State Univ., Bozeman, MT), and Robert Calvey (Design for America, Chicago, IL)

Twenty years ago, the increased occupant productivity linked with high performance, LEED, green, sustainable, Passivhaus, and day-lit buildings was generally backed by anecdote. A flurry of recent research, however, has consistently confirmed the once-anecdotal narrative: when buildings perform better, workers do more, students learn more, and sales spike. After a substantial financial investment in a high performance building, owners can expect meaningful energy savings with modest payback times, but for buildings where occupant performance has a value, both the construction costs and energy savings are a rounding error relative to occupant productivity benefits. Geothermal and passive thermal systems are explored as opportunities to align low-energy thermal systems to acoustics; passive ventilation and thermal mass are explored as low-energy thermal comfort strategies that challenge acoustic concerns. The author's recent research in daylighting, thermal mass, night insulation, and the effects of cafeteria noise on the eating habits of elementary school children will be highlighted.

Contributed Papers

11:00

4aAA7. Consideration of acoustics in leadership in energy and environmental design (LEED) version 4. John J. LoVerde, Samantha Rawlings, and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Acoustical criteria have been added to the LEED rating systems with each new revision, beginning with Schools and Healthcare. With the recent Version 4 update, an acoustical credit has been added for the majority of New Construction LEED rating systems. There are weaknesses in the acoustical LEED credits, including imbalance between cost and benefit, mitigation beyond industry standards, and implementation of requirements within non-sensitive areas. This paper reviews the purpose of acoustics within a sustainable design system, identifies areas where the current language hits the mark or should be revised, and adds areas for future consideration of acoustical design within a sustainable design rating system.

11:15

4aAA8. Concrete core activation and suspended ceilings: Designing for comfort, energy efficiency, and good acoustics. Martijn Vercammen and Hanneke Peperkamp (Peutz, Lindenlaan 41, Mook 6585 ZH, Netherlands, m.vercammen@peutz.nl)

The trend to design energy-efficient buildings continues. Both legislation as sustainability assessment methods have increased the popularity of thermally activated concrete slabs. It is a way to use low temperature heating and high temperature cooling which makes it very suited for the use in low energy systems. The efficiency of these systems relate to the surface area, often the ceiling area. Exactly that surface was already the domain for the sound absorbing ceiling. So in new buildings with high energy performance due to concrete core activation, the sound absorption is often banned, resulting in very poor acoustics. The use of open, sound absorbing ceilings will have an influence on the thermal capacity of the concrete slabs, However, little is known about this effect. To investigate the effect of open ceilings to both the cooling capacity as the sound absorption, theoretical/empirical models have been made to estimate the effect on the cooling capacity and the sound absorption. The method is also tested in a field situation. It turns out that optimization is possible, with both cooling capacity as sound absorption around 70% of the maximum.

11:30

4aAA9. Acoustics testing and simulation analysis of waiting hall in the line-side high-speed railway station. Gang Liu, Dan Hou, Lixiong Wang, and Rui Dang (School of Architecture, Tianjin Univ., Wei Jin Rd. No. 92, Nankai District, Tianjin 300072, China, youknowleft@sina.com)

Integrated the flow density changes of the waiting hall in a line-side high-speed railway station, an on-site measurement of noise environment is carried out. The characteristics of the acoustic environment are discussed in this study. Furthermore, combining the measured data of the background noise, acoustic computer simulation program ODEON calculates the reverberation time and speech transmission index of public broadcasting system. The results indicate that the reverberation time exceeds 5 s and the speech intelligibility of the south waiting room and dining area in second floor is lower than 0.4. Against the existing problems, various scenarios for upgrading the acoustic environment of waiting room are presented and proved to be efficient. Moreover, from the optimization process, it is certificated that the requirement of speech intelligibility can be also satisfied when appropriately relaxing the reverberation time limits specified in the regulation.

11:45

4aAA10. Acoustical considerations in design and construction of Turkish Contractors Association headquarters. Zühre Sü Gül (R&D, MEZZO Studyo LTD., METU Technopolis KOSGEB-TEKMER No112, ODTU Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com) and Mehmet Caliskan (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

A LEED candidate building of Turkish Contractors Association Headquarters is a prestige building located in Ankara, the project and construction of which is sponsored by leading contractor companies of Turkey. As it represents the status of Turkish construction industry, from concept phase to the very recent inauguration of the building, the major consideration has been the application of latest technology and concepts such as sustainability within the scheme of building that can pioneer future works in the field. Chilled beam ventilation is one example of new technologies applied in the building system design that takes into account high energy efficiency with minimum use of fuel or natural sources. In acoustical terms, the building envelope and structural members together with interior and environmental noise sources in relation to the building services are studied. In order to provide acoustical comfort levels in acoustically sensitive spaces and to control noise and vibration at the source and sound paths, materials and methods are developed. Specifically acoustical interventions and solutions proposed for multi-purpose hall, offices, board and meeting rooms, foyers, mechanical rooms, roof-top units, and generators located close by at the site are discussed within the context of this paper.

THURSDAY MORNING, 8 MAY 2014 — 554 A/B, 8:00 A.M. TO 12:00 NOON

Session 4aAB

Animal Bioacoustics: Acoustics as a Tool for Population Structure II

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Southwest Fisheries Sci. Ctr., 8901 La Jolla Shores Dr., La Jolla, CA 92037

Kathleen Stafford, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Contributed Papers

8:00

4aAB1. Detecting and locating manatees in a zero visibility environment. Mario R. Rivera-Chavarria (Comput. Sci., Universidad de Costa Rica, Centro de Investigaciones en Tecnologías de la Información y Comunicación, Universidad de Costa Rica, Sede, Montes de Oca, 30101, San Jose, Costa Rica, San Jose 2060, Costa Rica, mariorrivera@gmail.com), Hector Guzman (Smithsonian Tropical Res. Inst., Panama, Panama), and Jorge Castro (Centro Nacional de Alta Tecnologia, San Jose, Costa Rica)

Manatees are an endangered species, and their population numbers have not been estimated in Panamian wetlands. Traditional monitoring methodologies using aerial surveys and visual sighting are ineffective for such turbid environments. We did a nine-month passive-acoustic survey using a Kayak with a stereo hydrophone array to detect and locate west Indian manatees in San San Pond Sak and Changuinola rivers in Panama. Twice a day transects with a total covering of 1700 km resulted 110 localizations and the recording of 1339 manatee vocalizations. Individual counting and the identification of biologically relevant sites is possible based on passive acoustics. Only a 2% of the acoustic detections were accompanied by clear sightings, indicating that visual census methodologies are ineffective in turbid environments and drastically underestimate manatee populations. We recommend this low-cost methodology to estimate manatee population more reliably in previously unsurveyed areas.

8:15

4aAB2. Fin whale song characteristics recorded on ocean bottom seismometers in the Northeast Pacific Ocean. Michelle Weirathmueller and William S. D. Wilcock (School of Oceanogr., Univ. of Washington, 1503 NE Boat St., Seattle, WA 98105, michw@uw.edu)

Fin whales produce low frequency sequences of vocalizations that can be detected opportunistically on ocean bottom seismometers (OBSs). Using an automatic detection algorithm, we have analyzed fin whale calls recorded on OBSs in the Northeast Pacific Ocean over broad spatial and temporal scales. The Cascadia Initiative experiment consists of 70 OBSs deployed for a total of four years (2011–2015). It extends from Vancouver Island to Cape Mendocino, and several hundred kilometers offshore. Additional OBS data that overlap spatially with the northern portion of the Cascadia Initiative instruments are available from the Neptune Canada cabled observatory, which has been online since 2009, and from standalone deployments between 2003 and 2006. With this study, we examine call characteristics and seasonal call counts for patterns that might indicate migratory movements or distinct acoustic populations. Both frequency and inter-pulse interval (IPI) are automatically extracted for each detected call and seasonal and inter-annual calling patterns are examined using daily binned call count histograms. Preliminary analysis of a subset of Cascadia Initiative data from 2011 to 2013 shows a dominant sequence of alternating classic and backbeat calls at center frequencies of 20 and 18.5 Hz, respectively, and preceding IPIs of 16 and 18 s, respectively.

8:30

4aAB3. Characteristics of sounds detected and localized in Hawaiian waters in Oct. 2013 believed to be from a Bryde's whale. Stephen W. Martin and Brian M. Matsuyama (SSC PAC, 53560 Hull St., Code 71510, San Diego, CA 92152, steve.w.martin@navy.mil)

Pulsed acoustic sounds suspected to be from a single Bryde's whale (*Balaenoptera edeni*) were automatically detected and localized in real time between 1130 and 1304 local time on 6 August 2013 utilizing hydrophones at the Pacific Missile Range Facility, Hawaii. The bottom mounted hydrophones are located 40 km to 80 km northwest of the Napali Coast of Kauai in waters over 4 km in depth. The localized sounds moved from east to west on a course of 294 degrees true for a distance of ~21.6 km with an average speed of 13.8 km/h, which is within the range reported for Bryde's whales. The sounds resemble those previously identified as being from Bryde's whales associated with visual sightings (Oleson *et al.* 2003) and acoustic only observations (Heimlich *et al.* 2005). Detailed analysis of the sounds revealed 27 emissions over the period with an inter-pulse interval of 216.6 s (SD, 69.4 s; range, 33–358 s). The duration of the sounds was approximately 1.8 s with major energy apparent at 33 Hz exhibiting burst tonal characteristics often with lower frequency tonal content. Generic calibration data for the hydrophones allows estimation of the source levels of the sounds, which fit within the range previously reported for the species (Cummings *et al.* 1986).

8:45

4aAB4. Trends and variations in the baseline soundscape of America's first offshore wind farm. T Aran Mooney, Maxwell B. Kaplan, Annamaria Izzi, and Laela Sayigh (Biology Dept., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, amooney@whoi.edu)

With the development of Cape Wind, Nantucket Sound, Massachusetts may become home to America's first offshore wind farm. The goal of this ongoing project is to establish the baseline (pre-construction) soundscape of anthropogenic and biological activity, including diel and seasonal variability of various sound types, at the construction site and nearby comparison sites. Acoustic recorders have been deployed since April 2012, recording on a 10% duty cycle (sample rate: 80 kHz). Multiple fish sounds have been identified with the predominant signals attributed to cusk eels (Family Ophidiidae). Cusk eel sounds consist of a series of pulses, with energy between 400 and 2500 Hz. They are detectable from April to October, with dense choruses occurring during the summer months. Sound energy levels during these choruses increased near the hours of sunrise and sunset. Vessel traffic also showed diel and seasonal trends, with peaks during the daytime and in the summer. These trends in biological and human activity provide key baseline records for evaluating the possible influence of wind farm construction and operation on a local US soundscape.

9:00

4aAB5. Algorithmic analysis of sounds using morphometric methods. Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

The vast diversity of animal sounds makes it difficult to analyze them in a quantitative, yet general way. Morphological research faces the same scope of variation and has met the challenge of generally applicable quantitative analysis by using landmarks to describe shapes. Spectrograms are a ubiquitous tool in bioacoustics research because they portray sounds visually—as shapes. The SoundPoints application provides a modular set of algorithms that reduce time-varying signals, especially sounds, into sets of landmarks. The landmark stage of analysis provides a layer of abstraction between feature detection and statistics or pattern recognition algorithms. As a result, it is possible to measure the large numbers of sounds that are needed to quantify variation at individual, population, and species levels. To demonstrate measures of stereotypy, I will present a developmental series of Swamp Sparrow (*Melospiza georgiana*) calls composed of more than 600 000 individual notes. To demonstrate spatial applications of classification, I will use a meta-population analysis of similarity among 22 000 Tree Swallow (*Tachycineta bicolor*) dawn song syllables.

9:15

4aAB6. Acoustic characterization and vocal behavior of North Atlantic right whale surface active groups. Edmund R. Gerstein (Charles E Schmidt College of Sci., Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33486, gerstein2@aol.com), Vasilis Trygonis (FAU / Harbor Branch Oceanogr. Inst., Lesvos island, Greece), Jim Moir (Marine Resources Council, Stuart, FL), and Steve McCulloch (FAU / Harbor Branch Oceanogr. Inst., Fort Pierce, FL)

Focal acoustic surveys were conducted to assess the vocal behavior of North Atlantic right whales in the shallow waters of the southeast critical habitat. Underwater vocalizations were archived using autonomous buoys in close proximity to surface active groups (SAGs) providing sound production data vital for regional passive acoustic monitoring and conservation. Classification trees were used to examine the distinguishing characteristics of calls and quantify their variability within the surface active groups vocal repertoire. Calling rates were higher than those reported in the Bay of Fundy, which may be a factor of habitat demographics. Sound production rate and call type usage were correlated with group cohesion, revealing a consistent call distribution pattern across SAGs of varying sizes and composition. The within-bout clustering probability of low and high frequency calls suggest that temporal affinities between vocalization classes may be indicators of shared social functions. The results demonstrate that concurrent temporal and spectral analysis is powerful for investigating and presenting the interrelationships of calls with social behavior and group composition.

9:30

4aAB7. Using relative Doppler from multiple observations of dolphin whistles as an aid to localization and tracking. Paul Hursky (HLS Res. Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

A variety of techniques have been used to track marine mammals from their vocalizations. For example, hyperbolic fixing and cross-fixed beams are well-developed approaches, both requiring multiple separated sensors. Dolphin vocalizations, consisting of clicks and whistles, pose some interesting challenges as well as opportunities. Dolphins are often observed in groups and their vocalizations can be quite dense in time. Before their sounds can be used on separated sensors, they must be associated, so that different sounds, for example, are not mistakenly assumed to be the same sound. Since it is often difficult to distinguish different clicks, it becomes difficult to associate (and thus track) them, when there are a lot of them (either from the same animal, or from many animals). By contrast, whistles typically are much easier to associate, even if overlapping. Whistles last seconds at a time and have distinctive melodies that span tens of kilohertz in bandwidth, often with harmonics. The high frequency of dolphin whistles and the fact that these animals are in constant motion suggests a novel feature to incorporate into their tracking—we discuss using relative Doppler, estimated from observations of whistles on multiple separated sensors, as an aid to localization and tracking.

9:45

4aAB8. Large-scale automatic acoustic monitoring of African forest elephants' calls in the terrestrial acoustic recordings. Yu Shiu, Peter H. Wrege, Sara Keen, and Elizabeth D. Rowland (BioAcoust. Res. Program, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, atoultaro@gmail.com)

African forest elephants live in the rain forests of western and central Africa. The dense habitat prevents them from communicating visually within the family group. Automatic detection of African Forest Elephants' calls intercepts signals in their communication channel and enables fast processing of large scale acoustic data. In this work, first, an automatic detection system targeting at African forest elephants' rumble calls is proposed. De-noising pre-processing, design of acoustic feature vectors, and choice of classifiers are discussed respectively. Second, the detector's performance is evaluated by the cross-validation of a 432-h of acoustic recording from eight locations in Gabon, Africa. It shows that the detector achieve 79.19% true positive rate when the false positive number is the low 5.70 per hour. The F1-score (geometric mean of precision and recall) is around 0.77 when relatively high score threshold (over 0.8) is selected. Finally, a case study demonstrates the results of applying our automatic detection system to a large-scale data set, which amounts to 420 days of acoustic recording over 3 years from the Ivindo National Park, Gabon. Visualization of the call activities reveal the seasonal and daily patterns as well as the temporal variation over the 3 years.

10:00

4aAB9. Single-sensor, cue-counting density estimation of highly broadband marine mammal calls. Elizabeth T. Küsel, Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., Ste. 160, 1900 SW 4th Ave., Portland, OR 97201, ekusel@pdx.edu), and David K. Mellinger (Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR)

Odontocete echolocation clicks have been used as a preferred cue for density estimation studies from single-sensor data sets, studies that require estimating detection probability as a function of range. Many such clicks can be very broadband in nature, with 10-dB bandwidths of 20 to 40 kHz or more. Because detection distances are not realizable from single-sensor

data, the detection probability is estimated in a Monte Carlo simulation using the sonar equation along with transmission loss calculations to estimate the received signal-to-noise ratio of tens of thousands of click realizations. Continuous-wave (CW) analysis, that is, single-frequency analysis, is inherent to basic forms of the passive sonar equation. Considering transmission loss by using CW analysis with the click's center frequency while disregarding its bandwidth has recently been shown to introduce bias to detection probabilities and hence to population estimates. In this study, false killer whale (*Pseudorca crassidens*) clicks recorded off the Kona coast of Hawai'i are used to quantify the bias in sonar equation density estimates caused by the center-frequency approach. A different approach to analyze data sets with highly broadband calls and to correctly model such signals is also presented and evaluated. [Work supported by ONR.]

10:15–10:30 Break

10:30

4aAB10. Residency of reef fish during pile driving within a shallow pier-side environment. Joseph Iafrate, Stephanie Watwood (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, joseph.iafrate@navy.mil), Eric Reyier (InoMedic Health Applications, Inc., Environ. Services, Kennedy Space Ctr. Ecological Program, FL), Matthew Gilchrest, and Steven Crocker (McLaughlin Res. Corp., Naval Undersea Warfare Ctr., Newport, RI)

The potential effects of pile-driving on fish populations have received significant attention with the prevalence of construction at in-shore areas throughout the world. In this study, the movement and survival of free-ranging reef fish in Port Canaveral, Florida, in response to pile driving for 35 days at an existing wharf was examined through the use of acoustic telemetry. Twenty-seven Sheepshead (*Archosargus probatocephalus*) and 13 mangrove snapper (*Lutjanus griseus*) were monitored for a period of approximately 11 months. Underwater acoustic receivers were deployed within Port Canaveral to complement an existing array of compatible receivers spanning a range of over 300 kilometers (km) along the east coast of Florida. Baseline residency and diel patterns of movement were compared for fish in two adjacent locations with and without disturbance before, during, and after the event. There was a significant decline in residency index for mangrove snapper at the construction wharf noted during the pre-during period. Also, 16 of 25 fish tagged at the construction wharf were detected 3-months post tagging, and 11 fish were detected 6-months post tagging. Although there was no apparent impact on patterns of behavior for resident reef fish populations, alterations on behavior of individual fish were noted, including displacement.

10:45

4aAB11. Investigating the relationship between foraging odontocetes and ocean acoustic biomass off the Kona coast of the Island of Hawaii. Adrienne M. Copeland (Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, acopelan@hawaii.edu), Whitlow Au, Giacomo Giorli (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Honolulu, HI), and Jeffrey Polovina (Pacific Islands Fisheries Sci. Ctr., NOAA, Honolulu, HI)

To understand the distribution of deep diving odontocetes, it is important to investigate the relationship between foraging whales and their prey. Tagged sperm whales have been documented to dive as deep as 1202 m. Short-finned pilot whales in Hawaii dive deeper during the day down to 600–800 m and shallower dives at night, driven possibly by the migration of organisms at night. Foraging sperm and pilot whales off the Island of Hawaii were located using a hydrophone array detecting echolocation clicks. A 500 m by 500 m active acoustics survey box was set up over two foraging sites: one during the night above foraging sperm whales and one during the day over foraging pilot whales. A four-frequency (38, 70, 120, and 200 kHz) split-beam echosounder collected acoustic data over foraging populations and non-foraging control sites of a similar bottom depth and time. The Nautical Acoustic Scattering Coefficient (NASC) or acoustic biomass ($m^2 nmi-2$) profile over the complete water column was statically compared over foraging and non-foraging populations to analyze the relationship between foraging and ocean biomass.

11:00

4aAB12. Estimating the range of Baleen whale calls recorded by hydrophone streamers during seismic surveys. Shima H. Abadi (Lamont–Doherty Earth Observatory, Columbia Univ., 122 Marine Sci. Bldg., University of Washington 1501 NE Boat St., Seattle, Washington 98195, shimah@ldeo.columbia.edu), Timothy J. Crone, Maya Tolstoy (Lamont–Doherty Earth Observatory, Columbia Univ., Palisades, NY), William S. D. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA), and Suzanne M. Carbotte (Lamont–Doherty Earth Observatory, Columbia Univ., Palisades, NY)

Like all sources of anthropogenic sound in the oceans, seismic surveys have the potential to disturb marine mammals and impede their communications. Since airgun arrays produce a considerable amount of low frequency energy, their impact on Baleen whales may be most significant. For these reasons, extensive mitigation efforts accompany seismic surveys, including visual and acoustic monitoring, but additional approaches could be useful to verify these efforts and study the behavior of whales. One approach is to utilize the hydrophone streamer to detect and locate calling Baleen whales. To develop this method, data are being analyzed from a seismic reflection survey conducted with the R/V Langseth off the coast of Washington in summer 2012. The seismic streamer is 8 km long with 636 hydrophones sampled nearly continuously at 500 Hz. The work focuses on time intervals when only a mitigation gun is firing because of marine animal sightings or turns at the ends of lines. Ranges and orientations are estimated by calculating the signal arrival angles for different groups of receivers. Data from the marine mammal observers on the R/V Langseth and other ships in the area are used to verify the analysis. [Sponsored by NSF.]

11:15

4aAB13. Soundscapes and vocal behavior of humpback whales in Massachusetts Bay. Nathan D. Merchant, Susan E. Parks (Dept. of Biology, Syracuse Univ., 107 College Pl., Syracuse, NY 13244, ndmercha@syr.edu), Sofie M. Van Parijs (Northeast Fisheries Sci. Ctr., NOAA Fisheries, Woods Hole, MA), David N. Wiley, Michael A. Thompson (Stellwagen Bank National Marine Sanctuary, Scituate, MA), and Ari S. Friedlaender (Duke Univ. Marine Lab., Beaufort, NC)

In recent years, technological advances have revolutionized the study of acoustic communication in marine mammals. Exciting new perspectives on vocal behavior, acoustic habitats, and the influence of noise on communication are offered by passive acoustic monitoring (PAM) platforms such as acoustic tags (DTAGs), autonomous PAM recorders, drifting PAM buoys, and subsea gliders. These innovations bring the opportunity to integrate data from fixed and mobile PAM devices to gain deeper insight into the dynamic interactions between marine mammal vocalizations, behavioral context, and the acoustic environment. In this study, we bring together such data sources to study the vocal behavior and acoustic habitat of humpback whales in the context of their spring and summer feeding grounds. Recordings were made in Stellwagen Bank National Marine Sanctuary during 2008–2010, using arrays of autonomous PAM recorders and DTAGs. In addition, AIS ship-tracking data were obtained to study the influence of vessel movements. We present preliminary findings of this work and discuss future strategies for analyzing the spatiotemporal interactions between vocal behavior and acoustical context.

11:30

4aAB14. Measuring the sonic, infrasonic and seismic soundscape of the Southern White Rhinoceros *(Ceratotherium simum simum)* at a wildlife park conservation center. Suzi Wiseman (Environ. Geography, Texas State Univ.-San Marcos, 3901 North 30th St., Waco, TX 76708, sw1210txstate@gmail.com), Preston S. Wilson (Mech. Eng., Univ. Texas at Austin, Austin, TX), and Frank Sepulveda (Geophys., Baylor Univ., Killeen, TX)

Many creatures, including the myopic rhinoceros, depend upon hearing and smell to determine their environment. Nature is dominated by biophonic and geophonic sounds quickly absorbed by soil and vegetation, while anthrophonic urban soundscapes exhibit vastly different physical and semantic characteristics, reflecting off hard geometric surfaces, distorting and reverberating, and becoming noise. Noise damages human

physiologically, including reproductively, and likely damages other mammals. Rhinos vocalize sonically and infrasonically but audiograms are unavailable. They generally breed poorly in urban zoos, where infrasonic noise tends to be chronic. Biological and social factors have been studied but little attention if any has been paid to soundscape. To comprehensively describe the rhinos' sonic, infrasonic and seismic environment at Fossil Rim Wildlife Center, one of the few U.S. facilities to successfully breed white rhinos in recent years, I began by comparing the sound metrics at different times of day in categories, for example, during visitation hours versus park closure. Further analysis will seek particular parameters known to be injurious to humans, plus those already known to impact animals. Later, the soundscapes of other facilities could be compared to seek correlations between their soundscapes and the health and well-being of the rhinos within their care.

11:45

4aAB15. Near real-time detection, beam-forming, and telemetry of marine mammal acoustic data on a wave glider autonomous vehicle. Harold A. Cheyne, Dean Hawthorne (Lab of Ornithology, Cornell Univ., 95 Brown Rd., Rm. 201, Ithaca, NY 14850, haroldcheyne@gmail.com), Charles R. Key, and Michael J. Satter (Leidos, Long Beach, MS)

Impacts of anthropogenic noise on marine mammals are becoming increasingly important for regulatory and research study, yet assessing and mitigating these impacts is hindered by current technology: archival underwater acoustic recorders have their data analyzed months after the activity of interest, and towed hydrophone arrays suffer from nearby ship and seismic air gun noise. This work addresses these drawbacks by developing an acoustic data acquisition and transmission system for use with a Wave Glider, to provide near real-time data for marine mammal monitoring and mitigation. The goal of the system is to be capable of months of autonomous monitoring in areas that would otherwise not be surveyed, and to transmit acoustic data within minutes of acquisition to enable rapid mitigation. Sea tests have demonstrated the proof-of-concept with the system recording four channels of acoustic data and transmitting portions of those data via satellite. Ongoing work is integrating a detection-classification algorithm onboard the Wave Glider and a beam-forming algorithm in the shore-side user interface, to provide the user with a topographic view of the Wave Glider; a sound source direction estimate; and aural and visual review of the detected sounds.

THURSDAY MORNING, 8 MAY 2014 — BALLROOM E, 9:00 A.M. TO 11:20 A.M.

Session 4aBA

Biomedical Acoustics: Biomedical Applications of Low Intensity Ultrasound I

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

Invited Papers

9:00

4aBA1. Low intensity ultrasound—Diverse biomedical applications. Thomas L. Szabo (Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu)

Low intensity ultrasound has been useful in a surprisingly wide range of biomedical applications. Ultrasound can affect both the central nervous system (CNS; brain and spinal cord) and the peripheral nervous system (PNS). Ultrasound methods offer the possibility of stimulating receptor structures and unmyelinated nerve fibers not just on the surface but also those otherwise inaccessible, deeper in the body or brain. Preliminary work indicates that information can be transferred via unmyelinated nerve fibers when normal avenues of sensing are damaged or inoperative. Tones and acoustic speech have been directly transferred through nerves without direct hearing. Radiation force effects can provide a variety of sensational effects and can be employed in ultrasound dynamic tactile arrays. Early prototype devices for diagnosis of diseases of hearing and audio prosthetics were developed in the early 1980s in the Soviet Union by Gavrilov and Tsirulnikov. Neuromodulation of the brain transcranially has demonstrated activation of motor responses. Low intensity ultrasound has also been applied to bone and wound healing. In a related application, it has been applied to the growth of artificial neovessels.

9:20

4aBA2. Ultrasonic neuromodulation: Three conjectures common to the peripheral and the central nervous systems. Robert Muratore (Quantum Now LLC, 49 Cedar Dr., Huntington, NY 11743, wave@quantumnow.com)

The nervous system responds with finesse to incident ultrasound. Three conjectures are proposed, abstracted from the literature, which illustrate commonalities in the response of the peripheral and the central nervous systems, and serve as an introduction to the nascent field of ultrasonic neuromodulation. (1) A mathematical function fit to neuronal effect vs. acoustic dose has a root at a non-trivial dose. Above a threshold dose, nerves and brain regions are stimulated; at very high doses, normal neuronal activity is inhibited. Thus, there exists an intermediate dose balancing stimulus and inhibition. (2) An acoustic beam can modulate a neuronal region larger than

that which it insonifies. Nerves can be stimulated by insonifying a small portion of their axon. Brain regions exhibit responses, such as spreading depression, to localized insonification. (3) The spatial precision of ultrasonic neuromodulation can be considerably finer than the incident acoustic beam width. Thicker nerve fibers are more resistant to the effects of incident ultrasound than are thinner fibers in the same nerve. Across the cortex, displacements of acoustic beams smaller than the beam width can achieve fine motor control. Each of these conjectures plays a role in current neuromodulation experiments.

9:40

4aBA3. Capacitive micromachined ultrasonic transducers with integrated electronics for neuromodulation applications. Butrus T. Khuri-Yakub (E. L. Ginzton Lab., Stanford Univ., Spilker 217, Stanford, CA 94305, khuri-yakub@stanford.edu)

Capacitive Micromachined Ultrasonic Transducers (CMUT) are being made in practically any size (microns to mms), shape (flat or curved), and type (single element, 1-D array, 2-D array, rings, and annular arrays), and at frequencies from 10s of kHz to almost 100 MHz. Along with the transducers themselves, front-end electronics are being integrated as well to provide better performance and enable the use of arrays with a very large number of elements. One important aspect of these integrated arrays is that they can be used for imaging (anatomic and photo-acoustic functional), therapy (high intensity focused ultrasound), and more recently neuro-modulation. This talk will review CMUTs and the methods of integration, then show examples of ultrasound stimulation of lipid bilayers and Salamander retina. We show that the retina responds to ultrasound stimulation as well as it responds to light stimulation and that when the retina's optical response is suppressed chemically it still responds to ultrasonic stimulus. We postulate the possibility of using CMUT 2D arrays as contact lens prosthetic devices capable of restoring some vision in some type of blindness.

10:00

4aBA4. Localization of ultrasound induced *in-vivo* neurostimulation in the mouse model. Randy L. King (DSFM, US FDA, WO62 rm 2217, 10903 New Hampshire Ave., Silver Spring, MD 20993-0002, Randy.King@fda.hhs.gov)

Developments in the use of ultrasound to stimulate and modulate neural activity have raised the possibility of using ultrasound as a new investigative and therapeutic tool in brain research. The phenomenon of ultrasound induced neurostimulation has a long history dating back many decades, but until now there has been little evidence demonstrating a clearly localized effect in the brain, a necessary requirement for the technique to become genuinely useful. Here, we report clearly distinguishable effects in sonicating rostral and caudal regions of the mouse motor cortex. Motor responses measured by normalized EMG in the neck and tail regions changed significantly when sonicating the two different areas of motor cortex. Response latencies varied significantly according to sonication location suggesting that different neural circuits are activated depending on the precise focus of the ultrasound beam. Taken together our findings present good evidence for being able to target selective parts of the motor cortex with ultrasound neurostimulation in the mouse, an advance that should help to set the stage for developing new applications in larger animal models including humans.

10:20

4aBA5. Ultrasound for microvascular tissue engineering. Diane Dalecki (Biomedical Eng., Univ. of Rochester, 310 Goergen Hall, Rochester, NY 14627, dalecki@bme.rochester.edu) and Denise C. Hocking (Pharmacology and Physiol., Univ. of Rochester, Rochester, NY)

A critical obstacle currently facing the field of tissue engineering is the need for rapid and effective tissue vascularization strategies, both during construct development and upon implantation. To address this challenge, we have developed an ultrasound technology for microvascular tissue engineering. The technology utilizes radiation forces in an ultrasound standing wave field to rapidly and non-invasively spatially pattern cells in 3D within hydrogels. Ultrasound-induced patterning of endothelial cells accelerates the emergence of capillary-like sprouts stimulates cell-mediated collagen fibril alignment and results in the maturation of sprouts into lumen-containing microvessel networks throughout collagen hydrogels. Importantly, the morphology of resultant microvessel networks can be controlled by design of the acoustic field employed during fabrication. Specifically, the technology can produce microvascular networks having two distinct, physiologically relevant morphologies; one composed of a tortuous, capillary-like network, and one composed of hierarchical branching vessels (arteriole/venule-like). We have extended the versatility of the technology to lymph endothelial cells and have demonstrated the ability to engineer 3D lymphatic microvessel structures. Thus, this ultrasound technology holds promise as a new approach to induce microvascular network formation and direct vascular morphology in engineered tissues.

10:40

4aBA6. Low intensity (55 kPa) 20 kHz ultrasound heals venous ulcers. Joshua Samuels (Dept. of Biomedical Eng., Drexel Univ., Philadelphia, PA), Michael S. Weingarten (Dept. of Surgery, Drexel Univ. College of Medicine, Philadelphia, PA), Leonid Zubkov, Christopher Bawiec, Youhan Sunny (Dept. of Biomedical Eng., Drexel Univ., Philadelphia, PA), Jane McDaniel, Lori Jenkins (Dept. of Surgery, Drexel Univ. College of Medicine, Philadelphia, PA), David Margolis (Dept. of Epidemiology, Univ. of Pennsylvania Perelman School of Medicine, Philadelphia, PA), and Peter Lewin (Dept. of Biomedical Eng., Drexel Univ., 3141 Chestnut St., BIOMED DEPT, Philadelphia, PA 19104, plewin@coe.drexel.edu)

We report the results of a second clinical pilot study ($n = 19$) involving treatment of chronic wounds (venous ulcers) using novel, fully wearable ultrasound array applicator operating at 20 kHz and generating pressure amplitudes close to 55 kPa (about 100 mW/cm^2, Sptp). The applicator was designed as compact, tether-free, device that can be comfortably worn by subjects at home, permitting active (combined with traditional compression) therapy away from the clinical setting. Patients with venous ulcers documented for over 8 weeks were enrolled from the Drexel Wound Healing Center and, following consent, were randomly assigned into treatment or control groups. Patients were treated weekly (15 min) for a maximum of 12 visits or until wound closure. Treatments were in addition to standard of care compression therapy as ordered by the physician. Of the patients receiving at least three treatments ($n = 16$), the ultrasound treated group had statistically improved ($p < 0.04$) rate of wound closure (reduction of 8.2%/wk) compared to the rate of wound closure for the control group (increase of 7.5%/wk on average). This study represents further proof of the potential healing power of low intensity, low frequency ultrasound. Optical measurements and *in-vitro* work continue to support these findings as well.

11:00

4aBA7. Enhanced fracture repair and mitigation of fracture-healing risk factors using low-intensity pulsed ultrasound. Christopher R. Brodie (Bioventus LLC, 4721 Emperor Blvd, Ste. 100, Durham, NC 27703, chris.brodie@bioventusglobal.com) and Andrew Harrison (Bioventus LLC, York, United Kingdom)

Low-intensity pulsed ultrasound (LIPUS) is used clinically to enhance fracture healing. Level-I clinical studies demonstrate that a specific signal (1.5 MHz ultrasound pulsed at 1 kHz, 20% duty cycle, 30 mW/cm^2 SATA) can accelerate the healing of acute fractures. This result remains a unique benefit of LIPUS, and to date, no other drug or device has been approved by the FDA for accelerated fracture repair. The same signal has been shown in many studies to heal a high proportion of non-union fractures. LIPUS appears to be effective for all three types of non-unions—atrophic, oligotrophic and hypertrophic—even in the absence of revision surgery. The findings are broadly applicable to orthopedics, with similar results regardless of fracture type, fracture location and fracture-management technique. Given the varied causes of non-union, the ability of LIPUS to overcome a high proportion of obstacles to healing indicates that the signal is likely to have pleiotropic effects on multiple cell types within the healing process. Smoking, age, and diabetes are known risk factors for delayed union and nonunion. Clinical data, including randomized controlled trials and a registry of 1546 nonunion patients, suggest that LIPUS mitigates these risks and restores the course of normal bone healing.

THURSDAY MORNING, 8 MAY 2014 — 550 A/B, 8:15 A.M. TO 12:00 NOON

Session 4aEA

Engineering Acoustics: Session in Honor of Stanley Ehrlich

David A. Brown, Cochair
ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723

Kenneth G. Foote, Cochair
Woods Hole Oceanogr. Inst., 98 Water St., Woods Hole, MA 02543

Chair's Introduction—8:15

Invited Papers

8:20

4aEA1. Modal transducers and Stan Ehrlich. John L. Butler (Eng., Image Acoust., Inc, 97 Elm St., Cohasset, MA 02025, jbutler@imageacoustics.com)

There have been a number of transducer designs which use the dipole mode of a piezoelectric cylinder to obtain directionality since Stan Ehrlich's early patent (with P.D. Frelich), ["Sonar Transducer," U.S. Patent 3,290,646, December 6, 1966] was first published. There is now a whole class of transducers, called vector sensors or hydrophones, which use the dipole mode in one or more directions. In addition to this, other designs have emerged which use modes higher than the monopole and dipole modes. For example, the added use of the quadrupole mode has allowed beam patterns from cylinders which approximate patterns from piston transducers. Work on the dipole mode and higher modes of spherical transducers and arrays allow 3-D acoustical coverage from one transducer or array. This presenters interest in modal excitation from transducers and arrays began after reading Stan's patent, and interest developed further after working with Stan at Raytheon. A review of some of the transducers and arrays which we worked on will be presented, along with more recent work based on these modal concepts.

8:40

4aEA2. Multimode and other sonar transducer patents of Stanley Ehrlich. David A. Brown (Elec. Engineering/ATMC, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

Stanley Ehrlich was an innovative sonar transducer designer and inventor who made many important contributions to the field of acoustic transduction at Raytheon Submarine Signal Division in Portsmouth, Rhode Island. With the advent of searchable digital patent archives at the USPTO and google/patents, it is now relatively easy to review Stan's patents and gain a glimpse at his innovation and creativity. This presentation reviews some these patents including: Sonar Transducer, that describes a multimode transducer producing simultaneously two dipole patterns with mutually perpendicular acoustic axes and an omnidirectional pattern; a Spherical [Multimode] Acoustic Transducer (#3732535 filed 1969) that enables the radial and circumferential vibrating modes of the acoustically excited sphere to be processed to determine bearing. The presentation also draws connections of these and other Ehrlich inventions to more recent ongoing works in multimode transducers.

9:00

4aEA3. Nearfield of an electroacoustic transducer, with implications for performance measurement. Kenneth G. Foote (Woods Hole Oceanogr. Inst., 98 Water St., Woods Hole, MA 02543, kfoote@whoi.edu)

The spatial structure of the nearfield of an electroacoustic transducer is known to be complicated. This is illustrated by numerical modeling of the nearfield of an ideal planar circular piston in a rigid, infinite baffle. There are implications for performance measurements of electroacoustic transducers including hydrophones in tanks.

9:20

4aEA4. Analysis of nonuniform circular flexural piezoelectric plate transducers. Boris Aronov (ElectroAcoust. Res. Lab. - ATMC, BTech Acoust. LLC, Fall River, MA) and David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

An analytical treatment of the circular flexural plate transducer having a nonuniform electromechanically active-passive (bilaminar) mechanical system is presented. The analysis is made using the energy method that was previously applied to calculating the parameters of uniform fully active (bimorph) circular plate transducers [Aronov, J. Acoust. Soc. Am. **118**(2), 627–637 (2005)]. It is shown to first order approximation, that the vibration mode shapes do not change significantly for a large range of relative dimensions of the active and passive laminates of the mechanical system considered in terms of optimizing the effective coupling coefficient of the transducer. Therefore, the transducer can be considered as having a single degree of freedom, and its operational characteristics can be calculated using the same technique as previously used for uniform plates. The dependence of the resonance frequencies, effective coupling coefficients, and parameters of the equivalent electromechanical circuit on the relative dimensions of the active and passive laminates for several combinations of the active and passive materials are presented. The main results are in a good agreement with experimental data.

9:40–10:00 Break

Contributed Papers

10:00

4aEA5. Planar microphone based on piezoelectric electrospun poly (γ-benzyl-?,L-glutamate) nanofibers. James E. West, Kailiang Ren (ECE, Johns Hopkins Univ., 3400 N. Charles St., Barton Hall 105, Baltimore, MD 21218, jimwest@jhu.edu), and Michael Yu (ECE, Johns Hopkins Univ., Salt Lake City, Utah)

Velocity and pressure microphones comprised of piezoelectric poly (γ-benzyl, L-glutamate) (PBLG) nanofibers were produced by adhering a single layer of PBLG film to a Mylar diaphragm. The device exhibited a sensitivity of 65 dB/Pa in air, and both pressure and velocity response showed a broad frequency response, which was primarily controlled by the stiffness of the supporting diaphragm. The pressure microphone response was 3 dB between 200 Hz and 4 kHz when measured in a semi-anechoic chamber. Thermal stability, easy fabrication, and simple design make this single element transducer ideal for various applications including those for underwater and high temperature use.

10:15

4aEA6. Experimental results of motional current velocity control intended for broadband piezoelectric projectors. Robert C. Randall (Raytheon, 188 Hanover St. Apt. 3, Fall River, MA 02720, bobrandall81@gmail.com), David A. Brown (Univ. of Massachusetts Dartmouth, Barrington, Rhode Island), and Corey Bachand (BTech Acoust. LLC, New Bedford, MA)

Velocity control with active feedback can be useful for flattening a projector's frequency response, reducing distortion, and mitigating array interaction effects. This has been demonstrated and commercialized for HiFi audio, but has seen little attention for underwater SONAR and communications applications. The benefits and tradeoffs of using velocity control to drive an underwater piezoelectric transducer or array of transducers is presented, comparing array beam patterns both with and without velocity control. The theoretical effectiveness of motional current velocity control is discussed for various piezoelectric loads with coupling coefficients ranging from 0.3 to 0.9. The utility of using a digital feedback amplifier and in situ calibration methods with this approach is discussed. A prototype Class D amplifier using motional current feedback driving an equivalent circuit load for a BTech Acoustics single crystal segmented cylinder is presented. Experimental results of frequency response, bandwidth, and feedback stability are also considered.

10:30

4aEA7. Automated parameter fitting of two-port network transducer models. Daniel M. Warren (Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60134, daniel.warren@knowles.com)

Networks of linear lumped-parameter components, as may be simulated in electronic circuit design software such as SPICE, are a lightweight, fast, and convenient means of predicting the behavior of electroacoustic transducers in practical application. The development of these models can be quite burdensome. The selection and networking of the equivalent electronic components requires specialized domain knowledge of the transduction mechanisms and design of a transducer. Previously, a general network synthesis approach was proposed [Warren, Daniel, "Applications of network synthesis and zero-pole analysis in transducer modeling," J. Acoust. Soc. Am. **133**, 3360–3360 (2013)] but was deemed to be an unreliable and awkward means of developing transducer networks in practice. However, networks that represent transducer behavior are generally well-known for a given transducer type and design. The more difficult task, or, at least, the more often performed and thereby repetitive task, is the selection of component parameter values which correctly predict the transducer's behavior under all electrical drive and acoustical loading conditions which may be encountered in practical application. The approach taken here is to assume that the network itself is already known and seek to develop an automated means of determining the correct parameter values.

10:45

4aEA8. Model for the design of a pressure actuated self-sustained oscillator as an acoustic projector. Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., N-249 Millennium Sci. Complex, University Park, PA 16803, sct12@psu.edu)

Wind musical instruments are examples of a pressure operated self-sustained oscillator that acts as an acoustic projector. Recent studies have shown that this device can also be implemented underwater. However the design parameters for such a device are necessarily different due to the large difference in medium density and acoustic impedance. This paper describes a model that is sufficient to predict the performance of the projector and to understand the effects of design changes on the performance.

11:00–12:00 Panel Discussion

Session 4aID

Interdisciplinary, Public Relations Committee, and Education in Acoustics: Effective Communication Between Acoustics Professionals and the Media

Andrew A. Piacsek, Cochair
Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926

Steven L. Garrett, Cochair
Grad. Prog. in Acoustics, Penn State, Appl. Res. Lab., P. O. Box 30, State College, PA 16804

Chair's Introduction—10:00

Invited Papers

10:05

4aID1. What to expect when you're expecting media calls. Jason S. Bardi (NMS, AIP, 1 Phys. Ellipse, College Park, MD 20740, jbardi@aip.org)

A decade's worth of college, grad school, and post-doc work, countless sleepless nights toiling in your own laboratory, a long route to discovery, your ultimate breakthrough and it has come to this: the phone is ringing. A reporter is on the line. What does she/he want? What should you say? I'm here to tell you, "Don't panic!" You have been preparing for this interview your entire professional career. You are one of the world's leading experts in your area, and that's one of the reasons why the reporter is calling. You also have a story to tell, the reporter wants to hear it, and the interview should be more conversation than inquisition. This talk will help you realize that, helping you make the most of your time in the spotlight by putting the PR and press process into perspective, offering some tips of the trade, describing your rights and responsibilities as a source, and sharing best practices for handling media inquiries.

10:20

4aID2. Why should a U.S. Navy researcher discuss cicada mating calls for hours with several journalist? Derke Hughes (NUWC-DIVNPT, 1176 Howell St., Newport, RI 02841, derke.hughes@navy.mil)

The first journalist to comment officially on my cicada research was by American Institute of Physics (AIP), which was quickly followed up by a LA Times reporter. Unfortunately, the LA Times author wrote how the cicada sound system functioned like a Helmholtz resonator. However, when the author actually interviewed me, I contradicted the journalist by saying that I do not believe that theory was correct. Furthermore, I was interviewed by a radio commentator for National Public Radio (NPR) as well who diligently discussed my research and its inspiration for 50 min. My segment aired for 2 min and those few seconds consisted of "crude construction worker harassment and bar talk." Overall, the opportunity of communicating my research to an international audience was outstanding. The other correspondents that interviewed me for the article were from the *ScienceNow* and *Wall Street Journal*. Also, one of my interviewers was a writer for Le Presse so the immediate columnist coverage did span at least two countries. I am proud to have bought awareness to an insect that has been on earth with civilized mankind for millennia; nonetheless, our practical knowledge of the cicada is rather limited.

10:35

4aID3. Communicating with the media: From the laboratory to the real world. Diana Deutsch (Univ. of California, San Diego, 9500 Gilman Dr. #0109, La Jolla, CA 92037, ddeutsch@ucsd.edu)

Scientists often view communicating with the media as a risky process, based largely on concerns that they might be held responsible for inaccuracies in reporting their work. Yet my experiences with the media have generally been very rewarding. Most frequently, those who have interviewed me have been well prepared and have thought broadly about the subject matter of my research. Our conversations have often induced me to think outside the box and have led to novel ideas for studies that might otherwise have been left undone. The potential for feedback has recently been enhanced by the development of social networks—these often pick up on reports in newspapers and magazines, and provide an important additional forum for discussion. In this talk, I describe some experiences that illustrate these points and offer some suggestions for interacting with the media so as to communicate research findings and their implications most effectively.

10:50

4aID4. We don't bite; we want to get it right. Really. Peter Spotts (The Christian Sci. Monitor, 210 Massachusetts Ave., Boston, MA 02115, pspotts@alum.mit.edu)

If a full-time science writer calls you for an interview, count yourself lucky. These days, full-time science writers are a vanishing breed. The reporter about to interview you may be just as nervous about the impending conversation as you are. You know the subject cold. He or she may have had little time to prepare. We'll take a brief look inside one news organization's day (mine) to understand the context on our side of the so-called divide, and share some thoughts on how you can help us explain what you do to your Aunt Elsie or Uncle Sid.

11:05

4aID5. On becoming an expert witness in a high-profile patent-dispute case. Mark F. Hamilton (Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Several years ago, the author was contacted by the legal team representing a major smartphone manufacturer and asked if he would serve as an expert witness in a patent-dispute case to be tried before an administrative law judge at the International Trade Commission. The author had no significant prior experience as an expert witness, and he therefore had no inkling of what responsibilities lay ahead of him. The author will describe his experiences in this case, beginning with assisting the legal team with understanding the relevant acoustics, then writing expert reports, and finally preparing for deposition and trial.

11:20

4aID6. Interviews with the interviewers and interviewees. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

The basic conflict between scientist and journalist is that each wants to tell a story—but not necessarily the same story. Can they agree on a version that grabs and holds the attention of most people but is still true to the science? Despite the risks of being portrayed inaccurately, should researchers make an effort to talk to the press? This talk will address these questions by synthesizing a series of interviews conducted with two acoustics professionals who have had significant media exposure, a print journalist, a radio journalist, and an academic specializing in science journalism.

11:35–12:00 Panel Discussion

THURSDAY MORNING, 8 MAY 2014 557, 8:30 A.M. TO 11:45 A.M.

Session 4aNS

Noise and ASA Committee on Standards: Community Noise

Robert D. Hellweg, Cochair
Hellweg Associates, Wellesley, MA

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square - Ste. 103, Vernon, CT 06066

Chair's Introduction—8:30

Invited Papers

8:35

4aNS1. Progress report—American National Standards Institute Community Noise Standard. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com) and Lawrence S. Finegold (Finegold & So, Consultants, Dayton, OH)

The American National Standards Institute (ANSI) Accredited Standards Committee S12 (Noise) Working Group (WG) 41 has been developing a draft community noise standard document for over 13 years. The purpose of the document is to provide guidance to government officials, acoustical consultants, and other interested persons on how to develop a community noise ordinance or regulation, which is appropriate for the existing local circumstances. The current version of the document embodies significant revisions, based on the inputs of many stakeholders in the community noise arena, including industry, government, consulting, and the public. The

document addresses issues such as public and government priorities and values, and available resources, and also provides the technical basis to manage the local sound environment. The keys to the effectiveness of the document are that it provides a menu of options for the user, discusses the trade-offs involved for decisions that must be made by government officials, and emphasizes that enforcement of a community noise ordinance is crucial to its success. Recent progress made by the Working Group in drafting this standard is reported.

8:55

4aNS2. Massachusetts Wind and Noise Technical Advisory Group—Status report. Christopher W. Menge (Harris Miller Miller & Hanson Inc., 77 South Bedford St., Burlington, MA 01776, cmenge@hmmh.com) and Robert D. ONeal (Epsilon Assoc., Inc., Maynard, MA)

In June 2013, the Commonwealth of Massachusetts launched a Community Wind Energy Initiative to provide support and guidance to municipalities, developers and stakeholders for land-based wind projects. The initiative convened a technical advisory group of experts to solicit input on wind turbine sound policy. This Wind and Noise Technical Advisory Group (WNTAG) is led by the Massachusetts Department of Environmental Protection (MassDEP) and includes other state agency representatives, wind energy experts, industry representatives, affected community representatives, health experts, and acoustical consultants. The WNTAG has met several times since July 2013 and has addressed many aspects of wind turbine noise that may influence and/or become a part of a new statewide noise policy for land-based wind turbines. In this presentation, the authors provide perspective on the process, progress toward a revised policy, and the policy and technical aspects that were discussed, which included absolute vs. relative noise criteria, noise level metrics, measurement protocols for compliance evaluation, amplitude modulation, and modeling approaches for pre-construction permitting.

9:15

4aNS3. Regulatory inertia and community noise. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org)

Thirty and forty year old regulations are determining much of the acoustic environment. This paper examines regulations that have not kept up with the times or technology. Aviation, motorcycle, and truck regulations are examined in their historical context, as well as OSHA's backup beeper regulations. Also, in many communities, local noise regulations are decades behind today's best practices. The options and prospects for updating these regulations are discussed.

9:35

4aNS4. Update on regulations adding noise to hybrid and electric cars. Dennis Weidemann (2633 Granite Rd., Fitchburg, WI 53771, dweid@mac.com) and Leslie D. Blomberg (Noise Pollution Clearinghouse, Montpelier, VT)

The United States National Highway Traffic Safety Administration in nearing and may have completed is rulemaking concerning adding noise to hybrid and electric cars by May 2014. This paper will examine what has happened in early 2014 with respect to these regulations. Updates to international regulations will also be presented.

9:55

4aNS5. Simulated and laboratory models of aircraft sound transmission in residences. Ashwin Thomas, Erica Ryherd, Thomas Bowling (Woodruff School of Mech. Eng., Georgia Inst. of Technol., c/o Erica Ryherd, Mech. Eng., Georgia Tech, Atlanta, GA, apthomas@gatech.edu), and Javier Irizarry (School of Bldg. Construction, Georgia Inst. of Technol., Atlanta, GA)

Current aircraft noise guidelines are based primarily on outdoor sound levels. As people spend the majority of their time indoors, however, human perception is highly related to indoor sound levels. Investigations are being made to provide further insight into how typical residential constructions affect indoor sound levels. A pilot, single-room "test house" has been built using typical mixed-humid climate region construction techniques and the outdoor-to-indoor transmission of sound was directly measured—with specific focus on continuous commercial aircraft signatures. The measurements included a variety of construction iterations (e.g., window type, wall construction) and a variety of instrumentation iterations (e.g., source and sensor locations). The results of this study are being used to validate and improve modelling software that simulates a wide range of construction types and configurations for other US climate regions. Overall, the project intends to improve the ability to predict acoustic performance for typical US construction types as well as for possible design alterations for sound insulation.

10:15–10:30 Break

Contributed Papers

10:30

4aNS6. Spatial regression relations between urban forms and road-traffic noise. Seo I. Chang (Environ. Eng., Univ. of Seoul, 163 Seoulsiripdae-ro, Dongdaemun-gu, Seoul 130-743, South Korea, schang@uos.ac.kr) and Bum Seok Chun (Ctr. for GIS, Georgia Inst. of Technol., Atlanta, GA)

Recent development of noise mapping tools allows us to generate sophisticated environmental noise maps where complicated acoustic phenomena including reflection by building facades, diffraction by horizontal and vertical edges of a building, and absorption by pavements can be considered with high level of accuracy. Therefore, if we have a noise map of an existing city and plan to do minor modifications, such as adding lanes to a road or locating new residential buildings along highways, we can assess and mitigate the induced impacts by simulating upon the existing noise map, e.g., installation of noise barriers or control of traffic flows. But, if a totally new city is built separately, what and how can we plan about the environmental noise? How can we do city-planning based on minimum information? What minimum information should be provided? Identification of the relations between urban forms and environmental noise can be helpful to city-planners at very early stage of planning. We performed spatial statistical analysis of road-traffic noise and urban forms by utilizing a GIS tool. Urban forms in the spatial regression model include residential and employee populations, building forms, traffic properties, and land-use pattern.

10:45

4aNS7. The spectral effect of masking of intruding noise by environmental background-noise. Giora -. Rosenhouse (Swantech, 89 Hagalil Str., Haifa 3268412, Israel, fwamtech@bezeqint.net)

Annoyance by noise depends strongly on its informative, spectral contents and individual effect on people. Yet, standards dictate certain formal limitations, ignoring such details. In practice, it happens in many cases of recreational areas, industrial premises and other kinds of activities, that even when results of measurements satisfy the standards limits, complaints do not stop, yielding threats of legal acts. Case studies of the effect, based on actual acoustic measurements are analyzed here, showing factors that cause extreme sensitivity to certain noise patterns, even if the total amount of noise remains unchanged. The effect of color difference is enhanced if the added noise has a certain periodicity, located where the background noise has lower masking effect. Since in many cases the background noise has less effect or resembles white or pink noise, certain noise sources can be clearly heard, if they include higher local amplitudes in the frequency spectra domain of the background noise. Acoustic solutions include means for undesired noise reduction to levels much below the background noise¸ by as much as by 9 dB, to allow background noise masking of disturbing sources. Such reduction alters its status from being strongly heard to the privacy zone.

11:00

4aNS8. Active control of traffic noise radiation and propagation. Qi Hu and Shiu-keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, KLN, Hung Hom Na, Hong Kong, qi.bs.hu@connect.polyu.hk)

Active noise control tends to be challenging, especially for open space scenario. This work intends to control road traffic noise that is treated as an ideal line source with finite length, actively through the introduction of an array of secondary point monopole sources to modify the original sound field, which accordingly creates a quiet zone for the noise sensitive receivers. Three dimensional analytical formulation and numerical simulation are performed to compare the difference before and after the introduction of control sources, through which the optimal position and strength of each control source are studied.

11:15

4aNS9. Investigating human annoyance thresholds of tones in noise from a dose-response relationship. Joonhee Lee, Jennifer M. Francis, and Lily M. Wang (Durham School of Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, joonhee.lee@huskers.unl.edu)

Noise with prominent tones from building mechanical systems is often detrimental to the environmental quality and leads to complaints. Previous studies have investigated the relationship between existing tonal noise metrics and human annoyance perception, but little is known about at what level the tones at assorted frequencies induce human annoyance. This paper investigates human annoyance responses due to noise with tones to produce a dose-response relationship for estimating the thresholds of annoyance to tones in noise. The subjective test is conducted using noise signals with varied loudness and tonalness thorough an Armstrong i-Ceiling system in the Nebraska indoor acoustic testing chamber. Binary logistic multiple regression models are used to predict the percentage of annoyed people or likelihood-to-complain with confidence intervals. This paper also examines the statistical performance of models with assorted noise metrics and non-acoustical variables to calculate the probability of occupants feeling annoyed for any given background noise with tonal components.

11:30

4aNS10. Differences between sound pressure levels measured outdoors in three heights commonly used in environmental noise impact assessment. Olmiro C. de Souza (UFSM, Undergraduate Program in Acoust. Eng., DECC-CT-UFSM, Av. Roraima 1000, Camobi, Santa Maria, RS 97105-900, Brazil, Acampamento, 569, Santa Maria, Santa Maria 97050003, Brazil, olmirocz.eac@gmail.com), Stephan Paul, and Diego Garlet (UFSM, Undergraduate Program in Acoust. Eng., DECC-CT-UFSM, Av. Roraima 1000, Camobi, Santa Maria, RS 97105-900, Brazil)

In environmental acoustics, outdoor sound pressure level can be measured at different heights over ground. ISO 1996 recommends measurements at 4 m, which is representative for multi-storey residential areas and in accordance with EU directives for noise map modeling. For other areas ISO 1996 recommends measurements 1.5 m over ground, a height that corresponds to the median height of adults ears. In Brazil, 1.2 m are commonly used for outdoor measurements and noise map calibrations as this height is recommended by the Brazilian standard NBR 10.151-2000. The goal of this work is to investigate the relationship between SPL measurements obtained at these three heights. Measurements were taken close to roads at a university campus, roads that in some cases have high traffic flows. The differences between measured A-weighted equivalent SPLs (LAeq) at the different heights were statistically analyzed. Difference distributions were found to be closely to normal distribution with some outliers. Mean values of the SPL differences remained below 3 dB. From the data obtained, it seems acceptable to calibrate a noise map model at a different height from the measured one using the mean difference as a correction term.

Session 4aPA

Physical Acoustics: Acoustic Radiation Forces, Streaming, and Applications

Bart Lipkens, Chair
Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Box S-5024, Springfield, MA 01119

Contributed Papers

8:30

4aPA1. Prediction of acoustic radiation forces in three dimensional flow through resonators. Ari Mercado and Bart Lipkens (Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Springfield, MA 01119, a.mercado@fdsonics.com)

In large scale acoustophoretic particle separation systems, the acoustic radiation force exerted on the particles exceeds the combined effect of the fluid drag force and gravitational force on the particle. This results in the trapping of the particles in the acoustic standing wave, followed by aggregation of the particles, and ultimately gravitational settling and separation of the secondary phase. The separation system typically consists of a flow chamber in which a three dimensional acoustic standing wave is generated by piezoelectric transducers. Accurate prediction models of the acoustic radiation force are needed so that they can be used as a tool in the design and development of such separation systems. The prediction model consists of two steps. First, COMSOL Multiphysics™ software is used to predict the acoustic field in the separation devices. Next, theoretical models [Gor'kov, Sov. Phys. Dokl. **6**, 773–775 (1962) and Ilinskii *et al*., J. Acoust. Soc. Am. **133**, 3237 (2013)] are used to calculate the acoustic radiation force on a suspended particle. Numerical results were verified by comparison with the theoretical results for a rectangular cavity [Barmatz and Collas, J. Acoust. Soc. Am. **77**, 928 (1985)]. [Work supported by NSF PFI:BIC 1237723.]

8:45

4aPA2. Design of a multi-element transducer for large volume acoustophoretic phase separation. Jason P. Dionne (FloDesign Sonics, Inc., 499 Bushy Hill Rd., Simsbury, Connecticut 06070, j.dionne@fdsonics.com) and Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA)

Efficient separation technologies for multi-component liquid streams that eliminate waste and reduce energy consumption are needed. In previous experiments around this novel platform technology, a single element transducer has been used to generate a high intensity three-dimensional ultrasonic standing wave resulting in an acoustic radiation force that is larger than the combined effects of fluid drag and buoyancy. Acoustic trapping of particles followed by enhanced gravitational settling is used to separate the secondary phase. A typical transducer is made of a PZT-8 2-MHz ceramic. This work reports on the comparison of the performance of a single element transducer to that of a multi-element transducer. Parametric simulation studies of multi-element transducer designs were performed to accurately predict the acoustic pressure field in the fluid flow with the goal of generating large acoustic radiation forces to assist in phase separation. COMSOL Multiphysics™ was used to run simulations and results were compared to an experimental prototype that consisted of a 2-in. by 1-in. flow chamber driven by a 1-in. by 1-in. 2-MHz transducer. The designs of the multi-element transducers consisted of two PZT-8 2-MHz transducers; one consisting of 16 elements and another of 25 elements. [Work supported by NSF PFI:BIC 1237723.]

9:00

4aPA3. Yeast filtration using large volume flow rate acoustophoretic separation. Brian McCarthy, Ben Ross-Johnsrud (FloDesign Sonics, 380 Main St., Wilbraham, MA 01095, b.mccarthy@fdsonics.com), and Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA)

Cell processing occurs in many technologies such as lab-on-a-chip, biopharmaceutical manufacturing, and food and beverage industry. Centrifuges and filters are used in preprocessing and filtration stages. These technologies are not continuous flow filtration methods, a drawback for automation and miniaturization. Continuous cell filtration using ultrasonic standing waves has been successfully used at limited flow rates [Hawkes and Coakley, Enzyme Microbial Technol. **19**, 57–62 (1996)]. Advantages of ultrasonic particle filtration are continuous operation with no mechanical moving parts, no risk of membrane fouling, and no consumables. We present a novel design of an acoustophoretic particle separation system operating at large volume flow rates. The technology operates by creating ultrasonic standing waves that produce an acoustic radiation force on particles which exceeds the drag and gravitational forces thereby trapping the particles. Over time aggregation of trapped particles results in gravitational settling of the agglomerated particles. The system comprises a 1 in. $\times$ 1 in. flow section and is powered by a 2 MHz PZT-8 transducer and typically operates at flow rates up to 2 L/H. Concentration reductions in excess of 90% are obtained for yeast suspensions of rehydrated *S. cerevisiae* in RO-DI water with volume concentrations ranging from 0.5 to 3%. [Work supported by NSF PFI:BIC 1237723.]

9:15

4aPA4. The role of bubbles in the atomization of liquids and tissues. Julianna C. Simon (Appl. Phys. Lab., Ctr. for Industrial and Med. Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jcsimon@uw.edu), Oleg A. Sapozhnikov, Vera A. Khokhlova (Appl. Phys. Lab., Ctr. for Industrial and Med. Ultrasound, Univ. of Washington, Seattle, WA, and Dept. of Acoust., Phys. Faculty, Moscow State Univ., Seattle, WA and Moscow, Russian Federation), Yak-Nam Wang, Wayne Kreider, Lawrence A. Crum, and Michael R. Bailey (Appl. Phys. Lab., Ctr. for Industrial and Med. Ultrasound, Univ. of Washington, Seattle, WA)

Ultrasonic atomization, or the emission of droplets from a liquid exposed to air, has been studied for many decades. The most accepted theory of atomization, the cavitation-wave hypothesis, states that droplets are emitted by a combination of capillary wave instabilities and cavitation bubble collapses. Recently, it was shown that tissues could also be atomized and that the result of atomization was surface erosion. Using a high static pressure chamber, we investigated the role of bubbles in the atomization of tissues and liquids. A 2-MHz, aluminum-lensed transducer was focused at the surface of either water or *ex vivo* bovine liver. In water at 1200 W/cm^2 ($p_+ = 6.8$ MPa, $p_- = 5.3$ MPa), we found that atomization ceased at an overpressure of 6.9 MPa, yet droplets were again released when the static pressure was increased to 13.8 MPa. In tissue at a linear *in situ* intensity of

22 000 W/cm^2 ($p_+ = 67$ MPa, $p_- = 16$ MPa), we found that a small increase in the static pressure (1.4 MPa) produced a qualitative change in atomization and caused thermal denaturation of the fractionated tissue rather than ejection from the surface. [Work supported by NIH EB007643, NIH DK043881, and NSBRI through NASA NCC 9-58.]

9:30

4aPA5. Optical theorem for beams and application to radiation forces and torques by Bessel beams. Likun Zhang (Ctr. for Nonlinear Dynam.., Univ. of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199, lzhang@chaos.utexas.edu) and Philip L. Marston (Dept. of Phys. and Astronomy, Washington State Univ., Pullman, WA)

The optical theorem is known as one of the central theorems in scattering theory; the theorem for plane waves relates the extinction by an object to the scattering amplitude at the forward direction. For a general non-diffracting beam an extended theorem was given recently [Zhang and Marston, J. Acoust. Soc. Am. **131**, EL329–EL335 (2012)]. The theorem relates the extinction to the scattering amplitude at the forward direction of plane wave components of the invariant beam. The theorem was used to examine the extinction by a sphere centered on the axis of a non-diffracting Bessel beam [Zhang and Marston, Bio. Opt. Express **4**(9), 1610–1617 (2013)]. The results are applied to recover axial radiation force [Zhang and Marston, Phys. Rev. E 84, 035601(R) (2011)] and torque [Zhang and Marston, Phys. Rev. E **84**, 065601(R) (2011)] exerted by the Bessel beam on the sphere. This form of optical theorem may be extended to a broader class of incident wave fields. [Zhang was supported by the 2013-14 ASA F. V. Hunt Postdoctoral Research Fellowship. Marston was supported by ONR.]

9:45–10:15 Break

10:15

4aPA6. A sonic levitation system for the study of Faraday waves on bubbles. Jorge Escobedo and R. Glynn Holt (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, jorge189@bu.edu)

Acoustic levitation is a method by which the gravitational force on a sample can be balanced by the time-averaged acoustic radiation force in a standing wave. Levitation has in the past been utilized as an ideal system to study phenomena at fluid interfaces, since boundary influences are small. Previous work on Faraday waves on bubbles has been carried out at ultrasonic frequencies, where the disadvantages are the need for fast diagnostics at small spatial scales. In this talk we describe efforts to develop a sonic frequency levitation system, employing hardware elements from previous NASA investigations in drop physics. Proof-of-concept styrofoam levitation in air will be demonstrated, and injection and deployment schemes for large (1-in. diameter) bubbles will be discussed. [Work supported by the Robert W. Young Undergraduate Award of the ASA.]

10:30

4aPA7. Experimental investigation of acoustic streaming flows inside a standing wave tube. Yasser Rafat, Shahin Amiri, Rani Taher, and Luc Mongeau (Mech. Eng., McGill Univ., 845 Sherbrooke St. West,, Montreal,, QC H3A 0G4, Canada, yasser.rafat@mail.mcgill.ca)

Acoustic streaming is identified as one of the several phenomena which affect the efficiency of thermoacoustic system by causing convective heat transfer. In the context of thermoacoustic machines, most of the experimental and numerical studies were performed on Rayleigh acoustic streaming. In the present study, different acoustic streaming flows within a standing wave tube were investigated. Experiments were performed using particle image velocimetry. A rectangular Plexiglas resonator was used as an idealized standing wave thermoacoustic refrigerator. The experimental results were compared with linear acoustic theory to ascertain their validity. Acoustic streaming generated due to interaction of standing wave with thermoacoustic core was also studied. Simplified components were used to model the stack and heat exchangers. It was found that presence of rigid obstacles in the standing wave resonator changed the streaming flow completely. Both the magnitude and shape of the streaming cells changed when compared with the classical Rayleigh streaming cell. The resulting local streaming velocity due to rigid obstacles in the standing wave tube had very high magnitude when compared with streaming in an empty standing wave tube.

10:45

4aPA8. Acoustic streaming from a resonant elastic surface vibration. Megha Sunny (Carnegie Mellon Univ., Lowell, Massachusetts), Katherine Aho, John C. Miniter, and Charles Thompson (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, katherine_aho@student.uml.edu)

Acoustic streaming induced by the vibration of an elastic membrane is examined. Adjustment of the impedance along the membrane allows one to control the spatial characteristics of the time-averaged fluid motion. It is shown that under resonant conditions spatial localization of the time-averaged Reynolds stress occurs. As a result, the fluid motion outside of the viscous boundary layer is driven into motion by a wall-jet at locations of maximal vibration. The velocity field is evaluated in terms of elemental Stokeslets where streaming motion is expressed in terms of a surface force distribution. At high oscillatory Reynolds numbers, regularization procedures to deal with singularities that occur in the formulation are addressed. Streaming motion is expressed in terms of the surface force distribution. Results for oscillatory and time-averaged fluid motion are presented.

11:00

4aPA9. Interaction acoustic radiation force and torque on a cluster of spheres suspended in an inviscid fluid. Jose Henrique A. Andrade (Physical Acoust. Group, Inst. of Phys., Federal Univ. of Alagoas, conjunto pau darco rua d 23, maceio 57043394, Brazil, henriquealopes@gmail.com), Mahdi Azarpeyvand (Mech. Eng., Univ. of Bristol, Bristol, United Kingdom), and Glauber T. Silva (Physical Acoust. Group, Inst. of Phys., Federal Univ. of Alagoas, Maceió, AL, Brazil)

The acoustic radiation force and torque exerted by a time-harmonic beam of arbitrary wavefront on a cluster of suspended spheres in an inviscid fluid is theoretically analyzed. In the proposed method, the effective incident wave is modelled as a coherent sum of an external beam and the contributions from the re-scattering events by other spheres present in the medium. Using the translational addition theorem for spherical functions the effective beam-shape and scattering coefficients are numerically computed [J. Acoust. Soc. Am. **98**, 495 (1995)] for different external incident fields. The radiation force and torque exerted on the probe sphere can then be calculated using the farfield partial-wave expansion method [J. Acoust. Soc. Am. **130**, 3541 (2011); Europhys. Phys. Lett. **97**, 54003 (2012)]. The method was employed to obtain the radiation force due to an external plane and spherical waves on a cluster of three solid elastic or fluid spheres suspended in water. The results show that the radiation force deviates considerably from that exerted solely by the external incident wave and that the radiation torque arises on the spheres when a asymmetric spatial distribution of the effective incident acoustic field takes place in the medium. In addition, the proposed method may help on the study of acoustic tweezers devices and acoustofluidic systems, which involve several suspended particles.

11:15

4aPA10. Numerical study of Rayleigh-type acoustic streaming based on a three-dimensional incompressible flow model. Takeru Yano (Osaka Univ., 2-1, Yamada-oka, Suita 565-0871, Japan, yano@mech.eng.osaka-u.ac.jp)

Rayleigh-type acoustic streaming induced by a plane standing wave in a rectangular parallelepiped is numerically studied on the assumption that the streaming motion is an incompressible flow and induced by the so-called limiting velocity on the outer edge of the acoustic boundary layer on the wall of the rectangular parallelepiped. Solving the three-dimensional incompressible flow equations with a standard finite difference scheme, we show that the streamline indicates chaotic behaviors even when the streaming velocity field converges to a time-independent state (steady flow) for moderate Reynolds numbers. Based on the result, we can discuss an efficiency of mixing by the time-independent Rayleigh-type acoustic streaming motions in three-dimensional boxes.

Session 4aPP

Psychological and Physiological Acoustics and Speech Communication: Cambridge Contributions to Auditory Science: The Moore—Patterson Legacy

Andrew J. Oxenham, Cochair
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Robert P. Carlyon, Cochair
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Christopher Plack, Cochair
School of Psychological Sciences, Univ. of Manchester, Ellen Wilkinson Bldg., Oxford Rd., Manchester M13 9PL, United Kingdom

Chair's Introduction—8:00

Invited Papers

8:05

4aPP1. Psychophysics to the rescue! Translational hearing research in Cambridge. Robert P. Carlyon (MRC Cognition & Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB1 3DA, United Kingdom, bob.carlyon@mrc-cbu.cam.ac.uk)

Roy Patterson and Brian Moore, perhaps more than any other psychoacousticians, have succeeded in applying their research for the common good. Those who have benefited from this translation of basic research include users of hearing aids and of auditory warnings. I will describe the results of recent experiments aimed at improving hearing by another group, namely users of cochlear and auditory brainstem implants. These include attempts to exploit the polarity sensitivity of the electrically stimulated auditory system in order to extend the ranges of pitch that can be conveyed by each type of implant.

8:25

4aPP2. The auditory image model and me. Michael Akeroyd (MRC/CSO Inst. of Hearing Res. - Scottish Section, New Lister Bldg., Glasgow Royal Infirmary, Glasgow, Strathclyde G31 2ER, United Kingdom, maa@ihr.gla.ac.uk)

As a Ph.D. student in Roy Patterson's group at the MRC Applied Psychology Unit in Cambridge, UK, in the early 1990s, I was introduced to computer models of hearing. Much of my Ph.D. was devoted to exploring Roy's Auditory Image Model, a time-domain model for representing regularities in hearing sensations that we hear. It was built on a gammatone filterbank, a hair-cell model, and strobed temporal integration, and was programmed with a speed that was remarkable for the age. The pictures and movies that it made—and the insights into hearing that it gave—were exciting and inspiring; my resulting enthusiasm for what good models can do has remained with me throughout my scientific career. This talk will describe some of Roy's contributions to modelling and his influence on the field, as ever-improving computational models are crucial to making progress in understanding how hearing works. [Work supported by the Medical Research Council and the Chief Scientist Office, Scotland.]

8:45

4aPP3. Acoustic surface structure, across-formant integration, and speech intelligibility. Brian Roberts, Robert J. Summers (Psych., School of Life & Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk), and Peter J. Bailey (Dept. of Psych., Univ. of York, York, United Kingdom)

An important aspect of speech perception is the ability to group or select formants using cues in the acoustic surface structure—for example, fundamental frequency (F0) differences between formants promote their segregation. This study explored the role of more radical surface-structure differences. Three-formant (F1 + F2 + F3) synthetic speech analogues were derived from natural sentences. In one experiment, F1 + F3 were generated using second-order resonators (R1 + R3) and a monotonous glottal source (F0 = 140 Hz); in the other, F1 + F3 were tonal analogues (T1 + T3). F2 could take either form (R2 or T2). In some conditions, the target formants were

presented alone, either monaurally or dichotically (left ear = F1 + F3; right ear = F2). In others, they were accompanied by a competitor for F2 (F1 + F2C + F3; F2), which listeners must reject to optimize recognition. Competitors (R2C or T2C) were created using the time-reversed frequency and amplitude contours of F2. In the absence of F2C, the effect of surface-structure mismatch between F1 + F3 and F2 was typically modest. When F2C was present, intelligibility was lowest where F2 was tonal and F2C was a buzz-excited resonance, irrespective of which type matched F1 + F3. This finding suggests that surface structure type, rather than similarity, governs the phonetic contribution of a formant. [Work supported by ESRC.]

9:05

4aPP4. Enhancement of forward suppression begins in the ventral cochlear nucleus. Ian M. Winter (Cambridge Univ., The Physiological Lab., Downing St., Cambridge CB2 3EG, United Kingdom, imw1001@cam.ac.uk), Naoya Itatani (Univ. of Oldenburg, Oldenburg, Germany), Stefan Bleeck (Univ. of Southampton, Southampton, United Kingdom), and Neil Ingham (Kings College, London, United Kingdom)

A neuron's response to a sound can be suppressed by the presentation of a preceding sound (aka forward masking/forward suppression). Early studies in the auditory nerve have suggested that the amount of forward suppression was insufficient to account for behavioral data. Modeling studies have, however, suggested that forward suppression could be enhanced by coincidence detection mechanisms in the brainstem. Using a two-interval forced-choice threshold tracking algorithm, we compared forward suppression for different neuronal populations in the ventral cochlear nucleus (VCN) and the inferior colliculus of anesthetized guinea pigs. In both nuclei, onset-type neurons showed the greatest amounts of suppression (16.9–33.5 dB) and, in the VCN, these recovered with a faster time constant (14.1–19.9 ms). Neurons with sustained discharge demonstrated reduced suppression (8.9–12.1 dB) and recovery time constants of 27.2–55.6 ms. The growth of suppression, with increasing suppressor level, was compressive, but this compression was reduced in onset-type units. The threshold elevations recorded for most unit types were insufficient to account for the magnitude of forward masking as measured behaviorally; however, some units classified as onset responders demonstrated a wide dynamic range of masking, similar to that observed in human psychophysics.

9:25

4aPP5. Linear and log frequency rippled spectra. William Yost, Xuan Zhong, and Anbar Najam (ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Roy Patterson's Ph.D. dissertation investigated the pitch-shift of the residue and pitch perception has been a significant topic of interest to Roy ever since. I had the good fortune of collaborating with Roy on several studies involving pitch perception, especially on what we called regular interval stimuli (RIS), most notably iterated rippled noise (IRN). In addition to IRN stimuli which are characterized as stimuli with regularly spaced spectral peaks and valleys on a linear frequency axis, many studies have investigated stimuli that have regularly spaced spectral peaks and valleys on a logarithmic axis. Both set of stimuli produce a timbre/pitch-like sound quality. In some cases the sound quality of the two types of stimuli are difficult to perceptually separate. While RIS models of pitch processing (e.g., autocorrelation-based models) can account for many of the IRN pitch data, it is not clear what mechanisms produce the timbre/pitch-like qualities of log-frequency, rippled-spectra stimuli. The current paper involves three experiments designed to better understand auditory processing of rippled-spectra stimuli in order to determine if there may be some common perceptual elements that underlie the perception of such stimuli. [Research supported by the AFOSR.]

9:45–10:00 Break

10:00

4aPP6. Frequency selectivity and the auditory filter. Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Roy Patterson and Brian Moore laid the foundations for our modern understanding of frequency selectivity in the human auditory system and have defined and later refined the measurement and the modeling of the auditory filter. Although the auditory filter is a theoretical construct, the frequency selectivity it represents is thought to reflect the filtering in the cochlea. This talk will review recent work on comparing behavioral measures of the auditory filter with physiological measures of cochlear tuning in humans and other species. Although there are clear pitfalls in using linear systems analysis to characterize an inherently nonlinear system, such as the cochlea, the results suggest that the framework established by Patterson, Moore and their colleagues provides robust estimates of frequency selectivity that are consistent with more direct physiological measurements of cochlear tuning. [Work supported by NIH grant R01DC012262.]

10:20

4aPP7. The relationship between speaker size perception and the auditory filter. Toshio Irino (Faculty of Systems Eng., Wakayama Univ., 930 Sakaedani, Wakayama 640-8510, Japan, irino@sys.wakayama-u.ac.jp) and Roy D. Patterson (Ctr. for the Neural Basis of Hearing, Dept. of Physiol., Development and Neurosci., Univ. of Cambridge, Cambridge, United Kingdom)

When we hear a new voice on the radio, we can tell whether the speaker is an adult or a child. We can also extract the message of the communication without being confused by the size information. This shows that auditory signal processing is scale invariant, automatically segregating information about vocal tract shape from information about vocal tract length. Patterson and colleagues have performed a series of experiments to measure the characteristics of size/shape perception [e.g., Smith *et al.*, J. Acoust. Soc. Am. **117**(1), 305–318 (2005)], and provided a mathematical basis for auditory scale invariance in the form of the stabilized wavelet-Mellin transform (SWMT) [Irino and Patterson, Speech Commun. **36**(3–4), 181–203 (2002)]. The mathematics of the SWMT dictates the optimal form of the auditory filter, insofar as it must satisfy minimal uncertainty in a time-scale representation [Irino and Patterson, J. Acoust. Soc. Am. **101**(1), 412–419 (1997)]. The resulting gammachirp auditory filter is an asymmetric extension of the earlier gammatone auditory filter—one which can explain the level dependence of notched-noise masking. Thus, although it is not immediately intuitive, speaker size perception and auditory filter shape are both aspects of a larger, unified framework for auditory signal processing.

10:40

4aPP8. Novel paradigms to investigate temporal fine structure processing. Christian Lorenzi (Dept d'etudes cognitives, Ecole normale superieure, 29 rue d'Ulm, Paris 75005, France, lorenzi@ens.fr)

A wide range of evidence has been presented to support the idea that aging and cochlear hearing loss impair the neural processing of temporal fine structure (TFS) cues while sparing the processing of temporal-envelope (E) cues. However, the poorer-than-normal scores measured in tasks assessing directly TFS-processing capacities may partly result from reduced "processing efficiency." The accuracy of neural phase locking to TFS cues may be normal, but the central auditory system may be less efficient in extracting the TFS information. This raises the need to design psychophysical tasks assessing TFS-processing capacities while controlling for or limiting the potential contribution of reduced processing efficiency. Several paradigms will be reviewed. These paradigms attempt to either: (i) cancel out the effect of efficiency (leaving only the temporal factor), (ii) assess TFS-processing capacities indirectly via E-perception tasks where efficiency is assumed to be normal for elderly or hearing-impaired listeners, or (iii) assess TFS-processing capacities indirectly via E-perception tasks designed such that impaired listeners (i.e., elderly or hearing-impaired listeners) should outperform control listeners (i.e., young normal-hearing listeners) if aging or cochlear damage cause a genuine suprathreshold deficit in TFS encoding. Good candidates in this regard are interference tasks. Pilot data will be presented and discussed.

11:00

4aPP9. The temporal coding of pitch: Insights from human electrophysiology. Christopher Plack (School of Psychol. Sci., Univ. of Manchester, Ellen Wilkinson Bldg., Oxford Rd., Manchester M13 9PL, United Kingdom, chris.plack@manchester.ac.uk)

Brian Moore and Roy Patterson have made seminal contributions to our understanding of pitch perception, and in particular the use of temporal pitch information by the auditory brain. The pitch of sounds may be encoded, at least in part, by the tendency of neurons to phase lock to the temporal fine structure and/or envelope of basilar membrane vibration. Direct physiological measures of this mechanism are difficult in humans. However, the gross activity of neurons in the brainstem can be measured using electrophysiological techniques. The frequency-following response (FFR) is an electrophysiological measure of phase locking in the rostral brainstem. The FFR differs between musicians and non-musicians, and across linguistic groups, and is sensitive to short-term pitch discrimination training. These findings suggest that the FFR may reflect neural activity relevant to the encoding of pitch, although other results suggest that it may reflect basic peripheral encoding, rather than the output of a pitch extraction process. Recent results from our laboratory show that combining behavioral and FFR measures can provide insights into the coding of the pitch of pure tones and the coding of musical consonance. The FFR may be a blunt tool, but it provides information that cannot be obtained using other techniques, and this may be particularly useful in investigations of the effects of age and hearing loss on the neural coding of pitch.

11:20

4aPP10. Brain imaging the activity associated with pitch intervals in a melody. Roy D. Patterson (Physiol., Development and Neurosci., Univ. of Cambridge, Downing Site, Cambridge CB2 3EG, United Kingdom, rdp1@cam.ac.uk), Stefan Uppenkamp (Medizinische Physik, Universität Oldenburg, Oldenburg, Germany), Martin Andermann, and André Rupp (Sektion Biomagnetismus, Universität Heidelberg, Heidelberg, Germany)

Early attempts to locate brain regions involved in pitch processing employed sequences of notes with no pitch, fixed pitch, and melodic pitch. They revealed a region of Heschl's gyrus lateral to primary auditory cortex where sequences with pitch produced more activity than noise, and regions where melody produced more activation than fixed pitch (in planum polare and the superior temporal sulcus). Subsequent research has focused on the fixed pitch region in Heschl's gyrus and the degree to which the activity is pitch specific. Recently, MEG techniques have been developed to compare the responses to sequences of notes as they occur within bars of music, and to separate current sources associated with attention to melody. This paper illustrates how the techniques can be used to investigate the hierarchy of pitch and melody processing as it occurs in four bar phrases with brass instruments. The experiments show that a given note elicits a larger response when it is part of a melody and the increment is associated with a source beyond auditory cortex. The paper shows how we might track the responses to orchestral instrument sounds presented in a musical context as they proceed through auditory cortex and beyond the temporal lobes.

11:40

4aPP11. Loudness summation across ears for hearing-impaired listeners. Brian C. Moore and Brian R. Glasberg (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

The summation of loudness across ears is often studied indirectly by measuring the level difference required for equal loudness (LDEL) of monaurally and diotically presented sounds. Typically, the LDEL is 5–6 dB, consistent with the idea that a diotic sound is about 1.5 times as loud as the same sound presented monaurally at the same level, as predicted by the loudness model of Moore and Glasberg [J. Acoust. Soc. Am. **121**, 1604–1612 (2007)]. One might expect that the LDEL would be smaller than 5–6 dB for hearing-impaired listeners, because loudness recruitment leads to a greater change of loudness for a given change in level. However, previous data from several laboratories showed similar LDEL values for normal- and hearing-impaired listeners. Here, the LDEL was measured for normal-hearing and hearing-impaired listeners using narrowband and broadband noises centered on a frequency where the latter had near-normal audiometric thresholds (500 Hz) and at a frequency where audiometric thresholds were elevated (3000 or 4000 Hz). The LDEL was similar for the two center frequencies for the normal-hearing listeners, but was smaller at the higher center frequency for the hearing-impaired listeners. The results were predicted reasonably well by the loudness model of Moore and Glasberg.

Session 4aSA

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

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

Chair's Introduction—8:55

Invited Papers

9:00

4aSA1. Simple models for linear and nonlinear modal vibration of circular cylindrical shells. Jerry H. Ginsberg (School of Mech. Eng., Georgia Inst. of Technol., 5661 Woodsong Dr., Dunwoody, GA 30338-2854, j.h.ginsberg@comcast.net)

Beginning shortly after the Second World War, ONR sponsored projects at Columbia University, New York, that analyzed vibration and shock response of shell structures. This work necessitated ingenuity because it preceded the advent of numerical modeling tools. A central theme was energy methods. This paper will review two works of this type. Baron and Bleich [J. Appl. Mech. **21**, 178–184 (1954)] used the mechanical energies of the Flugge-Byrne-Lur'ye shell theory and a fundamental property of the modal energies to obtain accurate formulas for the three vibration branches and associated modal displacements of a simply supported cylindrical shell. This author [Ginsberg, J. Appl. Mech. **40**, 471–477 (1973)] used these modes to analyze finite amplitude effects at resonances. Examination of the order of magnitude of terms in the mechanical energies led to identification of nonlinear mode coupling. The resulting differential equations for the modal coordinates were solved by a singular perturbation technique. The outcome was a set of algebraic equations for the nonlinear frequency response, and disclosure of the conditions under which an azimuthally symmetric response destablizes in favor of an antisymmetric response as a consequence of nonlinear coupling.

9:20

4aSA2. Scattering by a cylinderical shell buried in elastic sediment. Angie Sarkissian, Saikat Dey, Brian H. Houston (Code 7130, Naval Res. Lab., Code 7132, 4555 Overlook Ave. S.W., Washington, DC 20375, angie.sarkissian@nrl.navy.mil), and Joseph A. Bucaro (Excet, Inc., Springfield, VA)

Scattering results are presented for the case of cylindrical steel targets buried in elastic sediment with sound incident from the air above. The STARS3D finite element program recently extended to layered, elastic sediments is used to compute the scattering and the resulting normal displacement at the interface since the specific focus here is detection by systems which rely on monitoring the acoustic displacements or displacement-related entities at the fluid-sediment interface. Results are compared for the scattered field produced by the cylinder buried in layered elastic sediment versus in fluid sediment and for the scattered field of a buried cylindrical shell versus a buried solid cylinder. [This work was supported by ONR.]

9:40

4aSA3. Acoustic radiation from fluid-loaded cylindrical shells—A review. Joe M. Cuschieri (Lockheed Martin MST, 100 East 17th St., Riviera Beach, FL 33404, joe@cuschieri.us)

The acoustic radiation from fluid-loaded cylindrical shells received significant attention in the past. Reviewing the literature, the number of papers published in this area is significant. The body of work covers thin walled small diameter shells applicable to the sound transmission in pipes, to large diameter shells with internal stiffeners and bulkheads. Also, considered is the influence of full and partial compliant coatings. A significant portion of work was based on analytical techniques useful to understand the phenomena and some of the critical parameters. More recent, with the availability of more capable computers and modeling codes, the focus has been on application of these computational tools to solving field problems. This paper reviews some of the past computational work and how some applications evolved from this work. However when presented with actual submerged cylindrical structures with complex internals, while modeling is useful, when dealing with the acoustic radiation from shell like structures at medium to high frequencies modeling tools still cannot handle the full extent of the problem and the prevalent approach still relies on implementation of good engineering practice.

10:00

4aSA4. Quantitative ray methods for scattering by tilted cylindrical shells. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu) and Scot F. Morse (Div. of Comput. Sci., Western Oregon Univ., Monmouth, OR)

Starting with a review of ray methods and phenomena associated with high frequency scattering by spheres and cylindrical shells in water viewed broadside, generalizations to tilted shells will be summarized. These extensions were found to be useful for meridional as well as helical ray backscattering enhancements associated with leaky (or supersonic) waves on shells [Morse and Marston, J. Acoust. Soc. Am. **112**, 1318–1326 (2002); Blonigen and Marston, J. Acoust. Soc. Am. **112**, 528–536 (2002)]. For such enhancements Fermat's principle is useful for identifying ray paths of interest. In the case of helical waves (and in the broadside special case), the scattering amplitude can be expressed in terms of a Fresnel patch area where the guided wave is excited on the shell. Fresnel patches also give insight into the relatively large magnitude of meridional ray contributions. The coupling coefficient is proportional to the radiation damping of the leaky wave on the shell and in some cases it is necessary to take into account the anisotropy of the phase velocity. Computational benchmarks include scattering into the meridional plane by tilted infinite cylinders. Related phenomena include enhancements from subsonic guided waves and applications to sonar imaging and time-frequency analysis. [Work supported by ONR.]

10:20–10:30 Break

10:30

4aSA5. Response of a cylindrical shell with finite length ring stiffeners. Andrew J. Hull (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, andrew.hull@navy.mil)

This talk derives an analytical model of a thin shell of infinite extent that contains periodically spaced ring stiffeners that have finite length. The governing model of the system is the Donnell shell formulation affixed to ring stiffeners that are modeled as translational springs in the radial direction. The shell is excited by an external load that is harmonic in time and space. An orthogonalization procedure is developed and the resulting system equations are an infinite set of algebraic equations containing a diagonal matrix that represents the shell dynamics and a sparse matrix that contains permutations of the Fourier coefficients of the Heaviside step function that represent the stiffener forces. This matrix equation is truncated and inverted and yields a solution of the shell displacements. An example problem is formulated and the effects of the stiffeners on the system dispersion curves are illustrated.

10:50

4aSA6. Prediction of a body's structural impedance and scattering properties using correlation of random noise. Sandrine T. Rakotonarivo (Mechanics and Acoust., IUT GMP Aix-En-Provence, Université de Aix-Marseille, Marseille, France, sandrine.rakotonarivo@univ-amu.fr), W. A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), and Earl G. Williams (Acoust., Naval Res. Lab., Washington, DC, DC)

The structural or surface impedance matrix (or equivalently the inverse of the structural Green's function) for an elastic body can be obtained by placing it in an encompassing and spatially random noise field and cross-correlating pressure and normal velocity measurements taken on its surface. The derived theory shows that the correlation method produces the exact analytic form of the structural impedance matrix. A numerical experiment is presented determining the structural impedance matrix of an infinite cylindrical shell excited by a spatially random noise field. These results are then used to compute the scattered field from a nearby point source, which is in agreement with known results. [Work supported by the Office of Naval Research.]

11:10

4aSA7. Acoustic directional response of distributed fiber optic sensor cables. Jeffrey E. Boisvert (NAVSEA Div. Newport, 1176 Howell St., Newport, RI 02841, jeffrey.boisvert@navy.mil)

Distributed fiber sensor systems based on Rayleigh backscatter interferometry have demonstrated the capability for highly sensitive strain measurements over tens of kilometers using low cost fiber-optic cable for terrestrial and maritime applications. These cables are typically multilayered in construction and contain a fiber optic (glass) core. A generalized multi-layered infinite-length cable is modeled using the exact theory of three-dimensional elasticity in cylindrical coordinates. The cable is excited by an acoustic plane wave with an arbitrary angle of incidence. At each angle of incidence the radial and axial strains within the cable are integrated over a desired sensor zone length to determine the optical phase sensitivity using an equation that relates the strain distribution in an optical fiber to changes in the phase of an optical signal. Results for the cable in a free-field water environment are presented for two different cable geometries. The analytical model was used to identify the root cause of a marked increase in cable sensitivity that exists at certain angles of incidence for plane wave excitation. Specifically, the enhancement occurs when the trace wavelength of the incident wave matches the propagation wavelength of a natural frequency of the cable. [Work supported by NAVSEA Division Newport ILIR Program.]

Contributed Papers

11:30

4aSA8. Real-time hybrid substructuring of a physical mass-spring system coupled to a fluid-loaded analytical substructure. Rui Botelho and Richard E. Christenson (Civil and Environ. Eng., Univ. of Connecticut, 261 Glenbrook Rd. Unit 3037, Storrs, CT 06269, rui.botelho@uconn.edu)

Real-time hybrid substructuring (RTHS) is a relatively new method of vibration testing that allows a coupled dynamic system to be partitioned into separate physical and numerical components or substructures. The physical and numerical substructures are interfaced together in real-time as a closed-loop hybrid experiment similar to hardware-in-the-loop (HWIL) testing, whereby the physical substructure is tested concurrently with a numerical simulation of the remaining system. This work describes uniaxial RTHS testing at the University of Connecticut Structures Research Laboratory applied to simplified fluid-loaded structural systems. These tests use a physical one degree of freedom (DOF) mass-spring system coupled to a fluid-loaded analytical substructure. One test uses a fluid-loaded plate as the analytical substructure, while another test uses a fluid-loaded cylinder. An overview of RTHS is also presented, including the details of the feedback control architecture for coupling physical and analytical substructures together using servo-hydraulic actuation with a model-based minimum-phase inverse compensation (MPIC) of the actuator dynamics. In addition, a convolution integral (CI) method for solving the fluid-loaded analytical substructures in real-time is described. Experimental results demonstrate that RTHS can accurately capture the dynamic interaction of a fluid-loaded structural system and provide physical insight into the coupled response.

11:45

4aSA9. Experimental research and analysis of the acoustical radiation of piezoelectric cylindrical transducers with various height-to-diameter aspect ratios. Corey Bachand (BTech Acoust. LLC, 151 Martine St., ATMC, Fall River, MA 02723, corey.bachand@cox.net), David A. Brown (ECE/ATMC, Univ. of Massachusetts Dartmouth, Fall, MA), and Boris Aronov (BTech Acoust. LLC, Fall River, MA)

Estimating the radiation characteristics of cylindrical transducers having moderate height-to-diameter aspect ratios (H/D $\approx$ 0.2–2) over a wide frequency range is subject to considerable error with closed-form analytical solutions. It is often the case that transducers for acoustic communication fall within this range of aspect ratios. Thus, most often numerical techniques are required to solve the acoustical radiation problem, particularly for cylinders where the surface configuration (end caps and curved walls) does not allow for separation of variables in the Helmholtz equation describing the acoustic pressure. Results of calculating radiation characteristics of finite-height cylinders based on a numerical technique developed by Kozyrev and Shenderov [Sov. Phys. Acoust. **23**(6), 230–236 (1980)] are presented. Several prototype piezoelectric cylindrical transducers with aspect ratios ranging from 0.3 to 1.0 were constructed as part of the research on the radiation characteristics of finite-height cylinders. The two cases of an air-backed internal cavity with shielded end caps and of a fluid-filled internal cavity without end caps are considered. Analytical and numerical radiation estimations are compared to measured results with the prototypes, and applicability of the analytical models for different aspect ratios and wave dimensions are discussed.

THURSDAY MORNING, 8 MAY 2014 — EAST PREFUNCTION, 8:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Cross-Language Topics in Speech Communication (Poster Session)

Megan Reilly, Chair
Dept. of Cognitive, Linguist., and Psychological Sci., Brown Univ., 190 Thayer St., Providence, RI 02912

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10.00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

4aSC1. Tone as a primary perceptual cue in cross-linguistic speech perception: A comparison of Cantonese and Mandarin second-language speech of English clusters. Yizhou Lan (Dept. of Chinese, Translation and Linguist, City Univ. of Hong Kong, Tat Chee Ave., Kowloon, Hong Kong, eejoe.lan@gmail.com)

The study examined the patterns of production and perception of L1 Cantonese and Mandarin speakers for English consonant-/l/ clusters in five consonant conditions (/ph, th, kh, f, s/) and three vowel conditions (/i, a, u/). Five Cantonese, 5 Mandarin, and 5 native English speakers were assigned to read aloud words in C[l]V and C[l]VC structure in carrier sentences. Results showed that Cantonese speakers' /l/ was often reduced, indicated by a shorter average duration of the CV transition compared with native English speakers. However, Mandarin speakers showed a longer duration in the same measurement. Despite the identical segmental and syllable structures of these two languages in the involved words, realizations were different. Nevertheless, we found pitch patterns of Mandarin speech of L2 English featured falling tone, whereas Cantonese speakers utilized level tones. To further examine the tonal effect, we normalized the tone from the production results to a level tone at 200 Hz and presented them, together with productions with real inserted vowels in between C and /l/, to another group of Mandarin and Cantonese speakers to discriminate in an ABX paradigm. Mandarin speakers scored a significantly lower accuracy rate, indicating that their perception of duration was influenced by tone structure.

4aSC2. Contrastive apical post-alveolar and laminal alveolar click types in Ekoka !Xung. Amanda L. Miller (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210-1298, amiller@ling.osu.edu) and Jeffrey J. Holliday (Second Lang. Studies, Indiana Univ., Bloomington, IN)

Ekoka !Xung has four contrastive click types—dental, alveolar, lateral, and "retroflex." We provide acoustic and ultrasound results of five speakers' productions of the typical alveolar click and the contrastive "retroflex" click. Ultrasound results show that the "alveolar" click is apical post-alveolar and the "retroflex" click is laminal alveolar. The burst duration of the post-alveolar click averages 12 ms which is "abrupt," while the burst duration of the

alveolar click averages 30 ms, which is "noisy." Mixed effects logistic regression models tested the effects of rise time and burst duration. Burst duration differed significantly among the two clicks ($p < 0.001$), while the effect of rise time was not significant. The ratio of energy in the click noise-bursts below 20 ERB to the energy above 20 ERB is between 1.0 and 1.5 for the post-alveolar click, but between 0.5 and 1.0 for the alveolar click. The ratio was a significant predictor of click type ($p = 0.014$). The highest concentration of energy for the post-alveolar click is between 12 and 18 ERB, while the highest concentration of energy in the alveolar click is between 25 and 30 ERB. We attribute the frequency difference to a larger lingual cavity volume in the post-alveolar click, and a smaller volume in the alveolar click.

4aSC3. Cross language speech-in-noise perception by early Spanish-English bilinguals and English monolinguals. Page E. Piccinini and Marc Garellek (Linguist, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0108, ppiccinini@ucsd.edu)

Bilinguals have shown a hyper-awareness of fine phonetic detail in speech, while also sometimes losing out on higher-level syntactic and semantic information in speech-in-noise studies. This study seeks to determine how bilinguals process speech in noisy environments across different language contexts. Specifically, this study tests whether bilinguals utilize certain phonetic cues to access higher-level information. Two experiments will be conducted. First, to determine how bilinguals process speech in noise in different language contexts, early Spanish-English bilinguals and English monolinguals learning Spanish listened to sentences mixed with white noise in English, Spanish, and code-switching (English to Spanish and Spanish to English) contexts. Preliminary results suggest early Spanish-English bilinguals perform significantly above chance on word identification in all contexts, performing best in the Spanish context. The second experiment will determine specifically which noise types (lower versus higher frequency) are most detrimental to word identification. This in turn will suggest what kind of phonetic information is utilized most by bilinguals versus monolinguals. These results will aid our understanding of how bilinguals could use their hyper-awareness of phonetic detail to overcome difficulties in other aspects of processing.

4aSC4. Native language interference on the overnight consolidation of a learned nonnative contrast. Sayako Earle and Emily B. Myers (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 123 Davis Rd., Storrs, CT 06268, frances.earle@uconn.edu)

In a prior investigation, discrimination of a trained nonnative (dental/retroflex, Hindi) contrast was mediated by different effects of overnight consolidation depending on the time of day of training. For individuals trained in the evening, sleep appeared to promote continued improvement in discrimination for ~24 h without further training. For participants who were trained in the morning, performance returned to baseline following the overnight session interval. A possible explanation for the lack of improvement in the morning training group is that incidental exposure to the alveolar /d/, the category in which dental/retroflex are considered allophones in English, throughout the daytime interval interfered with overnight consolidation of the nonnative variants. We tested this interpretation directly, by training all participants ($n = 44$) in the evening and assigning them to one of two conditions of interference: passive exposure to a stream of either 1500 /bV/ or /dV/ tokens immediately after training. We observed continuous improvement in discrimination for ~24 h in those who were exposed to /bV/ tokens, while those who were exposed to the /dV/ tokens did not improve. Our results support the interpretation that incidental exposure to English prior to overnight memory consolidation interferes with sleep-mediated improvement in discrimination of an L2 contrast.

4aSC5. Vowel systems of quantity languages compared: Arabic dialects and other languages. Judith K. Rosenhouse (Linguist, SWANTECH Ltd., 89 Hagalil St., Haifa 3268412, Israel, swantech@013.net.il), Noam Amir, and Ofer Amir (Commun. Disord., Tel-Aviv Univ., Tel-Aviv, Israel)

The acoustic phonetic features of colloquial Arabic vowel systems are still not entirely researched. This paper studies phonetic structure of several Arabic dialects and other languages. A basic issue is the fact that Arabic is a quantity language; but from the published literature we see that vowel systems of Arabic dialects differ in many acoustic details. We researched two colloquial Arabic dialects which are spoken in Israel, and hitherto not acoustically studied. These dialects constitute the axis around which we conducted the literature-based comparison with vowel systems of a few other Arabic dialects and other languages which share similar quantity features (i.e., long and short vowels). The study reveals similarities and differences in pitch (F0), the first three formants and duration. These differences appear between the two Arabic dialects spoken in Israel, between them and other Arabic dialects, as well as between non-Arabic languages (English, German, Swedish, and Hungarian). The findings of our study are discussed in relation with the questions of (1) vowel spaces of short and long vowels and (2) speaker's sex-dependent differences.

4aSC6. The articulation of lexical palatalization in Scottish Gaelic. Jae-Hyun Sung (Linguist, Univ. of Arizona , P.O. Box 210025, Tucson, AZ 85721, jhsung@email.arizona.edu), Diana Archangeli (Linguist, Univ. of Hong Kong, Hong Kong, Hong Kong), Ian Clayton (English, Boise State Univ., Boise, ID), Daniel Brenner, Samuel Johnston, Michael Hammond, and Andrew Carnie (Linguist, Univ. of Arizona , Tucson, AZ)

Scottish Gaelic (Gàidhlig, henceforth SG) exhibits a rich system of consonant mutation, which is mostly governed by its morphology (Ladefoged *et al.* 1998; Gillies 2002; Stewart 2004). For instance, *bàta* "boat" changes to [v] when the word undergoes morphological inflection—e.g., *a bhàta* "his boat", in which the sound spelled *bh* is pronounced as [v]. Using ultrasound imaging, the present study investigates palatalization in SG, which is considered as one of lexicalized consonant mutation types. Experimental data was collected in Sabhal Mòr Ostaig, a college on the Isle of Skye. Preliminary results show a clear sign of palatalization across different consonant types in palatalization environments (i.e., when morphologically conditioned), represented by higher tongue contours in the front region of tongue. While the articulatory distinction between plain and palatalized consonants is significant, different syllabic positions (i.e., word-initial vs. -final palatalization) often yield individualized patterns.

4aSC7. An acoustic-phonetic account of phonotactic perceptual assimilation. Eleanor Chodroff, Anthony Arnette, Samhita Ilango, and Colin Wilson (Cognit. Sci., Johns Hopkins Univ., Krieger Hall 237, 3400 N. Charles St., Baltimore, MD 21218, chodroff@cogsci.jhu.edu)

Previous research has identified a coronal-to-dorsal 'perceptual assimilation' in which English and French listeners identify Hebrew word-initial /tl/ and /dl/ as beginning with /k/ and /g/, respectively (Hallé and Best, 2007). However, the acoustic-phonetic factors that contribute to this misperception have not been thoroughly identified, and previous results indicate that /tl/ is misperceived more often than /dl/—an asymmetry that is surprising on phonological grounds. The present study further explored this perceptual assimilation in two experiments with English listeners and Hebrew stop-liquid-vowel syllables ([t,k,d,g] × [l,ʁ] × [a,o,u]). The first experiment, which used the same stimuli as Hallé & Best, replicated previous findings, including the asymmetry between /tl/ and /dl/. The second experiment employed stimuli produced by a different native Hebrew speaker. While coronal-to-dorsal assimilation was observed, the previous /tl/-/dl/ asymmetry was not found: /dl/ was perceived as dorsal-initial somewhat more often than /tl/, suggesting that there can be no consistent phonemic or phonotactic explanation of the rate of assimilation. In support of a phonetic account, we find that misperception rates in both experiments are highly correlated ($r > 0.65$) with the stimulus-specific degree of anticipatory coarticulation of the lateral, as reflected in the spectral shape of the stop burst.

4aSC8. The effect of talkers' language dominance on subjects' speech production of sibilant fricatives. Ya-ting Shih (Teaching Chinese as a Second Lang., Chung Yuan Christian Univ. , 200 Chung Pei Rd., Chung Li 32023, Taiwan, ninashih1982@gmail.com)

This study investigates the effect of talkers' language dominance on subjects' sibilant production in a bilingual community. Guoyu (Taiwanese Mandarin) has 3 sibilants: alveolar /s/, retroflex /ʂ/ and alveolo-palatal /ɕ/, while Taiwanese (a Southern Min dialect) only has /s/, which is palatalized

before /i/. Previous studies have shown that Taiwanese-dominant speakers in Taiwan has a merged category of /s/, /ʂ/ and /ɕ/. In addition, they treat [s] and [ɕ] as allophones of /s/. On the other hand, Guoyu-dominant speakers have a more distinctive three-way contrast of sibilants. This study explores whether listening to talkers with different language dominances affects subjects' speech production of Guoyu sibilants. Two female talkers, one is Taiwanese-dominant and the other is Guoyu-dominant, recorded Guoyu words containing target sibilants in word-initial position with comparable vowels. Forty bilingual adults' productions were elicited in a repetition task blocked by talker. The spectral centroid is obtained from the middle 40ms of each sibilant, along with the onset F2 of the following vowel. The two acoustic measures were plotted against each other and separated by talker. Preliminary results show that these subjects' productions differ when prompted by different talkers. Additional statistical tests will be performed to explore this production difference.

4aSC9. Speech intelligibility across native and non-native accents: Accent similarity and electrophysiological measures of word recognition. Louise Stringer and Paul Iverson (Speech, Hearing and Phonetic Sci., UCL, Chandler House, 2 Wakefield St., London WC1N 1PF, United Kingdom, l.stringer.11@ucl.ac.uk)

The intelligibility of accented speech in noise greatly depends on the pairing of speaker and listener, where two important factors are a listener's familiarity with a speaker's accent and the acoustic similarity between their accents. In this study, we present patterns of the intelligibility of standard British English, Glaswegian English and Spanish English accents for British and high and low proficiency Spanish listeners. We predict intelligibility will correlate with acoustic-phonetic similarity across accent pairings, in line with previous findings. As such, findings are expected to provide further support that accent similarity can predict patterns of accent intelligibility, even if listeners have little experience of a speaker's accent. Electrophysiological measures relating to phonological and semantic integration stages of word recognition will also allow the investigation of the influence of accent on the time course of word recognition, which has not previously been directly compared to accent intelligibility or applied in studies of accent processing by non-native listeners. Findings will be discussed in the context of previous exploratory findings in this field, and in reference to other factors influencing accent intelligibility.

4aSC10. Non-native vowel processing as reflected by neuronal architecture: Dynamic causal modeling of the magnetic mismatch response. Georgina Oliver-Roth, Paul Iverson (Speech, Hearing and Phonetic Sci., Univ. College London, Wakefield St. 2, London WC1N 1PF, United Kingdom, g.oliver@ucl.ac.uk), Sundeep Teki, and Alexander Leff (Inst. of Cognit. Neurosci., Univ. College London, London, United Kingdom)

The aim of this study was to examine how auditory vowel processing by native (L1) and non-native (L2) speakers is reflected in their neuronal source architecture and in coupling between brain regions. We used the magnetic Mismatch Response/MMNm to test automatic brain responses to within- and between-category vowels with English controls and English L1/French L2 speakers with a varying range of L2 proficiency. Additionally, participants performed a range of behavioral tasks which targeted vowel perception (category discrimination and vowel identification) and production. The MEG data from this study was analyzed conventionally and with Dynamic Causal Modeling/DCM in order to determine neuronal sources and the dynamic source architecture in the L1 and L2 brain. In summary, the right hemisphere supported the left during L2 vowel processing in low ability L2 speakers. Performance in L2 vowel category discrimination was linked to the MMNm for a vowel distinction which was particularly difficult for the L2 speakers. The MMNm indicated whether a speech sound had gained phoneme status in an L2. DCM showed that there was no difference both architecturally and functionally between an L1 speaker's and a highly proficient L2 speaker's brain with regards to vowel processing.

4aSC11. A cross-linguistic study of lexical stress shifts in level 1 [+ cyclic] derivations. Paul R. Keyworth (English, Saint Cloud State Univ., 3710 W. Saint Germain St., Apt. #234, Saint Cloud, MN 56301-7319, kepa1104@stcloudstate.edu)

Laboratory phonology has been widely employed to understand the interactional relationship between the acoustic cues of English Lexical Stress (ELS)—duration, fundamental frequency, and intensity. However, research on ELS production in polysyllabic words is limited, and cross-linguistic research in this domain even more so. Hence, the impacts of second language (L2) experience and first language (L1) background on ELS acquisition have not been fully explored. This study of 100 adult Mandarin (Chinese), Arabic (Saudi Arabian), and English (Midwest American) speakers examines their ELS productions in tokens containing seven different stress-moving suffixes; i.e., Level 1 [+ cyclic] derivations according to lexical phonology. Speech samples were systematically analyzed using Praat and compared using statistical sampling. Native-speaker productions provided norm values for cross-reference to yield insights into the proposed Salience Hierarchy of the Acoustic Correlates of Stress (SHACS). The author recently reported the main findings which support the idea that SHACS exists in L1 sound schemes, and that native-like command of these systems can be acquired by L2 learners through increased L2 input. Other results are expected to reveal the role of tonic accent shift, the idiosyncrasies of individual suffixes, conflicts with standard dictionary pronunciations, and the effects of frequency perception scales on SHACS.

4aSC12. Effects of observing or producing hand gestures on non-native speakers' auditory learning of Japanese short and long vowels. Yukari Hirata (East Asian Lang. and Literatures, Colgate Univ., 13 Oak Dr., Hamilton, NY 13346, yhirata@colgate.edu), Spencer D. Kelly (Psych., Colgate Univ., Hamilton, NY), Jessica Huang, and Michael Manansala (East Asian Lang. and Literatures, Colgate Univ., Hamilton, NY)

This study examined whether auditory training coupled with hand gesture can improve non-native speakers' auditory learning of phonemic vowel length contrasts in Japanese. Hirata and Kelly (2010) found that observing hand gesture that moved along with the rhythm of spoken short and long vowels in Japanese did not uniquely contribute to non-native speakers' auditory learning. The present study compared effects of four types of training to examine whether there is a more effective method: (1) producing syllabic-rhythm gesture, (2) observing syllabic-rhythm gesture, (3) producing moraic-rhythm gesture, and (4) observing moraic-rhythm gesture. Each of native English speakers ($N = 88$) participated in one of the four types of training in four sessions, and took a pretest and a posttest that measured their ability to auditorily identify the vowel length of novel words without hand gesture. Tested disyllable pairs had the contrast in the first and the second syllables, spoken in sentences at slow and fast rates. Results showed that all four groups improved significantly (9%), but the amount of improvement did not differ. However, 'observing syllabic-rhythm gesture' was the only condition in which auditory learning was balanced between the first and the second syllable contexts and between the slow and fast rates.

4aSC13. Perceptual learning of lexical tones by native speakers of English. Guannan Shen, Erika Levy , and Karen Froud (Teachers College, Columbia Univ., 509 West 122nd St., Apt. 18, New York, NY 10027, mandy.g.shen@gmail.com)

Whether native speakers of non-tonal languages can acquire categorical representations of lexical tones remains controversial. This study investigates the acquisition of lexical tone categories by native English speakers learning Mandarin Chinese as a foreign language by comparing the categorical perception of lexical tones between three groups of listeners: (1) native English speakers who had taken advanced Mandarin courses in colleges; (2) inexperienced native English speakers; and (3) native Mandarin speakers. Two tone continua derived from natural speech within carrier phrases were

created through interpolation within two tone contrasts (T1/T4; T2/T3). Assessments of categorical perception, including an identification task and a discrimination task, were conducted on all three groups of participants. Results showed classic categorical perception of tones by native Mandarin speakers. The inexperienced English speakers performed near chance on discrimination tasks and showed significantly broader identification boundaries. The learners of Mandarin showed similar categorical perception to native Mandarin speakers with comparable identification boundaries and discrimination scores. The results indicate that native speakers of non-tonal languages can learn to perceive lexical tones categorically. Experience-based perceptual categorization and acoustic cues for tonal language learners are discussed.

4aSC14. The effect of language experience on the ability of non-native listeners to identify Japanese phonemic length contrasts. Miwako Hisagi (Speech Commun. Studies, Iona College, 715 North Ave., New Rochelle, NY 10801, mhisagi@hotmail.com), Keiichi Tajima (Psych., Hosei Univ., Tokyo, Japan), and Hiroaki Kato (Universal Commun. Res. Inst., National Inst. of Information and Communications Technol. (NICT), Kyoto, Japan)

This study investigated how language experience affects second-language (L2) listeners' ability to perceive phonemic length contrasts in the face of stimulus variability. Native English-speaking learners of Japanese ($N = 42$) participated in an identification task in which the stimuli were Japanese words contrasting in vowel or consonant length, produced in isolation or embedded in a carrier sentence at slow, normal, or fast speaking rates, presented in a random order. Participants also received an Oral Proficiency Interview (OPI), developed by the American Council on the Teaching of Foreign Languages (ACTFL), to assess their Japanese proficiency on a 10-level scale. Results showed that identification accuracy as measured by d' was weakly correlated with OPI level ($r = 0.298$), and moderately correlated with number of semesters enrolled in Japanese language courses ($r = 0.401$). Speaking rate significantly affected performance, but its effect differed by context. For word-in-isolation context, d' was highest at the normal rate, while for word-in-sentence context, d' was highest at the slow rate. However, the effect of speaking rate was not reduced as a function of OPI level or number of semesters in Japanese courses. Thus, language experience may not always strongly predict L2 speech perception. [Work supported by JSPS-KAKENHI, MIT-RLE and -Linguistics & Philosophy.]

4aSC15. Can adjustment to accented speech affect native language perception? Eva M. Lewandowski (Psych., Emory Univ., 36 Eagle Row, Dept. of Psych., Ste. 270, PAIS Bldg., Atlanta, GA 30306, elewand@emory.edu), Teljer Liburd (Psych. and Learning Res. and Development Ctr., Univ. of Pittsburgh, Pittsburgh, PA), and Lynne C. Nygaard (Psych., Emory Univ., Atlanta, GA)

The human auditory system can quickly accommodate foreign-accented speech. However, the cognitive mechanisms underlying perceptual adjustment to non-native speech are not fully understood. The current study examined the perceptual consequences of adaptation toward foreign-accented speech on native language perception. Native English speakers performed an auditory shadowing task on word-length utterances in English. There were four blocks of trials. The words in the critical block (Block 3) were spoken by either a native American English speaker or a native Spanish speaker. The speaker in flanking blocks (Blocks 1–2, 4) was the same speaker, a different speaker with the same accent, or a different speaker with a different accent. Shadowing response times in the critical block were used to assess rapid perceptual adjustment and readjustment. Results showed that the nature of the preceding context influenced response times. Response times for items in the first quartile of the critical block were reliably slower when accent and talker changed than when accent and talker remained constant. These findings suggest that listeners develop perceptual expectations about ongoing speech, which when violated incur a short-term processing cost even for spoken words in the listeners' native language.

4aSC16. Word recognition in early bilingual adults for two degradation types. Rachel Shepherd and Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, rachshep@imail.iu.edu)

Early bilingual adults have difficulty perceiving speech in noise compared to monolingual adults; however, the cause of the deficit is unknown. Further, the extent to which this deficit extends to other types of degradation, such as source degradation (e.g., a nonnative accent) has not been investigated. The current study investigated word recognition under environmental and source degradation by 24 monolingual and 24 bilingual listeners, who learned English and at least one other language before age 6. Participants identified sentences produced by one native and one nonnative talker in both quiet and noise-added conditions. Although noise was more detrimental to bilinguals than monolinguals, the presence of a nonnative accent caused a similar decline for both groups. Results from standardized tests of vocabulary, reading, spelling, nonverbal intelligence, and phonological processing showed two differences between the groups: bilinguals outperformed monolinguals on the nonverbal intelligence test (Raven's Standard Progressive Matrices) and bilinguals performed less accurately than monolinguals on the vocabulary assessment (Peabody Picture Vocabulary Test). Therefore, the speech-in-noise deficit for bilinguals may be traced to their weaker vocabulary knowledge. This study demonstrates that early bilinguals experience a word-recognition disadvantage under environmental degradation but not source degradation.

4aSC17. The production of non-modal phonation types in English vowels by Brazilian speakers. Ana Paula Engelbert (Head and Neck Surgery, UCLA, 1000 Veteran Ave., Los Angeles, CA 90095, anaengelbert@ucla.edu)

Esling (2000) claims that each language has its own pattern of physiological behavior in which articulators are trained to operate in different ways based on the language's phonetic structure. To test this claim, this study compares phonation types in speech production when Brazilians speak Portuguese and English. More specifically, we investigate coarticulation effects of consonants on vowels in English with regards of non-modal phonation. According to Garellek (2012), non-contrastive non-modal phonation happens in English vowels due to adjacent glottalized and aspirated consonants. However, this coarticulation effect does not happen in Brazilian Portuguese because voiceless stops have short lag VOT and neither voiced nor voiceless stops are allowed as codas. Thus, our hypothesis is that bilingual Brazilians do not produce non-modal phonation due to coarticulation when producing English vowels. To test this hypothesis, native speakers of English and Brazilians who are proficient speakers of English were recorded performing reading tasks in a soundproofed room. The target words containing the vowels to be measured were placed in a carrier-sentence. The acoustic analysis was based on H1*-H2*, H1*-A2*, and HNR measures. [Research funding by CAPES (Brazil) and Fulbright.]

4aSC18. Acquisition of the complex three-way Korean plosive contrast by native English speakers. Tyler Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, tkp@bu.edu), Amy S. Finn, Jennifer Minas (McGovern Inst. for Brain Res., Massachusetts Inst. of Technol., Cambridge, MA), Caitlin Tan (Dept. of Brain and Cognit. Sci., Massachusetts Inst. of Technol., Cambridge, MA), Brian Chan, and John D. Gabrieli (McGovern Inst. for Brain Res., Massachusetts Inst. of Technol., Cambridge, MA)

Learning to perceive foreign language speech sounds is a core challenge in adult second language acquisition. Previous research has considered how listeners learn novel foreign language categories for a known phonetic continuum [e.g., voice onset time (VOT)], or how listeners learn to use a previously unattended phonetic feature (e.g., F3). We investigated perceptual learning of the Korean three-way plosive contrast (lenis, aspirated, and fortis) by native English speakers. Unlike VOT continua in other languages,

this contrast is distinguished by complex trading relations between VOT and pitch, with place of articulation differences in VOT adding further complexity. In this study, participants (N = 38) learned a vocabulary of 18 Korean pseudowords comprised of six minimal triplets (e.g., pan, ban, and ppan) by undergoing four days of high-variability (multi-talker) training on a lexical identification task. Mixture model analysis suggested two learner groups: (1) two-thirds of the participants were partially successful at learning words beginning with the fortis stops, but did not differentiate the lenis and aspirated stops; and (2) one-third of the participants successfully learned words beginning with the fortis stops, and exhibited progress distinguishing the lenis and aspirated stops. (Fortis stops most closely resembled listeners' existing English voiced stop categories.) Both groups acquired these contrasts best for bilabial stops and least accurately for alveolar stops.

4aSC19. Prosodic realization of focus in American English by Beijing Mandarin learners. Ying Chen (Dept. of Linguist, Univ. of Oregon, 124 Agate Hall, 1290 University of Oregon, Eugene, OR 97402, ychen12@uoregon.edu)

In addition to an increase of duration, F0 and intensity in phonetically realizing focus, post-focus compression (PFC) of F0 and intensity has been found in many languages, including American English and Beijing Mandarin. Recent studies found that PFC did not easily transfer from language to language (Wu and Chung, 2011); however, language experience impacted the realization of PFC (Chen *et al*., 2012). The effect of length of residence (LOR) in an L2-speaking environment on L2 pronunciation accuracy remains controversial (Piske, 2007). The current study examined English focus production of Beijing Mandarin learners, who were college freshmen, residing in the United States for 3 to 6 months, and college seniors for 3.5 to 4 years. Compared to the control group, both learner groups produced comparable patterns of duration change; the freshman group did not present significant PFC of F0 and intensity in either initial-focus or medial-focus condition; the senior group presented a native-like PFC in the initial-focus condition and an intermediate pattern of PFC among the three groups in the medial-focus condition. The preliminary results indicate that Beijing Mandarin learners with long LOR in the US produced more native-like prosodic focus in English than those with short LOR in the United States.

4aSC20. Transfer effects in perception of a familiar and unfamiliar language. Charles B. Chang (Dept. of Linguist, Rice Univ., P.O. 1892, MS 23, Houston, TX 77251, cbchang@post.harvard.edu)

Second-language (L2) speech perception is typically worse than first-language (L1) perception, a disparity often attributed to negative transfer (interference) from the L1 of L2 listeners. The current study investigated the hypothesis that L1 transfer is not always negative, but variable depending on the nature of L1 perceptual biases. In Experiment 1, four groups of L2 English speakers whose L1s (Japanese, Korean, Mandarin, and Russian) differ in the relative informativeness of vowel-to-consonant transition cues were tested on their perception of English segments that rely crucially on these cues: final unreleased voiceless stops. In comparison to L1 English listeners, L1 Japanese, Russian, and Mandarin listeners performed significantly worse, whereas L1 Korean listeners performed significantly better. In Experiment 2, when the same groups were tested on similar Korean stimuli, L1 Russian listeners outperformed all other groups except the Korean group. These results provide evidence that L1 transfer effects are diverse and suggest that they are diverse for two reasons: variability in the information value of relevant phonetic cues in the L1, as well as variability in the degree to which linguistic expectations associated with a target L2 (or the lack thereof) predispose the listener to make effective use of these cues.

4aSC21. Steady as /ʃi/ goes: The spectral kinematics of sibilant fricatives in English and Japanese. Patrick F. Reidy and Mary E. Beckman (Dept. of Linguist, The Ohio State Univ., 24A Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, patrick.francis.reidy@gmail.com)

Sibilant fricatives are often treated as having steady-state articulatory targets, which fix their spectra throughout their duration; however, Iskarous *et al*. (2011) reported that the centroid frequency of English /s/ varies considerably across the fricative's time course. This study replicates their spectral analysis using a psychoacoustic measure (peak ERB) and then extends it to English /ʃ/ and Japanese /s, ɕ/. The time-varying spectral pattern of each fricative was approximated with a nine-point peak ERB trajectory, computed from 20-ms windows spaced evenly throughout each token. There were three notable results. First, adults did not produce the same spectral kinematic pattern for all sibilants in a given language: the spectral peak of English /s/ followed a concave trajectory, while /ʃ/ remained relatively flat. Second, phonetically similar fricatives from different languages did not necessarily show similar dynamical spectral patterns: the peak trajectory of /s/ was curved in both languages, but reached its maximum much earlier in Japanese. Finally, three- to five-year-old children exhibited a developmental path toward language- and consonant-specific spectral patterns: as age increased, English-acquiring children produced /ʃ/ with decreasing curvature to its spectral peak trajectory, approaching that produced by the adults.

4aSC22. Effects of experience on the processing of phonetic contrasts in foreign-accented Spanish. Fernando Llanos (School of Lang. & Cultures, Purdue Univ., West Lafayette, IN 47907, fllanos@purdue.edu) and Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN)

Non-native accented speech is typically less intelligible than unaccented speech. However, intelligibility improves with experience. Experience might improve intelligibility by guiding listeners' expectations regarding the systematic divergence of specific acoustic cues from the native norm within the non-native context. If an accent imposes predictable changes on the acoustic cue patterns present in speech, then listeners experienced with that accent may change their judgment of what was said based on whether or not it was perceived in an accented context. In the present study, two groups of native speakers of Spanish with and without significant experience with English-accented Spanish listened to Spanish sentences produced with and without a strong English accent. Each sentence ended in a Spanish word produced with or without English accent, but the voice onset time (VOT) of the first consonant in the word was artificially varied to form a continuum from bata (robe) to pata (paw). Experienced listeners showed a category boundary at a VOT of approximately 5 ms with no significant difference between accent conditions, suggesting that listeners were not affected by the perception of a familiar foreign accent. Additional results from inexperienced listeners and using non-word targets and fully English context sentences will be discussed.

4aSC23. Multiple sources of information contribute to novel category formation. Emily Cibelli (Linguist, UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, ecibelli@berkeley.edu)

The acquisition of novel phonemes in a new language often presents a challenge for learners, particularly when target categories overlap with or assimilate to native categories. In the current study, English speakers who have no experience with Hindi are asked to learn the Hindi dental-retroflex place contrast and the four-way stop voicing contrast. The multi-day study includes an AX discrimination task and a repetition task with (V)CV syllables. In experiment 1, a training procedure was designed to manipulate multiple sources of information available to the listener. Training sessions include performance feedback and adaptive fading (a progression in exposure from clear tokens to more peripheral exemplars). Critically, the study also includes explicit articulatory training of the target sounds, to test the hypothesis that information about the existence of a new articulatory target can support the development of a perceptual category. Experiment 2, a control study, tests whether simple exposure to stimuli without feedback or training has any effect on performance. Without training, discrimination remained at baseline levels, suggesting that short-term exposure alone is insufficient for category formation. The place contrast presented the biggest challenge for listeners, with some improvement in performance if a voicing cue co-occurred with the place contrast.

4aSC24. Comparison across languages using the multilingual Matrix Test: Which language is best to survive in a cocktail party? Birger Kollmeier, Sabine Hochmuth, Tim Jürgens, Ania Warzybok, and Thomas Brand (Cluster of Excellence Hearing4all & Medizinische Physik, Universität Oldenburg, Cluster of Excellence Hearing4all, HörTech gGmbH, Oldenburg D-26111, Germany, birger.kollmeier@uni-oldenburg.de)

The Matrix test (i.e., sentence test with fixed syntactic structure, but ten alternative words in each position which may lead to nonsense utterances) has the potential to overcome the inherently language-dependent incompatibilities of speech audiology. It is meanwhile available (with varying degree of supportive data) in Swedish, German, Danish, Dutch, American English, British English, French, Polish, Turkish, Spanish, Italian, Persian, Arabian, Finnish, and Russian. Several measures have been taken to make the tests as efficient, reliable and comparable across different languages as possible and to establish a *de-facto* standard. Using the Matrix concept it is also possible to estimate the "communication efficiency" of the different languages for this kind of sentences against each other. To eliminate the influence of the individual speaker, recordings with accent-free bilingual speakers (German-Russian and German-Spanish) were used to assess the respective speech reception threshold (SRT) for native listeners using stationary and fluctuating background noise. The results show both an inter-speaker and inter-language effect in the order of 3 dB. The latter is larger between German and Spanish than between German and Russian. The origin of these effects (such as long-term speech spectrum and the relative information content of consonants and vowels) will be discussed.

4aSC25. Nasals resonances in diphthongization: A preliminary study by nasal and oral acoustic output recording separately. Rita Demasi and Didier Demolin (Gipsa-Lab, Université Stendhal , 1180, Ave. Centrale BP25, Grenoble, Rhone-Alpes 38031, France, ritademasi@gmail.com)

Our goal is to find new acoustical evidences that characterize the nasality correlation by recording the month and nostril signal separately. This allows visualize the nasal and vocalic resonances individually. Few studies cover the nasal diphthongization and this phenomenon combines a partial nasalized vowel and a nasal glide. In Brazilian Portuguese (BP) the diphthong /aw/ and /ãw̃/ are distinctive. All data were recorded by Handle Separator from Glottal Enterprises. This records two acoustics output in different channels using two microphones. The plate is supported between the mouth and the nose. Six speakers from Paulistano dialect were recorded. The corpus covered back diphthongs in offset: [paw]; [saw]; [maw]; [taw]; [kaw]; [pãw̃]; [sãw̃]; [mãw̃]; [kãw̃] and [tãw̃]. In this task, each subject had to read the carry-sentence three times: [dʒigʊ_todʊ dʒigɐ]. Because the acoustic outputs are mixed, this method simplifies the separation between the both signal. The partial results show that the waveform and the spectrogram in this signals have a different configuration. In /ãw̃/, the nasal waveform starts and finishes with a very-low energy. The higher energy is concentrated between the boundary of the both vocalic segments. In the spectrogram, the formants configuration are plates and they used to lose energy gradually (average: Fn1 374Hz, Fn2 2333 Hz, and Fn3 3049 Hz). This is different in oral format configuration where F2 has a descent movement and F3 has a ascend movement.

4aSC26. Effect of musical experience on tonal language perception. Abigail Chua and Jason Brunt (Biola Univ., 13800 Biola Ave, La Mirada, CA 90639, abigail.j.chua@biola.edu)

Potential connections between musical experience and language learning ability have been discussed and debated about in neurological and musical psychology literature. The identification of Mandarin tones was tested in non-Mandarin speakers. The dependent variable was the accuracy of tone identification in the mandarin phrases. A simple questionnaire was used to measure musical experience. Musical experience and experimental trial accuracy were related. Non-significant effects are also discussed. A t-test was used to compare identification accuracy of those with musical experience to those without musical experience. Musicians had higher test trial accuracy scores ($M = 0.311$, $SD = 0.056$), than nonmusicians ($M = 0.275$, $SD = 0.03$). The difference was significant $t(31) = 2.228$, $p = 0.033$, suggesting that the presence of musical training increases the effectiveness of skill in Mandarin tone identification for non-Mandarin speakers. There was a strong correlation for accuracy and overall years of practice $r(29) = 0.4$, $p = 0.02$. The more years of overall practice musicians had the higher the accuracy they demonstrated in the test trials. There was no correlation between accuracy and years of interval training, years of playing an instrument, years of advanced musical study, or years of current practice.

4aSC27. Acoustic variability in the speech of second language learners of American English as a function of accentedness. Bruce L. Smith (Communications Sci. and Disord., Univ. of Utah, 390 S. 1530 E., Rm. 1216, Salt Lake City, UT 84112, bruce.smith@hsc.utah.edu) and Rachel Hayes-Harb (Linguist, Univ. of Utah, Salt Lake City, UT)

The primary issue of interest in the present study concerned acoustic variability among L2 learners of English with different degrees of accentedness. Specifically, we were interested in determining whether L2 learners with stronger accents differ from L2 learners with weaker accents in terms of the amount of within-subject variability they manifest when producing English consonants and vowels. Twenty L2 English learners from nine different L1 backgrounds and a group of 20 native English control subjects produced a number of target sounds contained within CVC words that were embedded in a carrier phrase. Accent ratings for the twenty L2 talkers were obtained, and acoustic measurements were made of various consonants and vowels; coefficient of variation [(S.D. $\div$ mean) $\times$ 100] was also computed for each of the acoustic measures. A number of temporal and spectral comparisons were made between L2 talkers with stronger versus weaker accents and with the native control subjects. Results indicated that although L2 subjects with stronger accents sometimes showed greater inter-subject (i.e., group) variability, they did not typically show more within-subject (i.e., token-to-token) variability than subjects with weaker accents, regardless of how accurate they were in producing native-like consonants and vowels.

4aSC28. Native English speakers' perception of Arabic emphatic contrasts. Kristie Durham (Dept. of Linguist, Univ. of Utah, 255 S Central Campus Dr., Rm. 01400, Salt Lake City, UT 84112, Kristie.Durham@utah.edu), Aleksandra Zaba (Second Lang. Teaching and Res. Ctr., Univ. of Utah, Salt Lake City, UT), and Rachel Hayes-Harb (Dept. of Linguist, Univ. of Utah, Salt Lake City, UT)

In Arabic, emphasis (secondary velar/pharyngeal constriction) distinguishes some consonants. Native Jordanian Arabic speakers have been shown to rely more heavily on the rime than the onset of CVC syllables when identifying plain versus emphatic onsets (Jongman *et al.* 2011). We investigated whether native English speakers similarly rely on the rime when discriminating Arabic plain-emphatic pairs. We also investigated the influence of vowel quality on discrimination performance. Native English speakers (no Arabic experience) performed an AXB task involving cross-spliced CVCs with plain/emphatic onsets/rimes. Our subjects also relied more heavily on the rime than on the onset; this effect was most robust when the V was /æ/, followed by /u/ and /i/. A cross-language vowel identification task revealed that subjects identified Arabic /æ/ in emphatic contexts as systematically different English vowels than in plain contexts, with only 10% overlap in vowels identified. The overlap for /i/ and /u/ was much higher, at 84% and 91%, respectively. We thus found that native English listeners, like native Arabic listeners, rely on the rime to make onset emphasis judgments, this effect is moderated by vowel, and the influence of the preceding vowel may be related to the mapping between vowel allophones and English vowel categories.

4aSC29. Spontaneous speech variability across languages: Labial and velar stops. Natasha L. Warner, Miguel Simonet (Dept. of Linguist, Univ. of Arizona, Box 210028, Tucson, AZ 85721-0028, nwarner@u.arizona.edu), Benjamin V. Tucker (Linguist, Univ. of AB, Edmonton, AB, Canada), Dan Brenner, Maureen Hoffmann, Alejandra Baltazar, Andrea Morales, and Yamile Diaz (Linguist, Univ. of Arizona, Tucson, AZ)

Stops such as /ptkbdg/ are perhaps the most-studied type of consonant in all of phonetics, and they have well-defined acoustic properties that one expects to find in a typical pronunciation. However, casual spontaneous speech reveals highly variable realizations of stops, ranging from voiceless stops with silent closure, burst, and aspiration noise, to weak approximants

with only a slight weakening of formants, to deletion. Even careful speech reveals considerable variability. We examine acoustic realizations of intervocalic stops in Dutch, Spanish, Japanese, and English, as well as the L2 English speech of the native Dutch, Spanish, and Japanese speakers. For each speaker, we measure data from spontaneous casual conversation and from careful word-list reading. In this presentation, we focus on realizations of /pbkg/. Preliminary results indicate that Dutch speakers variably transfer word-final devoicing of Dutch voiced stops to their English /bg/, but that they do not rely on the Dutch /x/ (orthographic "g") as a source of their English /g/. Results also show that Spanish /bg/ in conversation are almost categorically approximants or nearly deleted. Spanish speakers, especially those who learned English late, appear to apply this pronunciation to English consonants as a reduced speech style.

4aSC30. Prevention of learning of a non-native phonetic contrast by prior exposure to the contrasting stimuli while performing an irrelevant visual task. Beverly A. Wright, Jessica S. Conderman, Matthew K. Waggenspack (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60202, b-wright@northwestern.edu), and Nicole Marrone (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

Exposure to an acoustic stimulus can facilitate learning when encountered near in time to practice on a perceptual task. Here, we explored the possibility that this learning enhancement arises in part because the stimulus exposures encountered in the absence of practice amplify the internal representation of the stimulus and this amplification then remains during the subsequent practice period. If this is the case, learning should be disrupted if the stimulus representation is instead suppressed during stimulus exposure without practice, because that suppression should also spread to the practice period. To test this idea, we trained listeners on a non-native phonetic-contrast categorization task using regimens in which a period of practice followed a period of stimulus exposure without practice in each daily session. We manipulated the extent to which listeners presumably suppressed the auditory stimuli that were presented without accompanying practice by varying the attentional demands of a visual task performed during their presentation. Learning decreased markedly as the attentional demand during these periods increased. Thus, it appears that the magnitude of the internal stimulus representation affects learning and that changes in this magnitude can spread beyond the time in which they are induced to promote or interfere with learning. [Work supported by NIH.]

4aSC31. The rhythm of Aviation English by Native American English speakers. Julia Trippe and Eric Pederson (Linguist, Univ. of Oregon, 439 Almaden St., Eugene, OR 97402, trippe@uoregon.edu)

Air traffic controllers (ATC) and pilots at international airports must speak Aviation English (AE). Native and non-native English speakers alike must learn and effectively communicate using this technical language based on standard English. This project calculates the rhythmic profile of Native Speaker Aviation English (NSAE), which serves as the target for learners of AE and against which potential communication failures can be evaluated. NSAE rhythmic profile can be contrasted with the first language (L1) prosody to evaluate learner AE production and model training methods for specific L1 AE learners. NSAE generally exhibits flat intonational contours, so we focus on rhythm metrics. Our previous study's findings demonstrated that NSAE metrics pattern differently than standard American English, falling between "stress-timed" and "syllable-timed" languages. Rhythm metrics based on consonant and vowel duration are affected by AE's lack of function words (i.e., fewer reducible vowels), standard phraseology (producing prosodic chunking), and rapid speech rate (reflecting compressibility differential between vowel and consonant segments). We are training an automatic speech aligner to segment ATC NSAE and calculating a baseline for American NSAE using qualitative metrics (Ramus 2000; Low *et al*., 2000; Dellwo 2006). We will present our findings on how NSAE patterns with similarly evaluated languages.

4aSC32. The potential segmental influence of Taishanese (first language) on English (second language) intelligibility. Tracy Mai and Emily Wang (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 1611 West Harrison, Suite 530, Chicago, IL 60612, emily_wang@rush.edu)

This study examined the influence of segmental differences in Taishanese Chinese on English speech intelligibility. There are over half a million Taishanese speakers in the United States. However, very little is known about the linguistic interference between the two languages. We hypothesized that the English speech intelligibility of Late Bilingual speakers would be significantly reduced by the interference of their first language of Taishanese. Furthermore, the main source of their reduced speech intelligibility would be the segmental interference from the Taishanese on English. Speech data from a focused set of vowels and consonants as well as controlled spontaneous speech were collected from three different Speaker Types: Late Bilinguals (4), Sequential Bilinguals (2), and Monolingual English speakers (2). Acoustic analyses and perceptual experiment were conducted. The primary outcome measures were perceived speech intelligibility and mean Number of Real Words (NRW) per utterance. The secondary outcome measures were duration and formant frequencies of vowels, VOT for syllable-initial stops, and syllable duration of syllable-final stops. The results showed that the Late Bilingual speakers had significantly reduced English speech intelligibility ($p < 0.01$). The segmental-level differences from both vowels and consonants between Taishanese and English were responsible for the reduced speech intelligibility.

4aSC33. Perception of conversational and clear speech syllables by native and non-native English-speaking listeners. Catherine L. Rogers, Marissa Voors, and Jenna Luque (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

In a recent study later, but not earlier, learners of English as a second language produced a smaller clear-speech benefit than native English-speaking talkers for vowels produced in six /bVd/ syllables (Rogers *et al*., 2010, JASA **123**, 410–423). The present study compares perception of the same syllables by native and non-native English-speaking listeners. Conversational and clear-speech productions of the target syllables, "bead, bid, bayed, bed, bad," and "bod," were selected from three monolingual English speakers who had produced a significant clear-speech benefit in Rogers *et al*. (2010). The syllables were then mixed with noise at several signal-to-noise ratios (SNRs). Perception of these stimuli by three groups of listeners will be examined: (1) monolingual native English speakers, (2) 'early' learners of English as a second language, with an age of immersion (AOI) of 12 or earlier, and (3) later learners of English as a second language, with an AOI of 15 or later. Analyses of results of the six-alternative forced-choice task will focus on comparisons across listener groups, for the following measures: (1) estimates of clear-speech benefit at approximately 50% correct; (2) performance at a common SNR; and (3) estimates of the slope of the psychometric function. [Work supported by NIH.]

4aSC34. A virtual environment for modeling the acquisition of vowel normalization. Andrew R. Plummer (Ohio State Univ., 1712 Neil Ave, Columbus, OH 43210, plummer@ling.ohio-state.edu)

Vowel normalization is a computation that is meant to account for the differences in the absolute direct (physical or psychophysical) representations of qualitatively equivalent vowel productions that arise due to differences in speaker properties such as body size types, age, gender, and other socially interpreted categories that are based on natural variation in vocal tract size and shape. We present a virtual environment for vocal learning which provides the means to model the acquisition of vowel normalization, along with other aspects of vocal learning. The environment consists of models of caretaker agents representing five different language communities—American English, Cantonese, Greek, Japanese, and Korean—

derived from vowel category perception experiments (Munson *et al.*, 2010, Plummer *et al.*, 2013) and models of infant agents (Plummer, 2012, 2013) that "vocally interact" with their caretakers. Moreover, we develop a model of caretaker social and vocal signaling in response to infant vowel productions, and of an infant's internalization of these signals and the internal computations over them. More broadly, we model the acquisition of vowel normalization within a developmental framework encompassing a suite of vocal learning phenomena, including language-specific caretaker vocal exchanges, perceptual warping, and multisensory matching and narrowing.

THURSDAY MORNING, 8 MAY 2014 552 A/B, 8:00 A.M. TO 12:00 NOON

Session 4aSP

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

Kainam T. Wong, Chair
Dept. of Electronic & Information Eng., Hong Kong Polytechnic Univ., DE 605, Hung Hom KLN, Hong Kong

Invited Papers

8:00

4aSP1. Deconvolution-based acoustic source localization and separation algorithms. Mingsian R. Bai and Chia-Hao Kuo (Power Mech. Eng., Tsing Hua Univ., 101 Sec. 2, Kuang_Fu Rd., Hsinchu 30013, Taiwan, msbai63@gmail.com)

In this paper, localization and separation of acoustic sources are examined. Depending on the number of sources in relation to the array channels, the problem is investigated in terms of underdetermined and overdetermined configurations. In the underdetermined configuration, virtual monopole sources are assumed in uniformly spaced angles. The problem is then formulated into compressive sampling (CS) problem which can be solved by using the linearly constrained -norm convex (CVX) optimization. The solution yields the directions of real sources and the source signal spectrum, which enables localization and reconstruction of sources at one shot. In the underdetermined configuration, source localization and signal separation is carried out in two steps. First, the directions of arrival (DOA) are estimated with Minimum Variance Distortionless Response (MVDR) or Multiple Signal Classification (MUSIC). Next, Tikhonov regularization (TIKR) is utilized to recover the source spectrum. In the localization problem for both configurations, Neyman-Pearson detector is employed to determine thresholds for source detection. Numerical and experimental results show that the proposed methods produce improved speech quality in terms of mean opinion score (MOS) in perceptual evaluation of speech quality (PESQ) test.

8:20

4aSP2. Reproduction of higher order virtual sources using loudspeaker arrays. Jung-Woo Choi and Yang-Hann Kim (Mech. Eng., KAIST, YuseongGu GuseongDong 373-1, Daejeon 373-1, South Korea, khepera@kaist.ac.kr)

Reproduction of a monopole virtual source has been extensively studied for providing the locatedness or directional information to the listener. The monopole virtual source is often assumed to be omni-directional, and the driving signals of multiple loudspeakers are determined such that the uniform radiation pattern can be reproduced in space and time. The virtual source of which radiation pattern consists of higher order radiation pattern, however, can also be reproduced for the control of perceived stage width or focusing of acoustic energy. In this work, we introduce various applications involving the higher order radiation pattern. The magnitude or phase distribution rapidly changing in space and time is reproduced by an array of loudspeakers, and the limitation of the conventional reproduction technique based on the stationary phase approximation, e.g., wave field synthesis, is demonstrated. A single layer formula to eliminate the artifact of the conventional technique is addressed.

8:40

4aSP3. Distance perception in the sound field reproduced by a linear loudspeaker array. Dong-Soo Kang, Jung-Woo Choi, and Yang-Hann Kim (Mech. Eng., KAIST, 291 Daehak-ro, Yuseong-gu, KAIST, ME Bldg Rm#4114, Daejeon 305701, South Korea, dooly0819@kaist.ac.kr)

When a sound field from a virtual sound source is reproduced by a linear loudspeaker array, a listener in the sound field can perceive the distance of the virtual source, as well as its direction. It has been known that the perception of distance is affected by many acoustic parameters, such as the loudness change, direct-to-early reflection energy ratio, and interaural level difference (ILD). Among them, ILD is the dominant cue to perceive distance when the source is located near the lateral side of the listener [Brungart and Rabinowitz, J. Acoust. Soc. Am. **106**(3), Sept. 1999]. Nevertheless, ILD of the reproduced sound field are not identical to that of the target sound field, because the loudspeaker array reproducing the sound field has many practical limitations such as spatial aliasing and truncation of array aperture. To identify these artifacts, especially for the virtual source in a close proximity to the listener, a head-scattering model is constructed using a simple rigid sphere. The ILDs at various head locations are then calculated and compared to those of the target sound field. From the observations on ILD change, a driving function is modified to reconstruct ILDs of the target sound field.

9:00

4aSP4. Broadband acoustic-source localization using passive sonar via multitask learning. Pedro A. Forero and Paul A. Baxley (Maritime Systems Div., SPAWAR Systems Ctr. - Pacific, 53560 Hull St., San Diego, CA 92152, forer002@umn.edu)

Passive sonar is an attractive technology for underwater acoustic-source localization that enables the localization system to conceal its presence and does not perturb the maritime environment. Notwithstanding its appeal, passive-sonar-based localization is challenging due to the complexities of underwater acoustic propagation. Different from alternatives based on matched-field processing whose localization performance severely deteriorate when localizing multiple sources and when faced with model mismatch, this work casts the broadband underwater acoustic-source localization problem as a multitask learning (MTL) problem, thereby enabling robust and high-resolution localization. Here, each task refers to a sparse signal approximation problem over a single frequency. MTL provides an elegant framework for exchanging information across the individual regression problems and constructing an aggregate (across frequencies) source localization map. The localization problem is formulated as a stochastic least-squares optimization problem with a group sparsity constraint enforcing a common support across frequency maps. Efficient algorithms based on block coordinate descent are developed for solving the localization problem. Predictor screening rules are also developed to further reduce the computational complexity of the proposed method. Numerical tests on real data illustrate and compare the localization performance of the proposed algorithm to that of competitive alternatives.

9:15

4aSP5. Transient detection via acoustic particle velocity multi-mission sensor. Latasha Solomon and Leng Sim (US Army Res. Lab, 2800 Powder Mill RD, Adelphi, MD 20783, latasha.i.solomon.civ@mail.mil)

In this research, we compare the direction of arrival (DOA) accuracy of a micro-electro-mechanical systems (MEMS) based acoustic particle velocity sensor developed by Microflown Technologies with that of a collocated, 1-m tetrahedral array. When deployed as an unattended sensor system, the Acoustic Multi-Mission Sensor (AMMS) greatly facilitates hardware set-up and periodic maintenance. An array of microphones is now replaced by a single sensor, saving in overall system cost, size, weight, and power usage. The single sensor has the capability to measure both the (scalar) sound pressure and the (vector) acoustic particle velocity, thus providing DOA estimates. This research will explore performance and determine limitation of the two sensors in complex environments as well as open fields for detection of both small arms fire (SAF) and rocket propelled grenades (RPGs).

9:30

4aSP6. A study of broadband sensor location selection using convex optimization in very large scale arrays. Yenming Lai and Radu V. Balan (Appl. Mathematics, Statistics, and Sci. Computation, Univ. of Maryland, 5010 Pierce Ave., College Park, MD 20740, yenming.mark.lai@gmail.com)

Consider a sensing system using a large number of N microphones, placed in multiple dimensions to monitor a broadband acoustic field. Using all the microphones at once is impractical because of the amount of data generated. Instead, we choose a subset of D microphones to be active. Specifically, we wish to find the set of D microphones which minimizes the energy of the interference gains at multiple frequencies while monitoring a target of interest. A direct, combinatorial approach—testing all N choose D subsets of microphones is impractical because of problem size. Instead, we use a convex optimization technique that induces sparsity through a l1-penalty to determine which subset of microphones to use. We measure the energy of the interference gains in three ways: the maximum gain, the average gain, and the average squared gain and compare the results. Furthermore, we assume one reflection off of each wall in our problem setup and minimize the gains of the reflections. We test the robustness of the our solution through simulated annealing and compare its performance against a classical beamformer which maximizes SNR. We also do exhaustive searches to compare the performance of our algorithm against the global optimum.

9:45

4aSP7. Estimation algorithm coordinates source signal towed long antenna. Igor Y. Anikin (Concern CSRI Elektropribor, JSC, 30, Malaya Posadskaya Str., St. Petersburg 197046, Russian Federation, anikin1952@bk.ru)

In some sonars is require the use of towed long antenna. By “large”long antenna is understood conventionally antenna directivity pattern width is less than some value, for example, less than 1°. Due to the large antenna length when towing a change in its form, as well as change of coordinates the source relative to antenna. For this reason, the source coordinates are determined with errors. The report discusses an algorithm for estimating the coordinates the source of towed long antenna. Algorithm consists in the separation of the antenna into several sections. Section length is chosen so that the antenna directivity pattern width of the section was 1°... 2°. Each section is formed by the fan of directivity patterns. The joint processing of signals from the output of the directivity patterns of the fan formed by sections of the antenna provides the coordinates of the source. Results of mathematical modeling error estimates of coordinates, offered algorithm are compared with the potential errors that follow from the Cramer-Rao inequality.

10:00–10:15 Break

10:15

4aSP8. Random matrix theory model for mean notch depth of the diagonally loaded minimum variance distortionless response beamformer for a single interferer case. Saurav R. Tuladhar, John R. Buck (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., ECE Dept, UmassD, North Dartmouth, MA 02747, stuladhar@umassd.edu), and Kathleen E. Wage (ECE, George Mason Univ., Fairfax, VA)

Adaptive beamformers (ABFs) suppress interferers by placing notches in the beampattern at interferer directions. This suppression improves the detection of weaker signals of interest even in the presence of strong interferers. The magnitude of the notch depth (ND) is an important parameter governing the adaptive gain obtained from using ABFs over conventional beamforming in the presence of interferers. This research derives models for the mean ND of a diagonally loaded minimum variance distortionless response (MVDR) beamformer for a single interferer case. The model describes the mean ND as a function of the number of snapshots, the number of sensors in the array, the interferer to noise ratio (INR) level, the interferer direction, and the diagonal loading level. The derivation exploits random matrix theory (RMT) results on the behavior of the eigenvectors of the spiked covariance matrix. The RMT based ND model predictions are in close agreement with simulation results over a range of INR values and number of snapshots.

10:30

4aSP9. Improved modal dispersion estimation using vertical array beamforming. Valerie Vinciullo (Appl. Physical Sci. Corp. , 4 Hillside Ave., unit 2, Pawcatuck, Rhode Island 06379, vvinciullo@my.uri.edu), Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Kevin Cockrell (Appl. Physical Sci. Corp. , Groton, CT), and James H. Miller (Appl. Physical Sci. Corp. , Narragansett, Rhode Island)

Geoacoustic inversions using modal dispersion data is a very robust technique to estimate properties of the shallow water sediments. Accurate estimation of the modal arrival times is required for improving the accuracy of the inversion. A time-frequency analysis of the single hydrophone data is typically used to extract modal arrival times. This study explores the possibility of incorporating the data from a vertical line array (VLA) to enhance the accuracy of arrival time estimation. The method relies on beam forming in horizontal wavenumber at each instant in time to produce a time-frequency-wavenumber diagram (movie) which will provide an extra dimension to help separate the modes. For a given time and horizontal wavenumber, the arrival time is unique since the group speed is uniquely determined by the frequency and horizontal wavenumber. So, even if the time and wavenumber resolution is not sufficient identify individual modes, the 3-D surface plot of arrival time versus frequency and wavenumber can be created. The shape of that surface can be compared to simulated surfaces for geoacoustic inversion, even if mode arrivals appear to overlap. This approach will be tested using synthetic data and the length and spacing requirements of the VLA will also be investigated. [Work supported by Office of Naval Research, code 322OA.]

10:45

4aSP10. Matched-field source localization with non-synchronized sensor arrays. Stan E. Dosso (School of Earth & Ocean Sci, Univ of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

This paper considers matched-field ocean acoustic source localization based on acoustic field measurements at an array of sensors which are not synchronized in time or at an array which is comprised of non-synchronized combinations of synchronized sub-arrays. Standard matched-field methods are based on acoustic-field measurements at a time-synchronized array to allow coherent processing over space (the array aperture). For non-synchronized systems, frequency-coherent/space-incoherent processing can be applied if the complex source spectrum (amplitude and phase) is known, but this is rarely the case in practical applications. However, time- and frequency-coherent processing are not the only possibilities. Maximum-likelihood methods can be applied to derive optimal matched-field processors for any state of source/receiver information. Using this method, optimal processors can be developed for broadband matched-field localization with any combination of synchronized and/or non-synchronized components based on the fact that the source amplitude spectrum is the same (although unknown) for all receivers (the phase spectrum is both unknown and variable for non-synchronized components). Bayesian inversion methods are employed to quantify the source-localization information content for various array scenarios.

11:00

4aSP11. Research on rotary spiral array applied in near-field acoustical holography. Chen Lin-Song (Power Eng. Dept., Naval Univ. of Eng., Jiefang St. 717, Wuhan, Hubei 430033, China, 13294153193@163.com)

This paper presents a new method to apply a spiral array in nearfield acoustical holography (NAH). Usually, a NAH array needs much more microphones than beamforming array does. Superior to a uniform planar array or linear scanning array, this spiral array rotates to get more measuring data. Without any static referring microphone, a numerical method was suggested to estimate the phase difference measured at different time. Numerical simulations and a series experiment confirmed that this method is adequate for the sound below 450 Hz. It is especially useful for using a random planar array at NAH mode, while the beamforming mode can only cover the higher frequency band.

11:15

4aSP12. A constrained adaptive beamforming algorithm for spherical microphone arrays. Gary W. Elko (mh Acoust. LLC, 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com) and Jens M. Meyer (mh Acoust., Fairfax, Vermont)

In this presentation, we will present a novel constrained adaptive beamformer algorithm that utilizes an inherent property of spherical harmonic eigenbeams which form the bases signals for spherical microphone array beamforming. Two simple constraints are placed on the weights to preclude the adaptive beamformer from nulling signals arriving from a desired "look" direction. The adaptive algorithm has been simulated for some simple acoustic fields as well as a diffuse field. We have implemented the algorithm in realtime on mh acoustics em32 Eigenmike® spherical microphone array and we will present some measurement results.

11:30

4aSP13. A rigid-body model for diffraction imaging of solid objects: Theory and experimental results. Edward H. Pees (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, edward.pees@navy.mil)

Acoustical imaging via the method of diffraction tomography is typically applied to weakly scattering, fluid objects, wherein the first Born approximation holds. Nonetheless, the technique can be applied to strong scatterers in a meaningful way if an appropriate object function is considered. In this talk, the theoretical form of the object function for a rigid body is developed along with an inversion formula for centripetal, broadband data collection. Applying the latter to experimental, underwater echo data from a variety of objects, reconstructions are presented and interpreted in terms of a Kirchhoff boundary condition. The approach can potentially reveal the relative importance of different scattering mechanisms in the overall pressure field reflected from a body by how closely the rigid body object function is reconstructed. Morphological characteristics may also be identified for objects that are, for example, hidden or buried.

11:45

4aSP14. Multichannel myopic deconvolution using ambient noise sources. Ning Tian, Justin Romberg (School of Elec. and Comput. Eng., Georgia Inst. of Technol., 30 5th St. NE, Unit 606, Atlanta, GA 30308, ningtian@gatech.edu), and Karim Sabra (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The ocean ambient noise has been increasingly utilized for ocean passive sensing and monitoring applications. By recording the received signals from the same individual noise source (for example, the shipping noise) at multiple hydrophones simultaneously, we develop a framework, called multichannel myopic deconvolution, which can allow us jointly estimate the source and the channel responses without any assumption about the source, but using some priori knowledge of the channel. Our work on this classical signal processing problem has two novel aspects. First, we recast the corresponding bilinear system of equations as a linear system with a rank constraint. This allows us to apply recently developed algorithms and analytical tools from the field of low-rank recovery to the blind channel estimation problem, yielding insight into the conditions under which accurate channel estimation is possible. Second, we incorporate (continuous-time) parametric uncertainty about the Green's functions as subspace constraints in the low-rank recovery problem. These subspaces are generated in a systematic way using the singular value decomposition, and their dimension can be directly related to the amount of priori knowledge we have about the channel. We will present simulations in shallow water environments of the proposed approach from relatively short observation times.

Session 4aUW

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

David R. Dall'Osto, Cochair
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Appl. Phys. Lab., Univ. of Washington, Mech. Eng., 1013 NE 40th St., Seattle, WA 98105

Chair's Introduction—8:20

Invited Papers

8:25

4aUW1. Using hydrophones as vector sensors. Selda Yildiz, LeRoy M. Dorman, W. A. Kuperman (Scripps Inst. of Oceanogr., University of California, San Diego, La Jolla, CA 92093-0238, wkuperman@ucsd.edu), Karim Sabra (School of Mech. Eng., Georgia Inst. of Tech, Atlanta, GA), Philippe Roux (Institut des Sci. de la Terre, Universite Joseph Fourier, Grenoble, France), Dale Green (Teledyne Benthos, 49 Edgerton Dr, N. Falmouth, MA), Stephanie Fried, and Henrik Schmidt (Mech. Eng., Mass. Inst. of Tech., Cambridge, MA)

Hydrophone arrays with spacing much less than an acoustic wavelength can be converted to vector sensors. Subsequent vector sensor signal processing can then be applied. Two particular applications are presented: The first is converting very low frequency acoustic data to seismic type data that contain polarization information and the second is getting directional information from sub wavelength acoustic arrays. We start with a review of the simple theory followed by some illustrative simulation examples. We then apply these signal processing methods to ocean acoustic data.

8:45

4aUW2. Tank acoustics, and sound source localization by plainfin midshipman fish (*Porichthys notatus*). David Zeddies (JASCO Appl. Sci., 2004 Coleridge Dr., #101, Silver Spring, MD 20902, David.Zeddies@jasco.com), Michael D. Gray, Peter H. Rogers (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Richard R. Fay (Parmly Hearing Inst., Loyola Univ., Falmouth, Massachusetts), and Joseph A. Sisneros (Dept. of Psych., Univ. of Washington, Seattle, WA)

A series of experiments was undertaken to investigate methods of sound source localization by fish. In these experiments, positive phonotaxic responses of gravid female plainfin midshipman fish (*Porichthys notatus*) to low-frequency, playback tones (80–90 Hz) were studied as they approached sound sources. The sound fields for simple (monopole) and relatively complex (dipole) sources within the behavioral arena were measured and characterized in terms of pressure and particle motion. Results indicate that female midshipman fish are able to locate sound sources in the near field using acoustic cues alone, and that they used the particle motion vectors to locate the source in both the monopole and dipole sound fields. The tank acoustics were modeled and compared to the measured pressure and particle motion sound fields. [This work was supported by the National Science Foundation.]

Contributed Papers

9:05

4aUW3. Real-time acoustic monitoring and source level estimates of walruses in the northeastern Chukchi Sea using particle velocity sensors. Xavier Mouy (JASCO Appl. Sci., Victoria, Br. Columbia, Canada), Julien Delarue, Bruce Martin (JASCO Appl. Sci., 32 Troop Ave., Ste. 202', Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), David Hannay (JASCO Appl. Sci., Victoria, Br. Columbia, Canada), Chadwick Jay, and Anthony Fishbach (US Geological Survey, Anchorage, AK)

Particle motion sensors measure the vector component of the sound field. In underwater acoustics, they are used for studying the physics of the sound field, evaluating the potential effects of sound on fish, and defining the direction of arrival (DOA) of sound sources. Measuring the DOA in the vertical and horizontal plane allows two separate receivers to localize an acoustic source in three dimensions. In July 2013, we used two custom-built, real-time particle velocity acoustic recording systems to record and localize vocally active walruses in the water near groups hauled out on ice in the northeastern Chukchi Sea. The system was equipped with a three-axis dipole sensors and a calibrated omni-directional hydrophone. It was deployed at the water surface and transmitted data in real-time to a support skiff. The range between the recorders, support skiff, and calling animals was usually less than 200 m and typically within a few tens of meters, allowing for simultaneous visual observations. Calling walruses were localized using cross-fixes of acoustic bearings. Source levels were estimated by adding modeled frequency-dependent transmission losses to the received levels in each 1/3-octave-band obtained from the calibrated omni-directional hydrophone. Only calls with high signal-to-noise ratio were used in this analysis. The use of the particle velocity sensor allowed for the first source level measurements of walrus grunts and bell calls in the wild.

9:20

4aUW4. Estimates of the bottom reflection coefficient involving vector sensor. Jee Woong Choi (Dept. of Marine Sci. & Convergent Technol., Hanyang Univ., 55 Hanyangdaehak-ro, Ansan 426-791, Korea, Republic of, choijw@hanyang.ac.kr), David R. Dall'Osto, and Peter H. Dahl (Appl. Phys. Lab. and Mech. Eng. Dept., Univ. of Washington, Seattle, WA)

Estimates of the bottom reflection coefficient made in the frequency range of 4–8 kHz, as part of the Targets Reverberation Experiment (TREX) are presented. The TREX experiment took place in the Gulf of Mexico, near Panama City, FL, in waters 20 m deep. At the measurement site the sediments are loosely classified as fine sand. The reflection coefficient R is estimated over the nominal grazing angular range 1 to 20 deg., using measurements made at ranges 50 to 800 m and received on a vertical line array (length 1.6 m). The arrival time and magnitude of the bottom reflection is determined by the matched filtered output of a frequency modulated signal, 4–8 kHz. In addition, the match filter processing technique is applied to the vector sensor data (measured simultaneously and co-located with the line array.) This allows for an extraction of the active intensity contribution associated with the bottom reflection, and provides a vector intensity-based estimate of the bottom reflection coefficient. The estimates of the bottom reflection arrival time are also used to time-gate simultaneously transmitted cw tones (1–4 kHz) to analyze the Lloyd's mirror pattern associated with seabed reflection. [Research supported by ONR, with partial support from ONRG.]

9:35

4aUW5. Underwater techniques to characterize the near scattered acoustic vector field. Robert J. Barton, Geoffrey R. Moss, Brian K. Amaral, Georges Dossot (NUWC, 1176 Howell St., Bldg. 1320, Code 1524, Rm. 260, Newport, RI 02841, georges.dossot@navy.mil), and Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA)

In this study, we investigate the properties of the scattered acoustic vector fields generated by simple geometric objects, including the infinite rigid plate, disk, and sphere. Analytical solutions are derived from acoustic target strength scattering models in the near-field region. Of particular interest is the understanding of the characteristics of energy flow of the scattered acoustic vector field in the near- to far- field transition region. We utilize the time and space separable instantaneous active and reactive acoustic intensities to investigate the relative phase properties of the scattered field. Numerical results are presented for the near region scattered acoustic vector field of simple objects in both two and three dimensions. Previous in-air measurements are summarized, and an approach to taking water-borne measurements is offered.

9:50

4aUW6. The interference structure of very low frequency vector acoustic field and its positioning application. Sun Dajun, Shi Junjie, Lv Yunfei, Lan Hualin, and Mei Jidan (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ. and College of Underwater Acoust. Eng., Harbin Eng. Univ., Rm. 1005, Shuisheng Bldg., No.145 Nantong St., Nangang District, Harbin, Heilongjiang, China, Harbin 150001, China, junjieshi@hrbeu.edu.cn)

Vector hydrophone has natural dipole beam-pattern, and can also simultaneously and colocatedly measure the scalar and vector information of ocean acoustic field, which makes it convenient to determine the direction of arrival (DOA) and represent the stable interference structure of vector acoustic field. Utilizing the DOA and interference structure information together acquired by single vector hydrophone, it is able to position the target of interest. Firstly, bearing-time course for passing-by target is obtained by using the line or continuous spectrum imbedded in the received signal of vector hydrophone. Then, CPA ratio between the CPA range and speed of target as well as CPA instant are estimated on the basis on LMS criteria. Finally, the speed of target can be determined through the theoretically predicted interference range and the real interference time interval to fulfill target positioning including DOA and range. The idea was effectively validated during the experiment that took place in October of 2010 in South China Sea nearly 100m depth. Moreover, the idea presented here can be easily extended for further application such as combining the waveguide invariant. [Work supported by the National 863 Project (No. 2011AA090502) and National Defense Foundation Project (B2420132004).]

10:05–10:25 Break

Invited Paper

10:25

4aUW7. Acoustic energy streamlines in inhomogeneous fluids. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

First introduced by Umov in 1873, wave energy streamlines offer an intuitive and informative description of energy flow much like conventional streamlines do for mass flow in fluid mechanics. Growing availability and increasing practical applications of acoustic vector sensors, such as sound-intensity meters, have led to a surge of interest to energy streamlines. In contrast to rays, which are essentially an asymptotic, short-wave concept, energy streamlines adequately represent arbitrary acoustic fields and reveal intricate and often unexpected details of the acoustic energy flow. Modern usages of the energy streamlines include studies of wave front dislocations, source localization, energy vortices in compressible fluids and elastic waveguides, and bounded beam diffraction. This paper will focus on applications of the energy streamlines to the description of reflection and refraction of acoustic waves at interfaces and to localization of low-frequency sound sources.

Contributed Papers

10:45

4aUW8. Observations of elliptical particle motion in shallow water and its dependence on source depth. David R. Dall'Osto (Acoust. Dept., Appl. Phys. Lab. Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Mech. Eng., Univ. of Washington, Seattle, WA)

Acoustic particle motion that follows an elliptical path coincides with a non-zero curl of the time-averaged intensity. This vector property is also observed as curvature in intensity (energy-flux) streamlines. Measurements of the acoustic intensity field made in shallow water are presented, along with simulations of the intensity field, to demonstrate some interesting relations between acoustic intensity and elliptical particle motion. Specifically, the direction in which intensity streamlines bend (sign of the curl of intensity) corresponds to the polarization of acoustic particle motion. For a source located in water, the polarization of particle motion depends on the modes of the underwater waveguide excited at a particular source depth. By raising a source up through the water column, an abrupt change in the polarization of particle motion can occur. This effect is examined with vector sensor data collected during an experiment near Panama City, FL. For a source located in air, elliptical particle motion is most evident a few wavelengths below the sea-surface where the contribution of the lateral (evanescent) wave is significant. This effect is examined with a recording of aircraft noise on both sides of the air-water interface made near Oak Harbor, WA.

11:00

4aUW9. Modeling the acoustic vector field to simulate glider-based acoustic processing methods. Georges Dossot (NUWC, 1176 Howell St., Bldg. 1320, Code 1524, Rm. 260, Newport, RI 02841, georges.dossot@navy.mil), Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA), and Edmund J. Sullivan (Prometheus Inc., Portsmouth, RI)

The feasibility of underwater gliders as passive acoustic receiving platforms is explored through simulated data. Over the last decade gliders have proven their worth as operational platforms to the oceanographic community, yet their merits as acoustic sensing platforms remain largely underexplored. Recently, the Office of Naval Research has equipped the Naval Postgraduate School with several gliders, which have now been fitted with acoustic vector sensors. To simulate real-world performance, the intensity vector field is modeled using the three-dimensional Cartesian version of the Monterey-Miami parabolic equation (MMPE) algorithm, which relies upon a split-step Fourier approach. Environmental information representative of the glider's sawtooth profile is incorporated as a three-dimensional sound speed profile, and incorporated into the PE model. These simulated data serve as the basis for signal processing techniques applicable to glider-based experimentation.

11:15

4aUW10. The influence of directional sea-surface waves on the acoustic intensity vector field. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Mech. Eng., Univ. of Washington, Seattle, WA)

The effects of a rough sea surface on shallow water acoustic propagation are examined using experimental data collected from the Target and Reverberation Experiment (TREX) which took place off of the coast of Panama City, Florida in May 2013. During the experiment, the sea surface directional-wave spectrum was measured by a pair directional buoys moored at the experimental site. Acoustic measurements were collected using a bottom deployed recording tower (depth 20 m), that coherently recorded data from an accelerometer-based vector sensor, and a horizontal and vertical line array. Measurements using an active source, lowered from the stern of a research vessel, were made along propagation paths perpendicular and parallel to the surface wind-waves at source receiver ranges corresponding to approximately 10, 20, and 40 water depths. Results show that the directional properties of the rough sea-surface influence both the azimuthal and vertical distribution of the forward scattered intensity. A frequency dependence in vertical angular spreading is identified for the frequency range 1 to 3 kHz. A partial explanation for this effect originates from differences in the directional wave spectral level corresponding to forward scattering Bragg wavenumbers that are computed from the angles of the trapped modes.

11:30

4aUW11. Research on the double vector hydrophones' location for underwater low frequency source depth identification. Anbang Zhao, Xuejing Song, Bin Zhou, and Xuejie Bi (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin Eng. University Underwater Acoust. Bldg., Rm. 812, Harbin, Heilongjiang, Harbin 150001, China, zhaoanbang@hrbeu.edu.cn)

The double vertically arranged vector hydrophones' pressure and horizontal velocity cross spectrum in Pekeris waveguide is derived, and the sign distribution of its active component is analyzed. The sign distribution varies with the horizontal ranges and source depths regularly, the signs change in a certain depth and the depth is defined as critical depth. By locating the vector hydrophones properly, a critical depth which is independent of horizontal range can be obtained, and this characteristic can be used for discriminating the source depth. The method of forecasting the vector hydrophones' locating depths according to the requirements of the critical depth is studied, and the forecast accuracy is validated by the simulation results. A reasonable set of the critical depth is conducive to discriminate the source depth accurately and effectively, which has extensive application prospects.

11:45

4aUW12. Joint estimation of frequency and azimuth using acoustic vector sensor signals based on sparse decomposition theory. Jinshan Fu (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Nantong St. 145, Haerbin 150001, China, fujinshan@hrbeu.edu.cn)

Acoustic vector sensor can obtain more information of sound field compared with scalar hydrophone. The sparse decomposition theory was put forward in the 90s of last century, and it provides a simple, flexible, and self-adaptive representation method of signal. Through sparse decomposition theory, it can essentially reduce the cost of signal processing and improve the compression efficiency. Space-time array manifold is constructed through signal analysis of single acoustic vector sensor (AVS). Based on sparse decomposition theory, the frequency and azimuth estimation algorithm is proposed, the frequencies and azimuths of multi-targets are estimated simultaneously by the joint estimation algorithm. Results using simulated data received from single acoustic vector sensor are illustrated. The accurate estimation of multi-targets' frequencies, azimuths, and signal amplitudes can be obtained using the estimation algorithm we deduced. Then, the influence of targets number, signal-to-noise (SNR), snapshots number on algorithm performance is analyzed.

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. 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.

4:30

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 $+5^{\circ}$ 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. λ/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.

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. λ/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.

Contributed Papers

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.

4p THU. PM

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