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 pierside 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^2nmi-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

4a THU. AM

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
Eng. Sci., Penn State, 953 McCormick Ave., State College, PA 16801-6530

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 noisebursts 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.