Session 4aAA

Architectural Acoustics and Noise: Experience in Forensic Architectural Acoustics

Matthew V. Golden, Cochair
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David Lubman, Cochair
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Invited Papers

11:00

4aAA1. Developing criteria for identifying acoustical defects. John LoVerde and David W. Dong (Veneklasen Associates, 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

In a construction defect lawsuit of a multifamily residential project, the determination of whether a defect exists often hinges on the criteria applied. For many acoustical items, such as plumbing and HVAC noise, there are no code requirements but a number of guidelines and recommendations. For items such as noise from traffic or airborne and impact sound isolation between units, minimum code requirements exist, but often a more stringent standard is applied. How does an expert decide when it is appropriate to apply an acoustical standard that is beyond that required by building codes? Project drawings, marketing materials, homeowner regulations, and other documents can provide indications of the intent and promise of the project as it relates to acoustical issues. The process is discussed with examples from recent cases.

11:20

4aAA2. Construction defect actions in residential construction in California after passage of senate bill 800. Pablo A. Daroux (Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, pdaroux@wiai.com) and Terry Wilkens (Law Offices of Ann Rankin, Oakland, CA)

Construction defects actions (lawsuits) involving the sale of new residences that are sold on or after January 1, 2003, are governed by California Civil Code Section 986, also commonly referred to as “SB800” or the “Builder’s Right to Repair Law.” This legislation has redefined the rights and responsibilities of purchasers, sellers, builders, and designers regarding construction deficiencies in significant ways. This presentation will discuss the details of this Legislation as it affects the work of Consultants as Designers and as Forensic Experts.

Contributed Paper

11:40

4aAA3. Overcoming flooring impact isolation performance failures due to adjacent condominium ceiling deficiencies. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

There is an increasing demand to renovate apartments and condominiums by installing hard-surfaced flooring. It has become typical for a homeowner’s association (HOA) to enact regulations requiring that upper level dwelling units making flooring upgrades from soft-surface flooring to hard-surface flooring meet specific minimum field impact insulation class (FIIC) or impact sound rating (ISR) performance standards. It is common for these HOA requirements to apply only to the party making the flooring surface change. It is rare that the adjacent ceiling condition or ceiling renovation is addressed by such HOA requirements. This paper presents field performance test results for several renovated condominiums in different multi-family residential buildings that failed homeowner’s association impact performance requirements due to lower-level ceiling deficiencies. Examples of such failures are discussed and successful custom remedies are presented, some of which avoided threatened litigation.

Session 4aAB

Animal Bioacoustics and Noise: Bioacoustic Contributions to Soundscapes I

John Hildebrand, Cochair
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Simone Baumann-Pickering, Cochair
Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093

Invited Papers

8:00

4aAB1. Bioacoustic contributions to the soundscape. John Hildebrand and Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093, jhildebrand@ucsd.edu)

The soundscape, the combination of all sounds found in the environment, has been divided into three components: geophony—sounds that are generated by non-biological sources such as wind and waves; biophony—sound generated by animals, excepting humans; and anthrophony—sound generated by human activities. The soundscape can be further divided by spatial, temporal, and frequency band variations. Spatial variations occur both owing to proximity to various sound sources and to the sound propagation environment. Temporal variations occur on seasonal, lunar, daily, and other cycles. Use of different frequency bands may be a response to spatial and temporal overlap of sounds. In this session, we consider biological components of the soundscape and how sounds produced by animals interact with each other and with non-biological and anthropogenic sounds. We suggest that studies using long-term passive acoustic monitoring may take a more holistic approach to analysis using the concept of soundscape. This approach emphasizes the connections and relations between sound events, rather than focusing on the extraction of individual sound events. The session draws on data collected both in terrestrial and marine settings to illustrate how the soundscape concept can be applied to better understand animal use of sound.

8:20

4aAB2. Soundscape ecology: A review of a new synthesis area of acoustics of landscapes. Bryan C. Pijanowski and Luis Villanueva-Rivera (Forestry and Natural Resources, Purdue Univ., 195 Marsteller St., 305 FORS Bldg., West Lafayette, IN 47906, bpijanow@purdue.edu)

Soundscape ecology is an emergent area of acoustics that attempts to synthesize the concepts of landscape ecology, bioacoustics, noise, music, ethics, and biogeography. By focusing on the interplay of three main sources of sound: biological, geophysical, and anthropogenic, we hope to understand how humans impact ecosystems at a variety of spatial and temporal scales. Another focus of our work is the identification of special ecological places that possess unique and highly valued soundscapes. I will review the current state of the science and efforts to move this field forward using the expertise across these varied disciplines along with the work that we are conducting in the temperate forest, tropical, and desert ecosystems of North America.

8:40

4aAB3. Composing soundscapes from real-time acoustic data streams. Michel Andre (Lab. of Appl. BioAcoust., Tech. Univ. of Catalonia, BarcelonaTech, UPC, Rambla Exposicio 24, Vilanova i la Geltrú, Barcelona 08800, Spain, michel.andre@upc.edu) and M. Andre (Ocean Sound Sci., CSA Ocean Sci. Inc., Stuart, FL)

The growing scientific and societal concern about the effects of underwater sound on marine ecosystems has been recently recognized through the introduction of several international initiatives aiming at measuring the environmental impact of ocean noise on large spatial and temporal scales. From a regulatory perspective, the European Marine Strategy Framework Directive includes noise as one of 11 descriptors to determine Good Environmental Status of the oceans. The Directive specifically requires Member States to provide a measure of annually averaged noise. The LAB has developed a software package that measures sound levels and monitors acoustic sources in real-time; this software was used for the LIDO project and is now operating to provide industry with an environmentally responsible approach (CSA, www.oceansound.com). The system is currently operating worldwide from several wired and radio-linked observatories. Recently, through a zero-cost contract with the CTBTO (Preparatory Commission for the Comprehensive nuclear-Test Ban Treaty Organization), years of data from hydroacoustic stations were analyzed to look for background noise trends and to detect cetacean presence. Here, we present the analysis of four CTBTO platforms, each covering 42 months of data, focusing especially on the estimation of background noise levels and the measurement of noise contributions from anthropogenic sources.

9:00

4aAB4. From pole to pole: Soundscapes of the Atlantic Ocean. Holger Klinck (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Hatfield Marine Sci. Ctr., 2030 SE Marine Sci. Dr., Newport, OR 97365, Holger.Klinck@oregonstate.edu), Jennifer L. Miksis-Olds (Appl. Res. Lab., Penn State Univ., State College, PA), Sharon L. Nieukirk, Haruyoshi Matsumoto, and Robert P. Dziak (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR)

Between July 2009 and December 2010, two identical calibrated hydrophone packages were deployed and continuously operated (2,000 Hz sampling rate) in the Fram Strait (79°N, 5.5°E) and the Bransfield Strait (62°S, 55.5°W). Analysis of these recordings were combined with a data set collected during the same time period by the Comprehensive Nuclear-Test-Ban Treaty Organization International Monitoring System (CTBTO IMS) hydroacoustic station HA10 (250 Hz sampling rate) at Ascension Island (8°S, 14.4°W). The combination of these datasets allowed a comparison of low-frequency noise levels in polar and tropic areas of the Atlantic Ocean. The recordings were analyzed for major natural (e.g., marine mammals, ice) and anthropogenic (e.g., shipping, seismic) contributors to the ambient sound field and their seasonal variability. Preliminary results indicate (1) a higher seasonal variability of ambient noise levels in polar regions compared to the tropics, (2) the seasonal variability of ambient noise levels in the Arctic and Antarctic were driven by different contributors, and (3) highest noise levels were observed in the Arctic in association with seismic oil and gas exploration. [Partial funding from NOAA/PMEL and the Korea Polar Research Institute.]

9:15

4aAB5. Nonlinear time series analysis of snapping shrimp sounds. Tyler Hee Wai, John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu), John Gebbie, and Martin Siderius (Elec. Eng., Portland State Univ., Portland, OR)

Snapping shrimp produce sounds through cavitation bubbles generated by the rapid closing of their claws. These sounds are a primary source of ambient noise in sub-tropical, shallow water environments. Though seasonal and daily variations of snapping shrimp sound levels have been reported, comprehensive studies of short time series have been limited. We investigate the respective spectral characteristics from acoustic arrays recordings at various locations in Oahu, HI, over hourly, daily and weekly periods. Nonlinear time series analysis methods encompassing recurrence plots are used to investigate the acoustic time series together with bivariate and cross recurrence plot analysis with that of tidal and sunlight cycles. Signal entropy and determinism are investigated and transient acoustic ship sounds are located

9:30

4aAB6. The soundscape experienced by the Southern White rhinoceros (*Ceratotherium simum simum*) at a wildlife park conservation center. Suzi Wiseman (Geography, Texas State Univ.-San Marcos, 601 University Dr., San Marcos, TX 78666, sw1210txstate@gmail.com) and Preston S. Wilson (Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Many creatures, including the myopic rhinoceros, depend on hearing and smell even more than on their sight. Noise impacts human health and reproduction, and may impact these other mammals similarly, or even more. Rhinos have been recorded vocalizing infrasonically and sonically. They have a poor breeding record in urban zoos, in which infrasonic noise tends to be chronic. Herd size and composition, the age of potential mates, substrate, enclosure design, and other factors have been studied but little attention if any has been paid to soundscape. As a first step to comparing the soundscapes of facilities in which rhinos have and have not bred successfully, this project recorded and analyzed the soundscape of white rhinos at Fossil Rim Wildlife Center in Texas, one of five Conservation Centers for Species Survival, and one of the few U.S. facilities to successfully breed this species in recent years. Animal responses to sound have been shown to depend on sound level, rate of onset, duration, number of events, spectral distribution of the sound energy, presence of pure tones, the relative level of background noise, and semantics. Similar parameters were analyzed for the data recorded at Fossil Rim and will be presented here.

9:45

4aAB7. An automatic single station multipath ranging technique for 20 Hz fin whale calls: Applications for ocean bottom seismometer data in the northeast Pacific. Michelle Weirathmueller and William S. D. Wilcock (Univ. of Washington, 1503 NE Boat St., Seattle, WA 98105, michw@uw.edu)

Ocean bottom seismometers (OBSs) deployed to monitor earthquakes also record 20 Hz fin whale vocalizations. Because OBSs are often deployed too far apart to record a call on more than one instrument, and call detections are dependent on the acoustic environment, techniques are required to determine call density from a single instrument. One approach is to use point transect distance sampling, which relies on estimates of horizontal range to each call. An automated technique has been developed to estimate range based on the amplitude and relative timing of multipath arrivals assuming a uniform water depth. The method has been evaluated using independently located calls recorded by a seismic network on the Endeavour Segment of the Juan de Fuca Ridge, a region of rough and partially sedimented seafloor. Results suggest that the method works well to ranges of ~5 km but at larger ranges ambiguities arise. Calls at >10 km range are sometimes located between 5 and 10 km because the times are erroneously fit by one too few water column multiples and the amplitudes have too much scatter to be diagnostic of range. The technique is presently being applied in different acoustic settings to determine whether it is more reliable in flat sedimented regions.

10:00–10:15 Break

Invited Papers

10:15

4aAB8. Comparison of soundscapes across the Bering Sea shelf, a biological perspective. Jennifer L. Miksis-Olds (Appl. Res. Lab, Penn State, PO Box 30, M.S. 3510D, State College, PA 16804, jlm91@psu.edu)

Selectively decomposing and visualizing different aspects of an acoustic time series provides a greater understanding of the sources and environmental dynamics contributing to and shaping the temporal and spatial patterns of the measured sound. Source contributions to the overall soundscape vary in space and time, and the biological component is highly dependent on temporal, geographic, and oceanographic factors. The biological contributions to soundscapes along the eastern Bering Sea shelf are tightly coupled to the presence of sea ice and have strong annual patterns. The differences between locations reflect the relative contribution of different species to the

soundscape. Using soundscape plots to display the acoustic environment during the overlap of peak vocal activity by different marine mammal species in the winter/spring season produces a visual representation of acoustic niche partitioning by Arctic species.

10:35

4aAB9. Larval settlement in response to estuarine soundscapes. David Eggleston, Ashlee Lillis, and Del Bohnenstiehl (Dept. of Marine, Earth & Atmospheric Sci., NC State Univ., Raleigh, NC 27695-8208, eggleston@ncsu.edu)

Ambient underwater sound has the potential to be an important orientation and settlement cue for marine invertebrate larvae, yet larval responses to relevant sound patterns are largely unknown. In estuaries of the Southeastern United States, oyster reefs are patchy productive habitats that harbor many soniferous fish and invertebrates, creating distinct sound characteristics. This habitat-related sound could provide a useful cue for the planktonic larvae of obligate reef dwellers and facilitate encounter with suitable settlement substrate. To investigate sound as a settlement cue in this system, larval settlement responses to oyster reef and soft-bottom sounds, as well as a no-sound control were tested for the Eastern oyster, *Crassostrea virginica*. Laboratory and field experiments suggest that sound has a significant effect on oyster settlement rates: higher numbers of larvae settled in the presence of oyster reef sounds than in soft-bottom sound or silent control treatments. Improved understanding of the relationship between habitat sound fields and subsequent larval recruitment is central to bio-physical studies of larval connectivity and recruitment in marine systems.

Contributed Papers

10:55

4aAB10. Locating invertebrate sound sources, including hermit crabs, on shallow water reefs in the Northwestern Hawaiian Islands using an L-shaped array of hydrophones. Simon E. Freeman, Lauren A. Freeman (Scripps Inst. of Oceanogr., 3725 Miramar St. Apt. C, La Jolla, CA 92037, sfreeman@ucsd.edu), Marc O. Lammers (Oceanwide Sci. Inst., Honolulu, HI), and Michael J. Buckingham (Scripps Inst. of Oceanogr., La Jolla, CA)

Using ambient noise for extracting ecological information from coastal waters is a tantalizing idea that has gained momentum with the increasing use of long-term passive recorders. Single-element hydrophone recordings from different reef locations reveal substantial variation in the biologically produced sound field, the spatial scales and sources of which are poorly understood. A seven-element L-shaped array was deployed in a spur-and-groove coral reef environment at Kure Atoll in the Papahanaumokuakea Marine National Monument, Northwestern Hawaiian Islands, in an effort to resolve small-scale spatial variability in reef-generated ambient noise. For each given time step, a fast, pair-wise coherence technique was initially used to estimate correlation in the ambient noise field. Curved-wavefront adaptive beamforming was then performed within range and azimuth windows that encompassed each estimated source location (areas of high correlation). Array processing performance was evaluated and frequency-dependent, spatiotemporal distribution of the sound field derived. The environment surrounding the array was photographically surveyed and ecologically classified through SCUBA-based observations. Survey data were used in an attempt to identify reef structures and the organisms that were the dominant contributors to the local sound field, notably hermit crabs and other hard-shelled invertebrates.

11:10

4aAB11. A disproportion in bowhead whale call counts around an acoustic array during the fall 2012 migration in the Chukchi Sea, Alaska: An investigation into the potential of acoustic masking. Jennifer L. Wladichuk (JASCO Appl. Sci., 2305 – 4464 Markham St., Victoria, BC V8Z 7X8, Canada, jennifer.wladichuk@jasco.com), Julien Delarue, Bruce Martin (JASCO Appl. Sci., Dartmouth, NS, Canada), Xavier Mouy, and Heloise Frouin-Mouy (JASCO Appl. Sci., Victoria, BC, Canada)

A long-term passive acoustic study in the Chukchi Sea monitors the bowhead whale (*Balaena mysticetus*) migration between their summering and wintering grounds in the Beaufort and Bering Seas, respectively. A hexagonal array offshore from Point Lay, AK was located near the center of their migratory corridor in the falls of 2009 through 2011. In 2012 however, there was a large difference in the number of acoustic detections across the array, with call counts close to three times higher in its northwest section compared to its southeast. An increase in vessel traffic in the area resulted in higher ambient noise conditions for much of the summer. This paper investigates the possibility of acoustic masking of calls due to this increase in background noise levels.

11:25

4aAB12. Passive acoustic monitoring of biological and anthropogenic sounds at America's first offshore wind farm. T. Aran Mooney, Maxwell B. Kaplan, Luca Lamoni (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS50, Woods Hole, MA 02543, mkaplan@whoi.edu), Aimee Boucher (Nicholas School of the Environ., Duke Univ., Durham, NC), and Laela S. Sayigh (Biology Dept., Woods Hole Oceanogr. Inst., Woods Hole, MA)

Cape Wind, situated in Nantucket Sound, Massachusetts, is poised to become America's first offshore windfarm. Our objective is to establish baseline (pre-construction) sound levels of human and biological activity, including diel and seasonal variability of various sound types, at the construction site and three nearby comparison sites. Acoustic recorders have been deployed since April 2012, recording on a 10% duty cycle (sample rate: 80 kHz). Biological contributions to the local soundscape are primarily fish sounds, with the dominant signal likely being cusk eel (Family Ophidiidae) calls. These calls, which are composed of stereotyped pulses with an average bout duration of 3.3 $\pm$0.8 s and mean peak frequency of 1030 $\pm$200 Hz, show both seasonal and diel variation. Dense choruses were detected during summer (July), but limited activity occurred in the fall and winter. During vocal periods, detections occurred throughout the day but peaked near dusk. Vessel traffic also showed diel and seasonal trends, with peaks during the daytime and in the summer, which indicates that boat activity can be tracked acoustically. These trends in biological and anthropogenic activity provide key baseline records for evaluating the influence of windfarm construction and operation on a local US soundscape.

Session 4aAO

Acoustical Oceanography and Animal Bioacoustics: Properties, Trends, and Utilization of Ocean Noise I

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Zoi-Heleni Michalopoulou, Cochair
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Chair's Introduction—7:55

Invited Papers

8:00

4aAO1. Ocean sensing using ambient noise: An overview. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

The random nature of ocean ambient noise, natural or man-made, tends to suggest limited utility: ambient noise is commonly considered to be a nuisance for acoustic sensing or imaging applications. However, whether noise is a nuisance or a useful signal actually depends on how it is processed. Indeed, acoustic fields from random sources are often considered to be incoherent, but there is some coherence between two sensors that receive signals from the same individual noise source. Consequently, by cross-correlating the ambient noise recorded at two hydrophones one can estimate the acoustic waves that propagate between them, thus yielding an estimate of the Green's function between those sensors. This passive approach provides a foundation for ocean sensing and imaging techniques using only ambient noise, thus without active transmitters. We will examine the basic background physics of extracting coherent information from ambient noise. Specifically we will highlight among others the role of frequency/bandwidth, noise structure/distribution, ocean fluctuations, sensor separation and, if employed, the array configuration on this process. Further we will give an overview of recent experimental results in shallow and deep water and discuss practical challenges for implementing noise-based ocean sensing techniques.

8:45

4aAO2. Extracting seabed properties using ocean noise measured on an autonomous underwater vehicle mounted array. Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, siderius@pdx.edu), Peter L. Nielsen (STO-CMRE, La Spezia, Italy), and James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Ocean ambient noise has been used to estimate seabed properties such as bottom loss and bottom layering. These ambient noise methods use beamforming and typically collect data on hydrophone arrays of 16 to 32 sensors that are moored or allowed to drift. Compact arrays containing only a few sensors can be mounted to autonomous underwater vehicles (AUVs). This configuration provides better control over the location of the measurement compared to a drifting system. However, compact arrays of this type present a challenge due to limited beamforming capabilities. In July 2012 and June 2013, the Centre for Marine Research and Experimentation conducted the GLASS'12 and GLASS'13 experiments to investigate the possibility of using an AUV equipped with a compact nose array for seabed characterization. The nose-mounted array consisted of a 5-hydrophone vertical and 4-hydrophone tetrahedral array. In this paper, the results from these experiments will be presented along with approaches to seabed characterization using hydrophone arrays with few sensors.

9:05

4aAO3. On the feasibility of estimating attenuation from the coherence of ambient noise. Ravi Menon and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC-0238, UC San Diego, La Jolla, CA 92093, rmenon@ucsd.edu)

There has been a long standing interest, especially in seismology, to be able to reliably retrieve attenuation information from ambient noise. While theoretical work has shown that it is possible to retrieve attenuation from noise, in practice, several factors such as anisotropy of noise distribution and intensity, site amplification factors, etc., make it an extremely challenging task. By assuming a sufficiently uniform noise distribution, researchers have obtained estimates of attenuation by fitting the theoretical noise coherence function modified by an exponentially decaying term, to the observed coherence. In this talk, we present results from analyzing seismic data from the Southern California seismic network in the microseism band (0.05–0.2 Hz) and discuss the feasibility of estimating attenuation from coherence. Specifically, we demonstrate that the assumption of a single wavespeed at each frequency is not valid at certain frequencies. Such variations in wavespeed are often due to inhomogenities in the medium and scattering effects, and it is very likely that interactions between these waves results in a reduction in amplitude due to phase cancellations, which might be erroneously perceived as attenuation. While it might still be possible to estimate attenuation if these interaction effects are accounted for, it raises questions on the validity of the simple exponential decay model.

9:25

4aAO4. Offshore impact pile driving as a source of opportunity for geoacoustic investigations. Gopu R. Potty, James H. Miller, and Huikwan Kim (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, potty@egr.uri.edu)

Noise generated by offshore impact pile driving can radiate into and propagate through the air, water, and sediment. Most of the recent studies have focused on predicting acoustic pressure field in water to assess environmental impacts. In this study, we focus on the propagation of acoustic energy along the interface between water and ocean bottom. We modeled the interface wave propagation using the commercial Finite Element (FE) code, ABAQUS 6.11. A field test is planned for this summer using a scaled model impact pile driving off the dock in the Bay Campus University of Rhode Island (URI). Interface data (particle velocities at the water-sediment interface) will be collected using the Shear Measurement System. Our efforts will focus on identifying the arrivals corresponding to various wave types using data-model comparison. In addition to the interface waves, we will also model the Mach wave front arrival angle. We will explore the possibility of utilizing the arrival times and angles corresponding to these wave types for setting up an inverse problem. This test will also verify the possibility calculating compressional wave speed using a single three-axis geophone by measuring arrival angle of Mach wave front at the interface generated by impact pile driving. [Work sponsored by the Link Foundation Ocean Engineering and Instrumentation PhD Fellowship program.]

9:45

4aAO5. Geoacoustic inversion of noise radiated by surface ships. David P. Knobles, Robert A. Koch, Jason D. Sagers, and Steven A. Stotts (ARL, UT at Austin, 10000 Burnet Rd., Austin, TX 78758, knobles@arlut.utexas.edu)

Addressed is the issue of determining physical properties of the seabed in shallow water using ambient noise from surface ships of opportunity. Both the amplitude and phase of the source spectrum are unknown. The coherent modal interference structure in the received spectra is sensitive to parameters such as the sound speed in the seabed, the speed of the ship, the source-receiver range, the water depth, etc. Such sensitivity allows in some cases for information on these parameters to be extracted from a statistical information approach such as maximum entropy. In shallow water, the rate of decay with propagation range of the received incoherent levels is sensitive to the intrinsic seabed attenuation. For ship of opportunity data, estimating values for the attenuation is complicated by the ambiguity existing between source level and attenuation. This work explores various methods that attempt to resolve the source level and attenuation ambiguity using experimental data taken in the Gulf of Mexico and the New Jersey continental shelf. One approach is to use short-range data to give a prior distribution for the source levels and to then estimate the marginal distributions of the attenuation with longer-range data. [Work supported by ONR.]

10:05–10:20 Break

Contributed Paper

10:20

4aAO6. Radiated ship noise level estimates from measurements in a fjord. Stian Coward (Norwegian Univ. of Sci. and Technol., Trondheim, Norway), Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no), and Hefeng Dong (Norwegian Univ. of Sci. and Technol., Trondheim, Norway)

This paper presents estimates of radiated ship noise levels (20 Hz—1 kHz) due to commercial shipping in the Oslofjord (Norway). Data were recorded at two Networked Intelligent Underwater Sensors (NILUS) units, each equipped with an omni-directional hydrophone placed near the seabed, at water depths of 45 and 110 m, respectively. Received noise levels due to traffic in two nearby shipping lanes (ranges, 0.4–3.0 km) were for these non-ideal test site conditions processed and corrected to monopole source levels by use of the RAM propagation model with range estimates from auxiliary data, environmental input from a prior survey of the area, and a vertically distributed source model [Trevorrow *et al*., J. Acoust. Soc. Am. **124**(2), 767–778] to model uncertain source depth. Estimates of source spectra and of broadband source levels of cargo ships are presented and compared with deep-water measurements from the literature.

Invited Papers

10:35

4aAO7. Models for non-stationary reverberation noise. Leon Cohen (Physics-Hunter, City Univ. of New York, 695 Park Ave., New York, NY 10065, leon.cohen@hunter.cuny.edu)

The random phasor approach to reverberation noise gives Gaussian statistics for intensity and is stationary in time and space. We show that fundamentally different things happen when the noise is defined by a superposition of random propagating pulses that have width. We calculate the nonstationary autocorrelation function of such a model and describe how it evolves in time and space in a dispersive medium. Furthermore, we give criteria for when a nonstationary autocorrelation function is locally stationary in time and/or space. We also consider how and under what conditions the noise evolves to the stationary case given by the ransom phasor sum. [Work supported by ONR.]

10:55

4aAO8. The effect of seawater attenuation on the directionality and spatial coherence of surface-generated ambient noise. Michael J. Buckingham (SIO, UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

Acoustic attenuation in seawater is sufficiently weak that it usually has little effect on the spatial characteristics of ambient noise. However, at frequencies above 10 kHz and depths below 6 km, seawater attenuation may influence the directionality of surface-generated ambient noise. In a semi-infinite, homogeneous ocean, all the noise is downward-traveling, since there are no bottom reflections, and, in the absence of attenuation, the directional density function varies as the cosine of the polar angle measured from the zenith. When attenuation is present, the angular width of this noise lobe becomes narrower, because sound from distant surface sources is attenuated more than acoustic arrivals from overhead. This narrowing effect increases as frequency rises, since the attenuation is a rapidly increasing function of frequency. The spatial coherence of the noise is also modified by the attenuation. For a pair of horizontally (vertically) aligned sensors, the zeroes in the spatial coherence function are shifted to higher (lower) frequencies. These effects of attenuation on the noise field are sufficiently large to be detectable using the recently developed, deep diving instrument platform Deep Sound, which is capable of measuring broadband ambient noise on multiple hydrophones to depths of 11 km. [Research supported by ONR.]

11:15

4aAO9. Ambient noise modeling using sound field reciprocity. David R. Barclay and Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., M.S. #11, Woods Hole, MA 02543, dbarclay@whoi.edu)

An efficient numerical calculation of the vertical and horizontal coherence of the noise field in the ocean due to a distributed sheet of sources can be carried out using a parabolic equation (PE) propagation model and the reciprocity property of the complex pressure field. In the case of an infinite sheet of sources near the surface in an infinitely deep, isovelocity ocean, the calculation of vertical coherence using this method is in agreement with the Cron and Sherman model for wind-driven surface noise, as well as with measurements made of vertical coherence in the Philippine Sea. This numerical model also gives the effective surface-listening radius for a single hydrophone by calculating the contribution to the noise field at the receiver from a small noise patch at the surface, as a function of range. The model is capable of calculating the vertical and horizontal coherence (directionality) of the noise field and the depth-dependent surface-listening radius of a hydrophone in a shallow water waveguide and over range-dependent environments, such as a wedge, canyon, or shelf break. Additionally, this technique can be used to calculate the spatial properties of the noise field due a finite patch of surface sources in motion, such as a rainstorm.

11:35

4aAO10. Computation of ocean noise fields. Michael B. Porter and Laurel J. Henderson (HLS Res., 3366 N. Torrey Pines Ct., Ste 310, La Jolla, CA 92037, mikeporter@hlsresearch.com)

Ships and winds provide two key examples of sound sources that “illuminate” the ocean. In recent years, we have seen a great interest in such noise fields. It has become clear that this noise field provides an important signal that can be used to image the environment and the ocean bottom in particular. Separately, conservationists have become very interested in the fields in terms of their impact on marine life. The modeling of such noise fields provides interesting challenges. First, the noise field is the result of not just a single propagation calculation but a fan of calculations connecting a receiver location to all the noise sources. Second, the characteristic cylindrical spreading from a point source is counter-balanced nearly perfectly by the increase in area of a cylindrical annulus as we accumulate the contributions of noise sources in range. Thus, the noise can travel to extremely long range. This talk will discuss such noise calculations in terms of both the numerics and the physics. We will discuss numerical options in terms of efficiency and accuracy. In addition, we will discuss the role of environmental uncertainties.

Session 4aBA

Biomedical Acoustics and Physical Acoustics: Field Characterization and Dosimetry for Therapeutic Ultrasound Applications I

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Gail ter Haar, Cochair
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Chair's Introduction—7:55

Invited Papers

8:00

4aBA1. Challenges in the characterization of high intensity therapeutic ultrasound devices and fields and regulatory guidance development. Subha Maruvada and Gerald R. Harris (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993, subha.maruvada@fda.hhs.gov)

As part of its regulatory responsibility for medical devices, the US Food and Drug Administration (FDA), Center for Devices and Radiological Health, reviews pre-clinical and, if necessary, clinical data submitted by device applicants relevant to safety, effectiveness, and accurate labeling. For HITU devices, pre-clinical testing can include ultrasound power measurement, pressure/intensity mapping, temperature field characterization (*in vitro* and *in vivo*), acoustic and thermal simulations, and demonstration of the targeting accuracy and treatment monitoring. To assist manufacturers and third parties in the submission process, the FDA provides pre-market submission guidance that often relies on technical standards and specifications such as published by the International Electrotechnical Commission (IEC). IEC documents for HITU are currently nearing completion. However, even with these documents, technical challenges remain because of the focused, large amplitude pressure fields encountered in HITU. Measurement and modeling issues include using radiation force balances at HITU power levels, the need for complex deconvolution with some hydrophone types, validation of simulation models, tissue-mimicking material development for temperature measurements, and cavitation detection and quantification. These challenges along with current FDA guidance efforts will be discussed.

8:20

4aBA2. Standards development and execution for therapeutic ultrasound applications. Mark E. Schafer (Sonic Tech, Inc., 23 Brookline Court, Ambler, PA 19002-1904, marks@sonictech.com)

Therapeutic ultrasound holds great promise in the treatment of a number of different disease states and patient conditions. In bringing these devices to market, the two main considerations are naturally safety and efficacy. Central to the safety evaluation of such devices is the determination of potential hazards through the use of international standards. The International Electrotechnical Commission (IEC), has several standards and initiatives in the area of therapeutic ultrasound, both in the measurement of ultrasonic fields and in safety characterization. To be effective, these standards must not only be technically accurate, but also sufficiently practical that their primary users, namely manufacturers, can properly implement them. This talk reviews the development of standards related to therapeutic ultrasound and also discusses the implementation approaches among different manufacturers. For instance, while diagnostic ultrasound systems can be safely managed with only type-testing, the energy levels from therapeutic systems require that every system be evaluated or calibrated before patient use. Typically, systems undergo detailed sub-system testing, with only limited testing performed on the final product at full energy levels. Assumptions with regard to output linearity are also used to further minimize the difficulties implicit in measurements at high ultrasonic energies.

8:40

4aBA3. Just doing our DUTy!—An international project on dosimetry for ultrasound therapy. Adam Shaw (Acoust. and Ionizing Radiation Div., National Physical Lab., Hampton Rd., Teddington, Middlesex TW11 0LW, United Kingdom, adam.shaw@npl.co.uk)

This presentation will describe a large international project which aims to develop the metrological infrastructure for the determination of ultrasound exposure and dose to tissue. Ultimately this will improve treatment planning and risk assessment. The 3-year project started in June 2012. It is co-ordinated by the UK National Physical Laboratory and is funded in part by the European Metrology Research Programme of the European Union. The other project partners are INRIM (IT), PTB (DE), UME (TR), CSIC (ES), ICR (UK), Moscow State University (RU), NIM (CN), and University of Merseburg (DE). The scientific work is divided into six Work Packages: —Quantities and definitions will review dose and *in situ* exposure quantities proposing alternatives where necessary; —Laboratory dosimetry standards will develop reference methods for thermal and non-thermal therapeutic dose parameters; —Dose modeling and

validation will develop theoretical and computer modelling procedures to calculate acoustic and thermal dose quantities with well understood uncertainties; —Intercomparison of methods will formally compare measurement methods and models developed; —Dosimetry Transfer Standards will develop phantoms and test systems for the assessment of clinical ultrasound equipment, transferring laboratory standards to end-users; —Application to clinical treatment will provide a direct link between the methods developed and the clinical use of therapeutic ultrasound with the long-term aim of improving treatment planning.

9:00

4aBA4. Dosimetry for therapy ultrasound. Gail ter Haar (Phys., Inst. of Cancer Res., Royal Marsden Hospital, Downs Rd., Sutton SM25PT, United Kingdom, gail,terhaar@icr.ac.uk)

Ultrasound is probably the only clinical therapy modality that does not as yet have a well established dosimetric parameter. There are a number of reasons for this. Unlike other energy forms, ultrasound produces cell killing by two distinct principle mechanisms: cavitation and heating. While dose parameters have been proposed for each mechanism, there has been no attempt to combine these into a single unit. Existing dosimetric methods will be reviewed and put into the context of other therapeutic methods. Biological and physical end points will be discussed.

9:20

4aBA5. Addressing nonlinear propagation effects in characterization of high intensity focused ultrasound fields and prediction of thermal and mechanical bioeffects in tissue. Vera A. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, vera@apl.washington.edu), Petr V. Yuldashev (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Wayne Kreider, Oleg A. Sapozhnikov, Michael R. Bailey, Tatiana D. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), and Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA)

Nonlinear propagation effects are present in most fields generated by high intensity focused ultrasound (HIFU) sources. In some newer HIFU applications, these effects are strong enough to result in the formation of high amplitude shocks that actually determine the therapy and provide a means for imaging. However, there is no standard approach yet accepted to address these effects. Here, a set of combined measurement and modeling methods to characterize nonlinear HIFU fields in water and predict acoustic pressures in tissue is presented. A characterization method includes linear acoustic holography measurements to set a boundary condition to the model and nonlinear acoustic simulations in water for increasing pressure levels at the source. A derating method to determine nonlinear focal fields with shocks in situ is based on the scaling of the source pressure for data obtained in water to compensate for attenuation losses in tissue. The accuracy of the methods is verified by comparing the results with hydrophone and time-to-boil measurements. Major effects associated with the formation of shocks are overviewed. A set of metrics for determining thermal and mechanical bioeffects is introduced and application of the proposed tools to strongly nonlinear HIFU applications is discussed. [Work supported by NIH EB007643, T32 DK007779, and RFBR 13-02-00183.]

Contributed Papers

9:40

4aBA6. Holography and numerical projection methods for characterizing the three-dimensional acoustic fields of arrays in continuous-wave and transient regimes. Wayne Kreider (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Adam D. Maxwell (Dept. of Urology, School of Medicine, Univ. of Washington, Seattle, WA), Petr V. Yuldashev (Faculty of Phys., Moscow State Univ., Moscow, Russian Federation), Bryan W. Cunitz, Barbrina Dunmire (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov, and Vera A. Khokhlova (Faculty of Phys., Moscow State Univ., Moscow, Russian Federation)

The use of projection methods is increasingly accepted as a standard way of characterizing the 3D fields generated by medical ultrasound sources. When combined with hydrophone measurements of pressure amplitude and phase over a surface transverse to the wave propagation, numerical projection can be used to reconstruct 3D fields that account for operational details and imperfections of the source. Here, we use holography measurements to characterize the fields generated by two array transducers with different geometries and modes of operation. First, a seven-element, high-power therapy transducer is characterized in the continuous-wave regime using holography measurements and nonlinear forward-projection calculations. Second, a C5-2 imaging probe (Philips Healthcare) with 128 elements is characterized in the transient regime using holography measurements and linear projection calculations. Results from the numerical projections for both sources are compared with independent hydrophone measurements of select waveforms, including shocked focal waveforms for the therapy transducer. Accurate 3D field representations have been confirmed, though a notable sensitivity to hydrophone calibrations is revealed. Uncertainties associated with this approach are discussed toward the development of holography measurements combined with numerical projections as a standard metrological tool. [Work supported by NIH EB007643, P01DK043881, R01DK092197 and T32DK007779, NSBRI through NASA NCC 9-58, and RFBR 13-02-00183.]

9:55

4aBA7. High pressure focused ultrasound field characterization. Baki Karaböce (Ultrasound Lab., TUBİTAK UME, Beşevler, Gebze, Kocaeli 41470, Turkey, baki.karaboce@tubitak.gov.tr), Ali Şahin (Phys. Dept., İnönü Univ., Malatya, Turkey), Eyüp Bilgiç, Enver Sadıkoğlu (Ultrasound Lab., TUBİTAK UME, Gebze, Turkey), and Süreyya Nur (SHMYO, İnönü Univ., Malatya, Turkey)

High intensity focused ultrasound (HIFU) transducers are novel and very attractive tools for cancer therapy. They produce very high pressure acoustic fields up to tens of MPa at the focus; thus, acoustic characterization of HIFU fields must be investigated in order to ensure the safe and effective use in clinical applications. A needle hydrophone has been used for HIFU transducer characterization in the newly designed home made system at TÜBİTAK UME (The Scientific and Technological Research Council of Turkey, the National Metrology Institute) Ultrasound laboratory. TÜBİTAK UME HIFU measurement system was controlled by a LABVIEW based data translation program. The driving signal was generated by signal generation card and output from the hydrophone was fed directly into a DSO oscilloscope card. The controlling program executed a "capture-analyze-move" cycle, allowing a large number of measurements to be made. Field scanning measurements are made between 1 MPa, 2 MPa, and 3 MPa pressures. Theoretical model for the beams of periodic waves with an initially uniform amplitude distribution was also performed, based on the Khokhlov-Zabolotskaya-Kuznetsov equation. Numerical solutions were compared with the experimental data and found to be in agreement.

10:10–10:30 Break

10:30

4aBA8. Feasibility of using infra-red system for absolute calibration of high intensity focused ultrasound fields. Svetlana Shmeleva (Dept. of Medical Phys., Moscow State Univ., Leninskie gori 1/2, Moscow 119991, Russian Federation, sveta@acs366.phys.msu.ru), Vera Khokhlova (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), Leonid Gavrilov (N.N. Andreyev Acoust. Inst., Moscow, Russian Federation), Eleanor Martin, Neelaksh Sadhoo, and Adam Shaw (Acoust. and Ionising Radiation Div., National Physical Lab., Teddington, United Kingdom)

In recent years, considerable progress has been achieved in the use of infrared (IR) method for measuring acoustic fields of high intensity focused ultrasound (HIFU) transducers. The principle of the method is to determine the intensity distribution at the surface of an acoustic absorber by measuring the distribution of the initial rate of temperature increase at the start of insonation. Here, the method is extended to estimate the absolute values of intensity in ultrasound fields of HIFU transducers. The approach compares the temperature rise measured at the surface of a thin absorber using an IR camera and the pressure distribution measured in water using a hydrophone. The measurements were carried out for two focused HIFU transducers and a flat physiotherapy transducer. During the IR measurement, the transducer was driven in tone burst mode to avoid interference from reflected acoustic waves between the absorber/air interface and the transducer. The method enables fast quantitative estimations of intensity distributions that agree well with hydrophone measurements and gives reasonably consistent results for medical transducers of different geometries. [Work was supported in parts by RFBR 12-02-31388, 12-02-00028, 13-02-00183; NIH 2R01EB007643-05, the European Metrology Research Programme (Joint Research Project HLT03) and the UK National Measurement Office.]

10:45

4aBA9. Sensitivity and power handling capacity of the Ultrasound Imparted Air-recoil Resonance method for acoustic power estimation. Sreekumar Kaiplavil (Inst. of Cancer Res., 15 Cotswold Rd., Sutton SM2 5NG, United Kingdom, sreekumarkaiplavil@yahoo.co.in)

Recently, we have introduced a novel method, the Ultrasound Imparted Air-recoil Resonance (UIAR), for the estimation of acoustic power with high sensitivity. Salient features of this approach over existing practices include fast response, electrical and magnetic inertness and hence MRI compatibility, portability, high damage threshold, and immunity to vibration and interferences, low cost, etc. The angle of incidence should be fixed for accurate measurement. However, the transducer-detector pair can be aligned in any direction with respect to the force of gravity. In this sense, the operation of the device is orientation-independent. The device is useful in the case of pulsed/burst as well as continuous ultrasound exposure. Sensitivity was found to be extendable down to the micro Watt range or even below that, however, critical issues related to the thermo-viscous loss mechanisms in the system need careful optimization. The power handling capacity is a few hundreds of Watts, which could be augmented using suitable materials for the Helmholtz resonator window that functions as the acoustic filed sensing head. A detailed account of the sensitivity and high-power estimation capability of the UIAR technique will be presented in the paper.

11:00

4aBA10. Algebraic reconstruction technique considering curved ray for sound-speed tomography with ring-array transducer. Hirofumi Nakamura, Tetsuya Kanagawa (Dept. of Mech. Eng., The Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, nakamura@fel.t.u-tokyo.ac.jp), Satoshi Tamano (Hitachi Aloka Medical, Mitaka, Japan), Takashi Azuma (Dept. of Mech. Eng., The Univ. of Tokyo, Bunkyo-ku, Japan), Kiyoshi Yoshinaka (Dept. of Human Life Technol., Adv. Industrial Sci. and Technol., Tsukuba, Japan), Akira Sasaki, Shu Takagi, and Yoichiro Matsumoto (Dept. of Mech. Eng., The Univ. of Tokyo, Bunkyo-ku, Japan)

Our objective is to develop an ultrasound treatment and diagnosis integrated system for breast cancer. Ultrasound computed tomography (UCT) in imaging and high intensity focused ultrasound (HIFU) in therapy was integrated to achieve ideal treatment system. Profiles of sound speed and attenuation obtained by UCT has informative parameters to correct deformation of HIFU beam. We try to develop an imaging system using ring-array transducer with 1024-elements, multiplexer connecting 1024 to 256 and Verasonics programmable imaging system with 256 channels. First, an iterative Simultaneous Algebraic Reconstruction Technique (SART) reconstruction methods with an assumption of straight path was employed. SART was applied to projection data calculated by a FEM simulator treating actual curved ray caused by tissue inhomogeneity. In these results, estimated error in sound speed difference was 53%. Then we introduced Linear Travel-time Interpolation (LTI) to SART to implement effects of curved ray. LTI is a ray tracing method based on Fermat's Principle. We evaluated the LTI and SART integrated method for sound-speed tomography. Travel-time for reconstruction was achieved from simulation based on finite difference method. The reconstruction image was highly corresponded with original image. Reconstructed image from experimental projection data will be reported in the presentation.

11:15

4aBA11. Albumin based gel phantoms with controllable thermal sensitivity for quantifying ultrasound thermal energy. Rei Asami and Kenichi Kawabata (Central Res. Lab., Hitachi, Ltd., 1-280 HIgashi Koigakubo, Kokubunji, Tokyo 185-0003, Japan, rei.asami.fq@hitachi.com)

Medical ultrasound safety is mainly concerned with two factors, thermal and mechanical. A simple method to indicate temperature rise by ultrasound is essential to standardize diagnostic and therapeutic techniques. A thermocouple is used for direct measurement but the effect of viscous heating is inevitable. MR thermometry provides accurate measurements but requires expensive machinery. Bovine serum albumin (BSA) gels are known to change colors from transparent to opaque at 70 °C and used as simple temperature indicator. We propose a new temperature indicator that combines additives with BSA gels to modify the denaturation temperature. Polyacrylamide gels were prepared with 9% BSA and various concentrations of additives such as ovarian albumin and gelatin. Gels were then placed in degassed water at 37 °C and adjacent to a focused ultrasound transducer ($f0 = 3.3$ MHz). Ultrasound irradiation process was video-recorded for image processing. Gels prepared with ovarian albumin resulted in ranges of denaturation temperature of 40—60 °C. Ovarian albumin, hardly soluble protein, could have lowered the stability of BSA. Recorded timing of the denaturation fitted well with the computer based simulation result. The result suggests that additives can widen the application of BSA gels in quantifying ultrasound thermal energy. [This study was, in part, funded by Japanese Menistory of Economy, Trade and Inductor.]

Session 4aEA

Engineering Acoustics: Energy Harvesting From Acoustic Phenomena

Kenneth Cunefare, Chair
Georgia Tech., Mech. Eng., Atlanta, GA 30332-0405

Invited Papers

8:30

4aEA1. Low-power electricity generation from dynamical systems. Alper Erturk (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, alper.erturk@me.gatech.edu)

This talk will review our research on energy harvesting from electroelastic dynamical systems for low-power electricity generation with an emphasis on piezoelectric transduction. The transformation of vibrations into electricity using piezoelectric materials with the goal of powering small electronic components has received growing attention over the last decade. Enabling energy-autonomous small electronic components can lead to reduced maintenance costs in various wireless applications, such as structural health monitoring of civil and military systems. After a brief discussion of energy harvesting methods for low-power electricity generation, this talk will be focused on linear and nonlinear energy harvesting using piezoelectric materials through the topics of distributed-parameter electroelastic dynamics of energy harvesters, performance and frequency bandwidth enhancement by exploiting nonlinear dynamic phenomena, deterministic and stochastic excitation of monostable and bistable configurations, effects of dissipative and inherent electroelastic nonlinearities, electroaeroelastic flow energy harvesting using airfoil-based and bluff body-based configurations, and enhanced harvesting of structure-borne propagating waves using elastoacoustic mirrors and metamaterial structures. A brief introduction to our efforts on multifunctional underwater thrust and power generation using flexible piezoelectric composites will also be given.

8:50

4aEA2. Energy harvesting from acoustic sources. Kenneth Cunefare (Georgia Tech., Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

While energy harvesting from a variety of ambient sources (vibration, light, and wind) has been demonstrated and sensing and communication applications to exploit those sources have been developed, acoustic energy as an ambient source has not received much attention. The reason for this comes down to the basic physics of how much energy is available within an acoustic field. For airborne sounds, the energy density in sound fields that are perceived by humans to be quite loud (e.g., 80 to 160 dB, or ~0.2 Pa to ~2 kPa) actually represent an extremely low available energy source. In consequence, means must be taken to intensify an acoustic response, for example, through resonance, but even so, available energy remains limited. The exception to this issue in airborne sounds is the sound field that exists inside of an operating jet aircraft engine. The situation is quite different, however, when one considers pumped and pressurized fluid systems, where acoustic pressure variations due to the operation of pumps and other devices may reach into the mega-Pascal (MPa) range. Energy harvesting from such a fluid-borne acoustic source is feasible for powering sensors and wireless communication systems and has been successfully demonstrated.

9:10

4aEA3. Aeroacoustic applications of acoustic energy harvesting. Stephen B. Horowitz (Emerging Technologies Group, Ducommun Miltec, 678 Discovery Dr., Huntsville, AL 35806, shorowitz@one.ducommun.com) and Mark Sheplak (Dept. of Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL)

In this paper, the development and use of acoustic energy harvesting technology as a source of local power for aeroacoustic sensing and control applications is discussed. As an example, the application of acoustic energy harvesting as a primary local power source for aircraft engine noise reduction technology is addressed. Noise generated in turbofan engine nacelles can easily exceed 150 dBSPL, presenting a primary motivation for aircraft noise reduction technologies. Adaptive noise control approaches require less power than active methods and can outperform passive techniques (e.g., by actively tuning an otherwise passive system to a changing noise spectrum). Locally sourced power is highly desirable in this application to eliminate cabling in a difficult to access, harsh environment. The low power requirements can be reasonably supplied with harvested acoustic energy, particularly given the large acoustic intensities in and around aircraft engine nacelles. The detailed development approach and experimental results of acoustic energy harvesting using an electromechanical Helmholtz resonator will be presented. Additionally, alternative Helmholtz resonator variants and other aeroacoustic applications of acoustic energy harvesting will be reviewed.

Contributed Papers

9:30

4aEA4. Coupling efficiency analysis of hydraulic pressure energy harvesters. Ellen Skow, Kenneth Cunefare, and Alper Erturk (Georgia Inst. of Technol., 620 Peachtree St. NE, Unit No. 1012, Atlanta, GA 30308, eskow3@gatech.edu)

The acoustic pressure within hydraulic systems, referred to as pressure ripple, is a high intensity energy source that can be utilized for powering sensor networks. A section of such a system can be modeled as a one dimensional waveguide, where the intensity can reach up to 1000 mW/cm^2 from a 300 kPa pressure ripple (peak-to-peak acoustic pressure) within a hydraulic system. Hydraulic pressure energy harvesters (HPEH) are devices designed to convert the pressure ripple into electrical energy, thereby enabling wireless sensor nodes. HPEH couple the dynamic fluid pressure to a piezoelectric stack, which is connected to a harvester circuit to optimize power output. A key aspect of the HPEH design is the fluid-mechanical coupling of the pressure ripple to the stack for maximizing the energy extracted. The efficiency of HPEH device and harvester circuit potential power output can be determined using the volume velocity of the pressure ripple, the coupling efficiency of the HPEH, and the conversion efficiency of the piezoelectric stack. In this work, the coupling efficiency and the power output efficiency of currently developed HPEH devices will be analyzed and compared to modeled efficiency of such devices.

9:45

4aEA5. An effective theory for meta-mass and meta-material mechanical/electrical devices. John J. McCoy (The Catholic Univ. of America, 1922 New Hampshire Ave., Washington, DC 20009, mccoy@cua.edu)

A dynamical system comprised of a rigid housing element that encapsulates a large multiplicity of oscillators is said to comprise a "meta-mass," in that there is no direct interaction of the external environment and the oscillators, the presence of which is inferred in an observed non-Newtonian housing element behavior. For a class of meta-masses, for which the oscillators have more-or-less equal masses with resonances that densely fill a frequency band more-or-less uniformly, the non-Newtonian behavior is most transparently seen in a different "energetics," i.e., the transfer of energy between the housing element and an external source, this occasioned by an inherently "transitory "internal energy transfer process. Encapsulating a large multiplicity of meta-masses in an ordinary material specimen is said to comprise a meta-material specimen, a somewhat different understanding in not having the direct interaction of each individual oscillator with every other individual oscillator obtain through the behavior of the material matrix. Constructed is a mathematical framework for quantifying limited observable system behavior while not accommodating all underlying physics, i.e., an "effective theory," this a prerequisite for the design of meta-mass and meta-material mechanical/electrical devices.

THURSDAY MORNING, 5 DECEMBER 2013 — SUTTER A/B, 9:30 A.M. TO 11:00 A.M.

Session 4aID

Interdisciplinary: Oral History Bootcamp

Victor Sparrow, Chair
Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802

In this bootcamp, participants will learn appropriate techniques to conduct oral history interviews. No advanced reservations are required. Dr. Gregory Good is the Spencer Weart Director of the Center for the History of Physics of the American Institute of Physics (AIP), and he will coach the session participants in the nuts and bolts of preparing for, conducting, and following up after an oral history interview session. Dr. Good is very experienced with collecting oral histories.

If you are interested in the history of acoustics and in preserving that history, the ASA Committee on Archives and History invites you to participate in this bootcamp. Oral histories are a very important part of documenting the background and motivations for administrative and scientific contributions, the part of history that is not usually available in the printed record, such as peer-reviewed publications. So oral histories fill the gaps on why someone dedicated much of their professional life to a particular topic or describes the journey they traveled to reach notable goals and/or make lasting contributions to the field.

Your help is needed to preserve this history, the history of acoustics. Thanks for participating!

Session 4aMU

Musical Acoustics and Structural Acoustics and Vibration: Acoustics of Percussion Instruments I

Thomas Moore, Chair
Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789

Invited Papers

9:00

4aMU1. Percussion, performance, pedagogy, and technology: The impact of virtual music Lessons in percussion education. Rohan Krishnamurthy (Musicology, Eastman School of Music, 544 Sunrise Circle, Kalamazoo, MI 49009, rohan.krishnamurthy@rochester.edu)

Real-time net-conferencing software such as Skype and Google Hangout has revolutionized music education in the last decade. This presentation examines the intersections of percussion, performance, pedagogy, and technology with specific reference to South Indian classical Carnatic percussion. Drawing on several scholarly perspectives and ethnographic research in India, various aspects of online teaching will be discussed, including its history and motivations, comparison of online and face-to-face pedagogical repertoire, negotiating audio/video lag or latency, and other significant social and cultural dimensions. Broader implications of online pedagogy in terms of reinforcing musical and intellectual centers, as well as strategies for improving the online learning experience, will be explored. A live, interactive performance on the mridangam, the principal percussion instrument from South India, will follow the presentation.

9:20

4aMU2. The contribution of spectrum and tempo to auditory streaming of simple and compound "bols" in tabla rhythms. Punita G. Singh (Sound Sense, 16 Gauri Apt., 3 Rajesh Pilot Ln., New Delhi 110011, India, punita@gmail.com)

Rhythms on the tabla, a north Indian percussion instrument, are generated by producing sounds on one or two drums separately or simultaneously to produce simple or compound "bols." At high speeds, auditory stream segregation based on spectral properties of adjacent bols can create parallel perceptual layers that can be leveraged strategically by percussionists. This observed phenomenon was studied experimentally by constructing sequences of bols in which adjacent sounds shared different spectral regions. For example, the bols "ghe" and "tin," which have very different spectra, were placed on either side of the compound bol "dhin," which contains both "ghe" and "tin." At a moderate tempo, the sequence is typically heard as a gallop rhythm. However, at quicker tempi, streaming occurs and components of the compound bol "dhin" group perceptually with their neighboring counterparts, to create parallel layers of pairs of "ghe," "ghe" and "tin," "tin", instead of the galloping triple. Listeners identified when streaming took place as a function of the specific bols and tempi used. Spectral analyses of the bols indicated that perceptual segregation was indeed based on proximity of spectral loci. At high speeds, spectral differences are perceptually highlighted and facilitate the formation of auditory streams.

9:40

4aMU3. Measurements of coupled drumhead vibrations using electronic speckle-pattern interferometry. Randy Worland and Benjamin Boe (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Many musical drums such as snare drums, tom toms, and bass drums consist of two membranes at opposite ends of a cylindrical shell. Striking one head causes both to vibrate, as their motions are coupled due to the enclosed air as well as the shell itself. An optical system consisting of two electronic speckle-pattern interferometers was constructed allowing operating deflection shapes of both heads to be viewed simultaneously while the drum is driven acoustically at a resonant frequency. This system allows the determination of the relative phases, orientations, and amplitudes of the vibrational patterns on the two heads. Previously reported results for coupled drumheads were verified and extended to include the effects of degenerate single membrane mode pairs that are split due to non-uniform tension in the heads. Examples of higher frequency coupled patterns are also shown. Parameters influencing the degree of coupling are described briefly and compared qualitatively with results from a finite element model.

10:00

4aMU4. Expressively actuated percussion instruments and interfaces. Edgar J. Berdahl (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu)

Percussion instruments and interfaces can be extended by incorporating actuation with motors. While many prior designs have employed actuation to simply trigger notes, this work emphasizes the importance of the musician's gesture using expressively controlled actuators. This is achieved by implementing force-feedback control of the actuators. For example, the Haptic Drum essentially consists of a drum pad attached to a woofer. Every time that a drumstick strikes the drum pad, the woofer briefly pushes it upward again, adding energy to the motion of the drumstick. In this manner, the Haptic Drum enables musicians to play one-handed drum rolls at optionally superhuman speeds and with arbitrarily complex dynamics. Furthermore, because the musician and the drumstick are inside the feedback control loop, the musician can change the rate of the drum roll by modulating the stiffness of his or her muscles or changing the downward force applied to the drumstick. Conversely, borrowing on technology for remote surgery, expressively actuated percussion instruments can be designed using force-feedback teleoperation. Simply by recording the position of the master haptic device during teleoperation, generalized motor programs can be created for expressively playing percussion instruments. This technology was employed in the composition "When the Robots Get Loose."

Contributed Papers

10:20

4aMU5. Teponaztli, an ancient percussion instrument. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

Ancient percussion instruments in Mesoamerica included the Huehuetl and the Teponaztli, both made out of a single piece of hollow trees, which pretty often were decorated with low profile representations of animals, warriors or other ceremonial symbols, both were often used by the Aztecs and Tarascas among many other cultures in the region. The first one is normally used in a vertical position and has an strained animal skin, typically an ocelot, in the top side in order to excite it with wooden sticks or by the bare hands, while the second is formed by a couple of tongues made out from the same tree cortex, and excited by a couple of wooden sticks covered with hard rubber. In both the hollowed tree forms a resonant chamber in order to increase the sound level, and include openings in order to allow the exit of the sound from inside of the drum. They were employed for ceremonies, festivities and communication. Here, there are presented the sound characteristics of a small "teponaztli," which is actually a small copy scaled about 5:1 of a real one, which can usually only be seen in archaeological museums.

10:35

4aMU6. Acoustical measurements and finite difference simulation of the West-African "talking drum". Florian Pfeifle (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstrasse 13, Hamburg 20354, Germany, Florian.Pfeifle@uni-hamburg.de)

A very iconic drum, commonly found in West-Africa, is the so called "talking drum." It has a large geographic distribution, spreading several countries and ethnic groups. The characteristic feature of this unique instruments is the membrane, which can be tuned, through varying the tension of the membrane fixture, while playing. This effect can be utilized to play melodies and pitch-glides on the drum. West-African musicians use this effect to mimic speech patterns and speech melodies from tonal languages. This highly non-linear excitation of the drum leads to several questions regarding the acoustical properties of the instrument. The acoustical research of the drum include high-speed measurements of the drum-head to quantify the changing tension distribution over the membrane. Microphone array measurements, with 128 microphones, are applied to examine the radiation characteristics of the instrument. A special focus is put on the resonance frequencies of the enclosed air volume and the influence it has on the top and bottom membrane as well as the damping characteristics of the drum head. A physical model of the instrument, including a Kirchhoff-Carrier-like tension modulated, 2-dimensional, orthotropic, stiff membrane, coupled to a 3-dimensional air-volume is implemented and compared to the measurements.

10:50

4aMU7. Acoustic properties of carbon fiber in percussive instruments. Alex Wion, Rustin Vogt, and Patrick Homen (Mech. Eng., California State Univ. Sacramento, 1255 University Ave., Apt. 117, Sacramento, CA 95825, wion.07@gmail.com)

With advancements in manufacturability and increasing accessibility of materials, composite materials are competing with traditional materials in the design and manufacturing of musical instruments. Specifically, in snare drum applications, carbon fiber encased in an epoxy matrix has been chosen for its physical properties of strength, durability, and weight as well as its psychoacoustic properties compared to wood. Through comparative data acquisition analysis, the acoustic properties of composite snare drum shells are examined against traditional wooden snare shells. By subjecting the shells to controlled audio frequencies, the sound energy response on the air medium outside of the shell was measured and compared to wooden shells. Experimental data were examined was compared to theoretical values found for carbon fiber.

THURSDAY MORNING, 5 DECEMBER 2013 — CONTINENTAL 9, 8:00 A.M. TO 9:45 A.M.

Session 4aNS

Noise: Outdoor Sound Propagation and Modeling

Lauren M. Ronsse, Chair
Audio Arts and Acoust., Columbia College Chicago, 33 E. Congress Pkwy, Ste. 601, Chicago, IL 60605

Contributed Papers

8:00

4aNS1. Comparison and evaluation of physics-based outdoor sound propagation assessment schemes. Lauren M. Ronsse (Audio Arts and Acoust., Columbia College Chicago, 33 E. Congress Pkwy, Ste. 601, Chicago, IL 60605, lronsse@colum.edu) and Dan Valente (Construction Eng. Res. Lab., Engineer Res. and Development Ctr., Champaign, IL)

Evaluating the effectiveness of outdoor sound propagation assessment schemes using experimentally collected data is a complex task. Specifically, when the acoustic data collected assume non-Gaussian distributions with differing sample sizes per assigned class, traditional parametric statistical techniques may not be used. As an alternative, this research introduces an original cost function to evaluate and compare various methods aimed at defining acoustic propagation classes based on meteorological measurements. Class assignments in each scheme are based on the atmospheric stability, the strength of the vertical effective sound speed gradient, and the vertical effective sound speed profile. The acoustic data included in this analysis were generated by a high-energy impulsive source and gathered at source-to-receiver distances of 1, 2, 4, and 8 km. The results indicate that the assessment scheme based on the strength of the effective sound speed gradient most effectively classifies the peak level sound propagation sampled in temperate and desert climate conditions at these distances.

8:15

4aNS2. Using simplified terrain and weather mapping in outdoor sound propagation predictions. Whitney Coyle, Victor Sparrow (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, wlc5061@psu.edu), and Bruce Ikelheimer (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

A complementary experimental and computational study was undertaken to assess the variability due to model sensitivities when predicting propagation in realistic outdoor environments by using a Green's Function Parabolic Equation (GFPE) method and including realistic weather and terrain profiles. In order to test the validity of the basic and enhanced model including real terrain and weather data, field measurements were conducted at Hogan's Mountain, North Carolina. By incorporating USGS terrain and measured weather profiles, this project further developed a Hogan's Mountain specific GFPE to include a stair-step terrain mapping capability and linearly interpolated sound speed profiles. This Hogan's Mountain GFPE uses matching conditions for comparison to the measured received levels from the field test data. Due to the vast data set acquired, which allows for comparison of the model to measurement in a multitude of propagation situations, several comments are made regarding the choice of calculation parameters as well as the validity of the basic and altered Hogan's Mountain GFPE for different propagation prediction settings. Sample results will illustrate the agreement based on frequency, specific terrain and weather measured along the propagation path for individual source-receiver pairs of interest. [Work supported by Spawar Systems Center Pacific.]

8:30

4aNS3. Data-driven prediction of peak sound levels at long range using sparse, ground-level meteorological measurements and a random forest. Dan Valente (US Army Engineer Res. & Development Ctr., PO Box 9005, Champaign, IL 61826, daniel.p.valente@usace.army.mil)

Outdoor sound propagation is highly dependent upon meteorological conditions. While this, of course, is a trivial statement, predicting sound levels based on meteorology is not. This is especially true for signals that propagate many kilometers, as is the case for those generated by high-energy impulsive sources such as explosions and heavy weaponry; waves have ample opportunity for refraction by and scattering from local atmospheric features along the entire propagation path. The range of received blast levels at distances greater than 2 km can span nearly 50 dB, depending on weather conditions. Using a statistical learning method known as a Random Forest, we demonstrate the prediction of levels from simple meteorological measurements in the face of this extreme variability. With simple, spatially sparse meteorological data, the model can predict levels to within 3 dB at 2 km and 5 dB at 15 km. The results presented here suggest that as more data are acquired through continuous noise monitoring programs, physics-blind, data-driven statistical models have the potential to supplant computationally intensive propagation models for noise prediction. Caveats and cautions when using these types of machine learning methods will also be discussed.

8:45

4aNS4. Variability in acoustic transmission loss over a rough water surface. Cristina Tollefsen and Sean Pecknold (Defence Res. and Development Canada - Atlantic, P. O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, cristina.tollefsen@gmail.com)

Variability in the acoustic transmission loss of impulsive sounds propagating in the atmosphere over a rough water surface was measured over time scales of 1–2 h at a fixed range of 250 m. On two separate days (26 and 30 May 2012), a propane cannon source was deployed on the upper deck of a ship. Once per minute, the propane cannon fired volleys of four shots that were recorded on a receiver deployed on a small boat tethered upwind of the ship. Meteorological conditions and sea state were comparable on both days, resulting in similar observations for transmission loss: mean and standard deviation of 64 $\pm$ 5 dB (26 May) and 66 dB $\pm$ 4 dB (30 May). The variability in transmission loss was high, with minimum and maximum values of 48 and 75 dB (26 May) and 53 and 77 dB (30 May). The transmission loss measured throughout the experiment exceeded the 48 dB predicted by assuming spherical spreading, since the receiver was upwind of the source. Measured results are compared to transmission loss computed from a parabolic equation model using an ensemble of turbulence and rough sea surfaces estimated from the meteorological conditions on board the ship.

9:00

4aNS5. Mapping the extent of noise on a national scale using geospatial models. Dan Mennitt, Kurt Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, daniel_mennitt@partner.nps.gov), Lisa Nelson (Inventory and Monitoring Div., National Park Service, Fort Collins, CO), and Megan McKenna (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

Because many wildlife habitats, geological processes, and anthropogenic impacts occur on a regional scale, acoustical analyses must encompass a similar extent. Geospatial sound modeling incorporates spatial representations of biological, geophysical, climatic, and anthropogenic factors to assess expected contributions to the existing sound pressure level from both anthropogenic and natural sources. This method enables mapping of sound pressure levels at national scales. The models do not directly apply the physics of sound propagation or characteristics of individual sound sources. Instead, long-term sound pressure level measurements from hundreds of sites across the contiguous United States were used to train regression models to predict acoustic conditions. This talk will focus on the implications of the resulting acoustic maps of the contiguous United States. In addition, noise and the relationship to light pollution will be considered.

9:15

4aNS6. Intensity analysis of peak-frequency region in noise produced by a military jet aircraft. Trevor A. Stout, Kent L. Gee, Tracianne B. Neilsen, Alan T. Wall (Phys., Brigham Young Univ., 688 North 500 East, Provo, UT 84606, titorep@gmail.com), and Michael M. James (Blue Ridge Res. and Consulting LLC, Asheville, NC)

Acoustic intensity measurements of the F-22A Raptor are analyzed as part of ongoing efforts to characterize the noise radiation from military jet aircraft. Data were recorded from a rig of microphones and an attached tetrahedral intensity probe at various locations to the sideline and aft of the aircraft. Recently, techniques such as coherence, similarity spectra analyses, and near-field acoustical holography have indicated a peak-frequency region comprised of two maxima that have very different radiation directionalities. Acoustic vector intensity is analyzed as a function of frequency to further assess the behavior of this double-peak phenomenon, which is not accounted for by current jet noise models. The results thus far confirm the discrete nature of the peaks and their directionalities.

9:30

4aNS7. Source motion modeling for high-speed aircraft noise propagation. Bao N. Tong and Kai Ming Li (School of Mech. Eng., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2031, bntong@purdue.edu)

A continuous source motion model has been developed to represent a cruising aircraft traveling at high altitudes. The numerical implementation is based on the Lorentz transform (LT) and a Fast Field Program (FFP) formulation, which is used to compute the sound fields due to a monopole point source traversing in a horizontally stratified atmosphere parallel to a ground surface. To reduce the computational expenses, a one-dimensional LT-FFP is performed to obtain the pressure time-history for overhead flight conditions. The continuous source motion model is compared against a heuristic model which applies a Doppler shift to the stationary source sound field. A linear sound speed profile was selected to simplify the ray model implementation. A parametric study involving the source Mach number and source emission frequency has been performed in a variety of environmental conditions. The differences in the predicted maximum sound pressure levels between the two models can be as large as 9 dB under certain conditions. Numerical simulations indicate that low source emission frequencies (e.g., 50–300 Hz) combined with high source Mach numbers tend to result in larger discrepancies. The numerical simulations suggest the importance of including the effects of convective source amplification, especially, for turboprop aircraft noise propagation.

Session 4aPA

Physical Acoustics: Advances in Infrasound Research I

Roger M. Waxler, Cochair
NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677

Daniel Kleinert, Cochair
National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677

Invited Papers

8:00

4aPA1. Infrasonic wind noise in a pine forest. Richard Raspet, Jeremy Webster, and JohnPaul Abbott (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, raspet@olemiss.edu)

Comparison of the ambient infrasonic noise levels at 39 stations of the International Monitoring System indicated that wind noise levels are as much as 6 dB quieter at heavily vegetated stations. This combined experimental and theoretical study investigates mechanisms for wind noise generation in and above a uniform pine forest (h ~7.0 m). A 10 m tower instrumented with ultrasonic anemometers measures the turbulence field and wind velocity profile above and below the canopy and an infrasonic transducer is collocated on the ground. A prediction of the turbulence-shear wind noise contribution from the turbulence above the canopy and also from the turbulence within the canopy are calculated and compared to the data. The above canopy contribution agrees well with the data for the low frequency regime (<0.5Hz). Surprisingly, this contribution is larger than would occur over a grassy plane with similar meteorological conditions. The under canopy turbulence-shear interaction calculation underpredicts the small high frequency contribution, however an estimate of the turbulence-turbulence interaction wind noise in the under canopy layer provides a good fit to this data. [Work supported by the U. S. Army Research Laboratory and the U. S. Army Research Office under grant W911NF-12-1-0547.]

8:20

4aPA2. New results exploiting correlation in wind noise to enhance detection of transient infrasound. William G. Frazier (Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, frazier@olemiss.edu)

A previous ASA presentation introduced a signal processing method for exploiting the correlation observed in pressure fluctuations measured by infrasound sensors on small spatial-temporal scales in order to achieve enhanced signal detection and estimation performance. An algorithm based on representing the pressure field with Fourier series and finite-element bases was presented and applied to time series data in the 0.1 to 1.0 Hz band with inter-sensor spacing of 1 m using small domes as wind screens. The coherence of the data was not inspiring, but the performance was consistent with this low coherence. In this presentation new data collected using flush-to-the-earth installed sensors spaced 1 and 0.5 m apart will be presented and analyzed. In addition, application of a generalized least-squares approach to estimating the infrasound signal will be examined and compared to the previous method. Results demonstrate significant improvements over previous results at frequencies below 1.0 Hz.

8:40

4aPA3. Calibration and characterization of the response of infrasound sensors to environmental factors. Carrick L. Talmadge (NCPA, Univ. of Mississippi, 1 Coliseum Dr., Unviersity, MS 38677, clt@olemiss.edu)

An infrasound calibration system has been developed at the National Center for Physical Acoustics. The calibration tank is comprised of a 1 in. cylindrical shell 40-in. in diameter, 40-in. long, with 40 in. hemispherical end caps. The interior volume of the tank is approximately 1.8 cubic meters. Up to eight normal-sized infrasound sensors can be enclosed in the volume for one measurement session. Pressure and temperature in the interior and exterior of the tank are also monitored. The pressure source is a 10-in. driver, which is calibrated using multiple barometers that are placed internal to the tank. The measurement paradigm is to drive the tank at a constant amplitude (typically 10-Pa peak-to-peak) and known frequency (typically 0.5 Hz), and to track the variation in the measured response of the test sensor with respect to multiple reference sensors in order to characterize the effect of manipulation of the environment (static pressure, temperature, and seismic motion) on the test sensor. For sensors with very low nonlinearity, allowing the static pressure to change while the driving amplitude of the speaker was held constant was found to be necessary for assessing the effect of nonlinearity on these sensors. Variations in sensitivity on the order of 50 ppm can be measured in a 10-s interval.

9:00

4aPA4. Passive acoustic remote sensing and anomalous infrasound propagation studies. Jelle D. Assink (DAM/DIF/DASE, CEA, Bruyeres-le-Chatel, Arpajon F-91297, France, jelle.assink@gmail.com), Roger Waxler (NCPA, Univ. of Mississippi, University, MS), Laslo Evers, Pieter Smets (KNMI, De Bilt, Netherlands), Alexis Le Pichon, and Elisabeth Blanc (DAM/DIF/DASE, CEA, Bruyeres-le-Chatel, France)

In this talk, we will present recent work on various infrasound remote sensing studies. We will focus on bi-directional stratospheric ducting during a Sudden Stratospheric Warming (SSW) event and the associated infrasonic signature. We present infrasound data in which the described effect is captured with microbarom signals in the Mediterranean region. Microbarom source locations are modeled using operational ocean wave models. The modeling reveals a previously unidentified microbarom source region in the Eastern Mediterranean besides the more typical microbarom source region in the Atlantic Ocean. This work illustrates that the classic paradigm of a unidirectional stratospheric duct for infrasound propagation can be broken during a SSW event. Furthermore, we will present a case study in which the influence of atmospheric dynamics on infrasound propagation is studied. We make use of over 6 years of nearly continuous volcanic infrasound recordings from Mount Etna, Italy (37 N) that are available through the Atmospheric dynamics Research InfraStructure in Europe (ARISE) network. The infrasound observables are compared to theoretical estimates obtained from propagation modeling using existing European Centre for Medium-Range Weather Forecasts (ECMWF) atmospheric databases. While a good agreement is often found, we also report on significant discrepancies around the equinox period and during intervals during which anomalous detections occur during the winter.

Contributed Papers

9:20

4aPA5. Refraction of impulsive signals by a mountain slope. Roger M. Waxler (NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu) and Doru Velea (SAIC, Reston, VA)

As part of a recent experiment on signals from blast waves, it was possible to instrument a mountain slope to the east of the source locations. Four infrasound sensors were deployed about 1 km apart from the base of the mountain up. In addition, infrasound sensors were deployed close to the mountain peak, and then along a ridge extending to the east at roughly constant altitude. It was observed that signals developed a long, low frequency tail as they propagated up the slope. Data and theoretical analyses will be presented.

9:35

4aPA6. Ground and aerostat measurements of wind noise in the infrasonic range. W. C. Kirkpatrick Alberts (US Army Research Lab., 2800 Powder Mill, Adelphi, MD 20783, kirkalberts@verizon.net), Roger Waxler (Univ. of Mississippi, University, MS), Christian Reiff, and Leng Sim (US Army Research Lab., Adelphi, MD)

Conventional deployment of infrasound sensors typically requires the sensors to be on the ground with extensive wind screens, e.g., porous hose or pipe arrays, in order to minimize wind induced noise on the sensor. Thus, flying an infrasound sensor on a balloon would seem ill-advised because of the inability to carry sufficient wind screens to mitigate the increased winds aloft. However, during an experiment designed to monitor short-range characteristics of low-frequency impulsive events, an infrasound sensor was placed aboard a tethered aerostat while the balloon flew to a maximum height of approximately 300 m. Comparisons between noise levels at the airborne sensor and at a nearby ground based sensor will be discussed. Noise levels aboard the aerostat are found to be similar to those on the ground.

9:50

4aPA7. Direct measurement of acoustic impedance for wind-noise-reduction pipe systems and components. Chad M. Smith and Thomas B. Gabrielson (Appl. Res. Lab., Penn State Univ., PO Box 30, State College, PA, cms561@psu.edu)

A wind-noise-reduction pipe system can have a significant effect on the frequency response of an infrasound array element especially above a few tenths of a hertz. If there is a defect in the pipe system—a clogged resonance suppressor, a blocked or broken pipe, or a flooded manifold, for example—measurement of the frequency response may indicate the presence of the defect but not necessarily the type or location. Direct measurement of the acoustical impedance made at accessible points in the pipe system may be a useful adjunct to *in-situ* response measurement for identification of the type and location of defects, which may expedite repair especially if most of the system is buried. A prototype of a portable impedance instrument has been constructed and tested both in the lab and under field conditions. Measurements of individual pipe-system components—inlets, pipes, manifolds, resonance suppressors, or gravel piles—are shown along with comparison to theoretical predictions. In addition, defects are intentionally introduced in a rosette pipe system to determine the sensitivity of the acoustical impedance measurement to such defects.

10:05–10:20 Break

10:20

4aPA8. Experimental investigation of large porous wind fences for infrasonic wind noise reduction. JohnPaul R. Abbott, Richard Raspet, and Jeremy Webster (Dept. of Phys. and Astronomy, National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., Rm. 1044, Oxford, MS 38677, johnpaul.abbott@gmail.com)

An extensive experimental investigation of large porous wind fences constructed from commercially available materials has recently been completed. Measured changes to the wind noise, turbulence, and wind velocity profile inside the fence resulting from varying the fence's height, diameter, and porosity have been measured. The effect of other variables, including porous roofs, a bottom gap, and the addition of secondary windscreens were also studied. An empirical model based on measurements of the turbulence correlation length around the outside of the fence and the velocity gradient across the fence's surface has been derived to develop a better understanding of the wind noise reduction mechanisms. The measurements and model are then used to suggest an optimized wind fence design and predict the band width and magnitude of the wind noise reduction curve.

10:35

4aPA9. On the interaction of infrasonic waves with internal gravity waves perturbations. Jean-Marie Lalande (Jamie Whitten National Ctr. for Physical Acoust., NCPA, 1 Coliseum Dr., University, MS-38677, jean-marie.lalande@gmail.com), Roger Waxler, and Joel Lonzaga (Jamie Whitten National Ctr. for Physical Acoust., NCPA, Oxford, MS)

Infrasonic waves propagate at long range through atmospheric ducts resulting from the stratification of atmospheric properties. These ducts are characterized by their spatio-temporal variability. Hence, infrasonic waves integrate atmospheric information along their propagation paths. In order to study infrasonic wave propagation, we resort to atmospheric specification combining Numerical Weather Prediction and climatological models. However, due to the lack of observations, these models fail to describe small scale variability such as perturbations associated to the presence of internal gravity waves. These waves play an important role in the atmospheric dynamic by transferring momentum to the mean flow at critical levels and at wave-breaking altitudes. In this study we intend to describe the interaction of infrasonic waves with internal gravity waves in order to understand the long-tail behavior observed in infrasound broadband signals. We developed a model for the propagation of internal waves used to generate realistic perturbations of the background atmospheric states. By using a linear full-wave model of infrasound propagation, our goal is to ultimately relate infrasound characteristics to internal waves properties.

10:50

4aPA10. Preliminary study of infrasonic attenuation and dispersion in the lower thermosphere, based on non-continuum fluid mechanics.
Akinjide Akintunde and Andi Petculescu (Univ. of Louisiana at Lafayette, PO Box 44210, Lafayette, LA 70504, akin@louisiana.edu)

A framework for predicting thermospheric attenuation and dispersion of infrasound between 80 and 160 km will be presented. The work is part of a pilot study whose goal is to complement the currently established models of thermospheric propagation, the standard being the Sutherland-Bass (SB) model [J. Acoust. Soc. Am. **115**(3), 1012–1032 (2004)], which overestimates the observed attenuation noticeably. Based on the Navier-Stokes equation and its associated momentum and heat fluxes, the SB model treats the higher atmosphere as a continuum. However, for a given wavelength, the Knudsen (Kn) number increases rapidly in the thermosphere due to the high mean free path gradient. For $Kn > 0.01$, the continuum Navier-Stokes equations are no longer accurate. The present work shows how the predicted wavenumber changes when non-continuum approximations (e.g., the Burnett and 13-moment equations) are used. The ambient parameters and thermophysical properties are extracted from NIST, at the partial pressures of the main constituents (N_2 and O_2). The effects of rotational relaxation, gravity and tidal waves, neutral-charged and charged-charged particle interactions, UV heating/cooling rates, and other thermospheric processes will be addressed. [The work was funded by NSF-EPSCoR/Louisiana Board of Regents.]

11:05

4aPA11. An exact solution of a Burgers' equation governing the nonlinear propagation of infrasound in a range-dependent, windy atmosphere.
Joel B. Lonzaga and Roger Waxler (National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, jblonzag@olemiss.edu)

We present the derivation of an exact solution of an inviscid Burgers' equation that governs the nonlinear propagation of infrasound in a range-dependent, windy atmosphere with arbitrary sound speed, wind speed, and density profiles. The Burgers' equation is a reduced form of a nonlinear transport equation obtained using weakly, nonlinear ray theory. The solution extends known solutions for a homogeneous, windless medium to include the effects of wind and the inhomogeneity in the properties of the atmosphere. Analytical expressions for the shock velocity and period lengthening are readily obtained from the solution. The exact solution is used to validate our numerical algorithm which uses a spectral method to integrate the Burgers' equation. These models are compared with observed infrasound arrivals that clearly demonstrate nonlinear pulse steepening and stretching while propagating in the upper atmosphere.

Session 4aPP

Psychological and Physiological Acoustics: The Ear Club: Honoring Ervin R. Hafter and His Contributions to the Study of Binaural Processing and Auditory Cognition I

Neil F. Viemeister, Chair
Dept. of Psych., Univ. of Minnesota, Minneapolis, MN 55455

Chair's Introduction—8:55

Invited Papers

9:05

4aPP1. Selective attention and auditory filters. Robert S. Schlauch (Speech Lang. Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, 115 Shevlin Hall, Minneapolis, MN 55455, schla001@umn.edu)

Erv Hafter has made significant contributions to our understanding of role of selective attention in detection tasks. I was fortunate to learn from and collaborate with Professor Hafter and his students on some projects addressing this topic as a post-doctoral fellow in his laboratory. We used the probe-signal method to measure auditory filters to learn about the allocation of listening bands under various cuing conditions. Our experiments demonstrated that auditory filters are under a listener's cognitive control. Our results also helped to explain why large losses are not observed in simple, tonal frequency uncertainty experiments. In Schlauch and Hafter (1991), an analysis of filter widths as the number of cue tones (monitored bands) increased supported an earlier finding by Hafter and Kaplan (1976) that auditory filters widen as the cost of shared attention increases. More recently, in a study inspired by my time in Hafter's lab, I completed an informational masking experiment that revealed individual differences in the liability of auditory filters that explains the poor performance of some persons in an informational masking task.

9:25

4aPP2. Auditory informational masking and the Ear Club connection. Robert Lutfi (Commun. Sci. and Disord., Univ. of Wisconsin, 1410 E. Skyline Dr., Madison, WI 53705, ralutfi@wisc.edu)

Erv Hafter has always made a special effort to promote recognition of young researchers in our field. His Ear Club of 40-plus years has provided a conversant audience for new investigators to test new ideas and his attendant hospitality has helped forge relationships valued throughout one's career. In this talk, I will describe how my own work benefited from visits with Erv. The focus is on informational masking (IM). I will present recent work suggesting a strong connection between two major, but seemingly unrelated, factors associated with IM: masker uncertainty and target-masker similarity. Experiments involving multitone pattern discrimination, multitalker word recognition, sound-source identification, and sound localization are described. In each case, standard manipulations of masker uncertainty and target-masker similarity (including the covariation of target-masker frequencies) are found to have the same effect on performance provided they produce the same change in the information divergence of target and masker, a measure of statistical separation between signals from information theory. The results seem to reflect a general perceptual principle that segregates signals based on differences in their statistical structure. Future plans are to test the generality of this result in a simulated-cocktail-party environment [Hafter *et al*. Intl. Symp. Hearing, in press].

9:45

4aPP3. Encoding of amplitude modulation upon interference in remote frequency regions. Yi Shen and Virginia M. Richards (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697, shen.yi@uci.edu)

The modulation detection threshold for a sinusoidally amplitude modulated target tone was measured in the presence of another amplitude modulated interference tone. The interferer had a carrier frequency of 700 Hz and a modulation rate of 40 Hz. The target carrier frequency was 1300 Hz, and its modulation rate was 23, 33, 43, 63, and 80 Hz in separate conditions. During the psychophysical experiment, the auditory frequency following response (FFR) phase-locked to the 700-Hz carrier and the auditory steady-state response (ASSR) phase locked to the 40-Hz modulator were recorded. It was hypothesized that when greater interference occurs (when the interferer and target had closer modulation rates), the neural representation to the interferer might be enhanced or suppressed to a lesser degree. Neither the FFR nor ASSR demonstrated dependency on the target modulation rate, and no significant correlation was found between the behavioral and these two types of physiological measures. Results suggest the possibility that the modulation detection interference might originate from a location that is more central to the neural structures that give rise to FFR and the 40-Hz ASSR.

10:05

4aPP4. Division of processing resources in auditory judgments. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), Anne-Marie Bonnel, and Ervin R. Hafter (Dept. of Psych., Univ. of California, Berkeley, Berkeley, CA)

The attention operating characteristic (AOC) displays joint performance in a dual-task paradigm. Sampling theory allows the AOC to be used to distinguish two tasks that share resources from two tasks that call upon independent resources. Work in the Hafter lab over the past twenty years has examined the division of resources from several different perspectives. This talk will review data on intensity discrimination and identification showing that "easier" tasks can cause dual-task costs, while "harder" tasks can have no costs associated with the dual-task. Recent data on intensity discrimination and identification will be examined in the context of the division of processing resources, showing that the costs of sharing resources can increase with age of the listener. Finally, suggestions will be made for ways in which the current enthusiasm for the dual-task paradigm in clinical research can be used to improve our theoretical understanding of attention and memory as well as demonstrating the effects of various impairments and prostheses.

10:25–10:40 Break

10:40

4aPP5. Revisiting Hafter and De Maio: How precision of coding of interaural delay varies with both magnitude of interaural delay and center frequency. Leslie R. Bernstein and Constantine Trahiotis (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., MC3401, 263 Farmington Ave., Farmington, CT 06030, Les@neuron.uchc.edu)

Erv Hafter's contributions span a broad range of topics fundamental to the understanding of monaural and binaural auditory perception. His publications and presentations are typified by historically important empirical, conceptual, and theoretical analyses. One issue he studied concerned how precision of neural coding of interaural time delay (ITD) varies with the magnitude of the delay and the spectral content of the stimulus. Such information remains central to contemporary quantitative models of binaural processing. Extending Erv's efforts, we have developed a new way to measure precision of ITD-coding as a joint function of ITD-magnitude and center frequency. The novel twist entails transforming the classic $NoS\pi$ condition into $(NoS\pi)\tau$ by imposing an ITD on the entire signal-plus-masker waveform. With that stimulus, uniform internal compensation of external ITDs would yield thresholds both independent of ITD and equal to those obtained under $NoS\pi$. In accord with the results of Hafter and De Maio [J. Acoust. Soc. Am. **57**, 181–187 (1975)], however, thresholds increased with ITD and did so more rapidly at 4 kHz than at 500 Hz. Our data were accounted for by assuming an internal, interaurally uncorrelated "processing noise," the power of which increases exponentially with the magnitude of the internal, "matching," delay.

11:00

4aPP6. Lateralization, discrimination, detection, and Ervin. Richard M. Stern (Dept. of Elec. and Comput. Eng., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, rms@cs.cmu.edu)

Erv Hafter's early work focused on the development of a comprehensive understanding of the binaural system, combining a characterization of the subjective phenomena associated with binaural lateralization with the results of objective experiments measuring interaural discrimination and binaural detection. To a large extent, this author's attempts to characterize a broad range of auditory phenomena based on a Jeffress-Colburn cross-correlation-based model can be viewed as a reformulation of the stimulus-based lateralization model that Hafter developed to provide a unified theoretical framework for his own findings. Erv Hafter has a gift both for designing clever and revealing experimental stimuli and paradigms, as well as for developing provocative interpretations of his experimental results. This paper will discuss a selection of complex binaural phenomena revealed by Erv's research in the context of their impact on contemporary theories of binaural interaction, as well as their impact on computational models of speech processing for robust speech recognition. [Work supported by DARPA and Cisco Research.]

11:20

4aPP7. On the temporal weighting of binaural cues: precedence effects, rate limitations, and binaural adaptation. G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232-8242, g.christopher.stecker@vanderbilt.edu), Andrew D. Brown (Physiol. and Biophys., Univ. of Colorado School of Medicine, Aurora, CO), Anna C. Diedeisch (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., Nashville, TN), and Jacqueline M. Bibee (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

Over the course of three decades, Erv Hafter and his colleagues have investigated the relative effectiveness of binaural cues carried by different temporal portions of brief, mainly, high-frequency, stimuli. Those studies pioneered the use of filtered impulse trains (avoiding many limitations of sinusoidal amplitude modulation, and presaging the use of "transposed tones" for delivering envelope ITD), and revealed rate-limited processing of post-onset cues (Hafter and Dye 1983). Careful modeling of the effects on both ITD and ILD later led Hafter *et al.* (1988) to propose a pre-binaural origin for this binaural adaptation, and Hafter and Buell (1990) to present a simple quantitative model of monotonic, onset-triggered reduction of cue effectiveness over time. Erv's later students adopted observer-weighting paradigms to more directly measure the influence of each click in a train (Saberi 1996, Stecker and Hafter 2001), revealing some important and cue-dependent non-monotonicites. Our own work has continued this line of research, most recently investigating the temporal weighting of binaural cues in lower-frequency sounds. Consistent with a large body of Hafter's work, the results reveal an important temporal asymmetry in the effectiveness of binaural cues at both low and high frequencies and a key role for pre-binaural mechanisms. [Work supported by R03DC009482, R01DC011548.]

11:40

4aPP8. Temporal effects when localizing targets defined by spatial consistency: Relation to Hafter's work on "binaural adaptation". Robert H. Gilkey (Dept. of Psych., Wright State Univ., Dayton, OH 45435, gilk@wright.edu), Brian D. Simpson (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH), Eric R. Thompson (Ball Aerosp. & Technologies Corp., Fairborn, OH), Nandini Iyer, and Grffin Romigh (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH)

Simpson *et al.* [Proceedings of Meetings on Acoustics **19**, 050140 (2013)] measured localization accuracy for sequences of noise bursts masked by simultaneous sequences of noise burst maskers (1 or 2). Targets were distinguishable from the maskers only in that all bursts within a target sequence arose from the same location, whereas each burst in a masker sequence arose from a different, randomly selected location. In this task, the spatial information extracted from each burst both helps define which element is the target ("target identification") and where that target is located ("target localization"). Presumably, the listener is more focused on target identification early in the sequence and on target localization later in the sequence. And so, it is not surprising that localization accuracy increases dramatically as the length of the sequences is increased. This talk will compare these results to the foundational work of Hafter and his colleagues [e.g., Hafter, 1997, in *Binaural and Spatial Hearing in Real and Virtual Environments*, edited by R. H. Gilkey and T. R. Anderson (Erlbaum, Mahwah, NJ), pp. 211–232] investigating changes in the ability to extract binaural information during temporal sequences of sounds.

THURSDAY MORNING, 5 DECEMBER 2013 POWELL, 10:00 A.M. TO 11:35 A.M.

Session 4aSA

Structural Acoustics and Vibration, Architectural Acoustics, and Noise: Human-Induced Vibration in Buildings

Benjamin M. Shafer, Chair
Building Acoust., Conestoga-Rovers & Associates, Inc., 1117 Tacoma Ave. South, Tacoma, WA 98402

Chair's Introduction—10:00

Invited Papers

10:05

4aSA1. Prediction and measurement of floor response due to walking. Thomas M. Murray (Civil Eng., Virginia Tech, 537 Wisteria Dr., Radford, VA 24141, thmurray@vt.edu) and Brad Davis (Civil Eng., Univ. of Kentucky, Lexington, KY)

The procedure for determine if a floor design will result in annoying vibrations due to walking in the American Institute of Steel Construction Design Guide 11 Floor Vibrations due to Human Activity is based on single degree of freedom response with a number of assumptions which reflect actual occupant conditions. Understanding the development of the procedure is necessary when comparing the results of field testing of problem floors. Development of the procedure, its accuracy when compared to a large data base of tested floors, and comments on appropriate testing protocols are presented.

10:25

4aSA2. Floor vibration testing in hospital operating rooms. Anthony Nash (Charles M. Salter Associates, 130 Sutter St., Ste. 500, San Francisco, CA 94104, anthony.nash@cmsalter.com)

Operating rooms in several hospitals have been studied for their *in-situ* floor vibration properties. The motivations ranged from identifying the cause of unstable images observed in a surgical microscope to determining whether a portion of a long span floor in a hospital parking garage could be adapted for a future surgical center. As part of these studies, several types of floor excitation techniques were employed including a vibration exciter, a "heeldrop," and a human walker. The input magnitudes from the first two of these excitation sources can be measured at a given location using a calibrated force plate. The human walker, however, is a "wild card" source since the location of the footfall is constantly changing; moreover, the nature of the applied force pulse is unknown under field conditions. Thus, one cannot easily quantify the dynamic force of the test signal that is most germane to the assessment of floor vibration in the field. The paper will review the literature governing floor vibration limits in sensitive buildings and summarize the utility of information obtained by various floor testing protocols.

10:45

4aSA3. Improvement of footfall vibration of concrete floors: Two case studies. Michael Gendreau and Hal Amick (Colin Gordon Associates, 150 North Hill Dr., Ste. 15, Brisbane, CA 94005, mgendreau@colingordon.com)

It is sometimes necessary to improve the footfall performance of an existing concrete floor in a manufacturing facility, as is the case when the vibration criterion changes because of new technology. The paper examines two case studies in which the floors were modified simply by placement of an additional concrete slab atop the existing slab. This approach has some drawbacks, most notably the additional weight, which must be accommodated by the available framing and foundation capacity. Measured data are presented which illustrate the changes in structural properties and footfall performance.

Contributed Papers

11:05

4aSA4. Investigation on floor impact noise difference between Rhamen and wall-column constructions using vibration analysis. Dukyoung Hwang, Sinyoeb Lee, and Junhong Park (Hanyang Univ., Haengdang 1-dong, Seongdong-gu, Seoul 133-791, South Korea, dyhwang@hanyang.ac.kr)

Floor impact noise generation depends on the configuration of building structures. Especially, the Rhamen construction is known to radiate less than the wall-column construction. In this study, the sound radiation mechanisms between the two different structures are compared through experimentation using laboratory setup of the scaled model. The transfer of vibration energy from external impact source is calculated and compared with the measured results. Parameters affecting the radiated sound energy are determined and evaluated. The in and out-of phase vibrations of the building floors resulted in tonal sound radiation at two-closely located frequencies which resulted in modulated floor impact sound. This modulation increased the annoyance on the residents. Eventually, a design method to efficiently reduce the floor impact sound is proposed.

11:20

4aSA6. Dissipative effects in the response of an elastic medium to a localized force. Douglas Photiadis (NRL, 4555 Overlook Ave. SW, Washington, DC 20375, douglas.photiadis@nrl.navy.mil)

The effect of dissipation on the real part of the admittance of an elastic half-space is typically thought to be unimportant if the loss factor of the elastic medium is small. However, dissipation induces losses in the near field of the source and, provided the size of the source is small enough, this phenomenon can be more important than elastic wave radiation. Such losses give rise to a fundamental limit in the quality factor of an oscillator attached to a substrate. Near field losses associated with strains in the elastic substrate can actually be larger than intrinsic losses in the oscillator itself if the internal friction of the substrate is larger than the internal friction of the oscillator. [This research was sponsored by the Office of Naval Research.]

THURSDAY MORNING, 5 DECEMBER 2013 — PLAZA B, 8:30 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Speech Production I

Bryan Gick, Chair

Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T1Z4, Canada

Contributed Papers

8:30

4aSC1. A comparison of speech errors elicited by sentences and alternating repetitive tongue twisters. Stefanie Shattuck-Hufnagel (Speech Commun., MIT, 36-511 MIT, 77 Mass Ave., Cambridge, MA 02139, sshuf@mit.edu), Cathy Bai (Wellesley College, Wellesley, MA), Mark Tiede (Haskins Labs., New Haven, CT), Argyro Katsikis (Univ. of Potsdam, Potsdam, Poland), Marianne Pouplier (Univ. of Munich, Munich, Germany), and Louis Goldstein (USC, Los Angeles, CA)

Sound-level errors collected by ear from continuous communicative speech have been interpreted as mis-selections of planning elements, which are then produced fluently without residue of the original target (Lashley 1957, Fromkin 1972, Garrett 1975, Shattuck-Hufnagel 1982). In contrast, articulatory measures of tongue twister errors reveal gestural intrusions: target and intrusion elements are co-produced, sometimes resulting in a gestural error which is imperceptible to listeners (Pouplier 2003, Goldstein *et al.* 2007; see also Mowrey and MacKay 1970). Is this apparent difference due to structure and processing differences between the two utterance types, i.e., sentences (e.g., The top cop saw a cop top) vs alternating repetitive word lists (e.g., top cop top cop top cop) generally produced with quasi-periodic timing? Or, do articulatory measures simply capture the nature of sound-level errors more accurately? We elicited errors using both types of stimuli in the same experimental session; perceptual and acoustic analyses show that sentences provoke more apparent whole-segment substitutions (e.g., /tap/ for /kap/), while alternating repetitive lists provoke more errors with two onset bursts (e.g., /tkap/), resembling gestural intrusions. This suggests that there may be more than one mechanism underlying spoken errors, and that different materials may engage these mechanisms to different degrees.

8:45

4aSC2. An artificial neural network model for serial speech production and speech error simulation. Erin Rusaw (Univ. of Southern California, 1150 Stanford St., Apt. 1, Santa Monica, CA 90403, ecrusaw@gmail.com)

This paper presents serial speech gesture encoding and recall extension to the Neural Oscillator Model Speech Timing and Rhythm (NOMSTR), previously used to simulate prosodic and syllable timing (Rusaw 2010, 2013), and describes how it can be used to simulate speech errors and

speech error patterns. This model uses NOMSTR to provide the phonological context vector which drives its serial recall, activating a series of gesture output nodes. Besides the prosodic context signal, the gesture output nodes have two other sources of input: (1) Nuclear gestures nodes provide excitatory input to the nodes required to produce the onset and coda gestures they share a syllable with; this mechanism is the ANN instantiation of the nucleus-centered syllable model. (2) Second, nuclear gesture nodes receive excitatory input from NOMSTR's syllable-level thresholded node, which ties the timing of the gestures in a syllable to the time at which the syllable occurs in the phonological output. This extended version of NOMSTR is shown to able to simulate aspects of speech error behavior which other models have been unable to explain, such as C/V error (O-C/N) asymmetry, and error dependence (Rusaw and Cole 2009). [Work supported by NSF and NIH.]

9:00

4aSC3. Simulations of sound change resulting from a production-recovery loop. Benjamin Parrell (Dept. of Linguist., Univ. of Southern California,GFS 301, Los Angeles, CA 90089, parrell@usc.edu), Adam Lammert (Signal Anal. & Interpretation Lab., Univ. of Southern California, Los Angles, CA), Shrikanth Narayanan (Signal Anal. & Interpretation Lab., Univ. of Southern California, Los Angeles, CA), and Louis Goldstein (Linguist., Univ. of Southern California, Los Angeles, CA)

We present a computational model of lenition-based sound change. Speech production targets for constriction degree are modeled by differential equations with a single stable fixed point at the target constriction degree that interact with higher order equations that reflect prosodically conditioned variation. This output is then input to the articulatory-to-acoustic forward map, the quantal nature of which causes continuous variation in constriction degree to result in two outcomes: closure or spirantization. The quantized acoustic outcomes are then used as the input for the language learner, who must recover the produced constriction degree in the ambient language environment to produce the speech unit appropriately. We model this recovery as estimating the parameters of an articulatory distribution that is most likely to have produced the acoustic observations (i.e., the maximum likelihood estimate). In particular, we optimize the parameters of a Gaussian distribution with respect to their log-likelihood function, given the acoustic data that results from applying the quantal forward map to that distribution. We show how this production-recovery loop may lead to sound change in a language by varying the nature of the articulatory-to-acoustic map as well as the amount of knowledge the learner has about both the prosodic structure and articulatory-to-acoustic map.

9:15

4aSC4. Feedback-driven corrective movements in speech in the absence of altered feedback. Caroline A. Niziolek, Srikantan S. Nagarajan, and John F. Houde (Univ. of California, San Francisco, 513 Parnassus Ave., Rm. S-362, Box 0628, San Francisco, CA 94118, cniziolek@ohns.ucsf.edu)

Altered auditory feedback often evokes a compensatory vocal response in speakers, providing evidence that errors in speech production can be rapidly corrected online. To assess whether the same compensatory mechanism is employed in natural, unaltered speech, we carried out an acoustic analysis characterizing formant movement during single speaking trials. Subjects produced 200 repetitions each of three different monosyllabic words in the MEG scanner and, separately, in the presence of varying background noise levels. To assess corrective responses, we compared the centricity of formant values at the beginning to that at the middle of each trial. In all subjects, we found strong evidence of vowel "centering"—that is, a corrective movement mid-utterance that caused utterances at the periphery to move closer to the center of the formant distribution. Across subjects, the magnitude of vowel centering was correlated with auditory cortical suppression preceding the corrective movement, suggesting that the suppression may serve as a neural mechanism for error detection and correction. These findings suggest that less-prototypical utterances, which make up a large proportion of natural speech, are processed as potential errors, and that feedback-driven speech error correction is occurring constantly on a small scale.

9:30

4aSC5. Coarticulation as an epiphenomenon of syllable-synchronized target approximation—Evidence from fundamental frequency aligned formant trajectories in Mandarin. Hong Gao (English Lang. and Lit., Sichuan Univ., Chengdu, China) and Yi Xu (Speech, Hearing and Phonetic Sci., Univ. College London, Chandler House, 2 Wakefield St., London WC1N 1PF, United Kingdom, yi.xu@ucl.ac.uk)

An experiment was carried out to test the hypothesis that the syllable is a time structure that synchronizes tonal, consonantal, and vocalic target approximation movements. The strategy was to align formant movements with F0 turning points of lexical tones as time reference, and then assess the temporal scope of articulatory movements by comparing formant trajectories and their turning points across minimal pairs. Native Mandarin speakers produced C1V1#C2V2 disyllabic sequences where C2 is /y/, /w/ or /l/, and V1 and V2 varied in height and frontness. Analysis of F0-aligned F2-3 (average of F2 and F3) trajectories revealed patterns in support of the main hypothesis. First, movements clearly discernable as approaching either C2 or V2 targets started at about the same time from the center of V1, i.e., well before the conventional landmark-based syllable boundary. Second, some F2-3 trajectories extended continuously from the center of V1 to the center of V2, across the intervening /l/, indicating a long and uninterrupted V2 approximation movement. These results provide support for the view that genuine CV co-production occurs only between onset C and the following V, while the rest of the "coarticulation" is only an epiphenomenon (arising from landmark-based segmentation) of syllable-synchronized target approximation.

9:45–10:00 General Discussion

10:00–10:30 Break

10:30

4aSC6. Gradiency vs categoricity in the production of prosodic boundaries as reflected in different kinematic measures. Jelena Krivokapic (Linguist., Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109-1220, jelenak@umich.edu), Christine Mooshammer (Institut für deutsche Sprache und Linguistik, Humboldt-Universität zu Berlin, Berlin, Germany), and Mark Tiede (Haskins Labs., New Haven, CT)

Two questions are examined in an EMA experiment: (1) are speech gestures at prosodic boundaries produced in a categorical or in a gradient manner (extending work by Krivokapic, Mooshammer, and Tiede 2013), and (2) how do different articulatory measures reflect boundary strength. Forty-eight sentences were constructed, each containing one, two, or three prosodic boundaries, for a total of 56 boundaries per sentence set. The range of predicted boundary types varied from a weak clitic boundary to a strong sentence boundary. Each boundary fell between the words "column and." Seven subjects read six to eight repetitions of these sentences. Various temporal properties of boundaries have been examined, including the duration of the lip opening movement for [m], which is the movement closest to the boundary (and therefore most strongly reflects the boundary properties), and the duration of the jaw movement from the opening movement for the first vowel in "column" to the opening movement for the postboundary vowel (a variable which spans the boundary). These measures have been subjected to Gaussian mixture model analysis. Results show evidence for a more fine-grained structure than can be predicted by the three prosodic levels of traditional models. [Work supported by NIH DC003172-16, DC008780, DC002717].

10:45

4aSC7. Vowel locus equations as a measure of vowel coarticulatory aggressiveness. Wei-rong Chen (Graduate Inst. of Linguist., National Tsing Hua Univ., 3029 S. Grand Ave., Apt. 10, Los Angeles, CA 90007, waitlong75@gmail.com) and Khalil Iskarous (Dept. of Linguist., Univ. of Southern California, Los Angeles, CA)

Studies in locus equations, a quantification of the degree to which F2 at vowel onset (or consonant place) can be predicted by F2 at vowel midpoint (or vowel place), have shown that the slope of locus equations is a reverse measure of coarticulatory resistance of consonants, in consonant-vowel (CV) sequences with C fixed and V varying. This study presents the first application of locus equations to the measure of coarticulatory properties of

vowels. The locus equations analysis for vowels, on our articulatory data (collected with electromagnetic articulograph, EMA) of CV syllables from 7 Taiwan Mandarin speakers, with V fixed and C varying, shows that vowel /i/ has the greatest slope among vowels at tongue body, whereas vowel /a/ and /u/ mostly do not distinguish from each other, and at the tongue tip, the slope does not vary across the three vowels /i/, /a/ and /u/. Based on the CV model theory of gestural coordination (C-V in-phase relation) and the formula of linear regression, we claim that the slope of vowel locus equations is positively related to coarticulatory aggressiveness of vowel, which is further supported by a comparison with a well-established measure of vowel coarticulatory aggressive, based on contextual variability, on the same data.

11:00

4aSC8. An MRI comparison of /s/ production in four subject conditions. Andrew D. Pedersen (Neural and Pain Sci., Univ. of Maryland Dental School, 650 W. Baltimore St Rm 8207, Baltimore, MD 21201, mstone@umaryland.edu), Jun Hwang (Orthodontics, Univ. of Maryland Dental School, Baltimore, MD), Jonghye Woo (Neural and Pain Sci., Univ. of Maryland Dental School, Baltimore, MD), Fangxu Xing, Jerry L. Prince (Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD), and Maureen Stone (Neural and Pain Sci., Univ. of Maryland Dental School, Baltimore, MD)

This study examined a control subject and three patients who had surgery to remove tongue cancer. One patient's surgery was closed with sutures, one with a radial forearm free flap reconstruction, and one with sutures plus radiation. This study aims to ascertain the effects of these traumas on internal and surface tongue motion during speech. The flap consists of soft tissue that is vascularized, but not innervated; it increases tongue bulk, but has no direct motor control. The other two patients have missing tissue and a scar where the cut regions were sewn together. This morphological change may increase difficulty creating properly formed palatal contacts. The supplemental radiation treatment may cause additional muscle stiffness due to fibrosis. The cine and tagged datasets were recorded in axial, coronal, and sagittal orientations using identical parameters so their data could be overlaid. Each dataset was reconstructed into 3D volumes, one for each time-frame in the word. From each cine-MRI volume, the 3D tongue surface was segmented and used as a "mask" in the tagged-MRI volume. In the tagged-MRI volumes, 3D displacement fields were calculated to show motion of each tissue point inside the tongue mask during the speech task.

11:15

4aSC9. Visualization of the laryngeal muscles with high-quality magnetic resonance imaging. Sayoko Takano (Media Information Sci., Kanazawa Inst. of Technol., 7-1 Ohgigaoka, Nonoichi 921 - 8501, Japan, tsayoko@neptune.kanazawa-it.ac.jp) and Kiyoshi Honda (Tianjin Univ., Tianjin, China)

The larynx is the organ to adjust voice in speech important for controlling voice phonation, and the tensions of the vocal folds cord is controlled by the relative positions among the thyroid, arytenoid, and cricoid cartilages. Magnetic resonance images (MRI) have been employed to observe the positions of those cartilages, while however, visualizing the laryngeal muscles for activating moving those cartilages is still difficult with the conventional MRI. This study aims to visualize and identify the laryngeal muscles based on the composition of the three different sets of gray-scale MRI data into red, green, and blue channels, namely RGB-MRI. The types of imaging parameters image variations are field-echo opposed-phase (FE(op)), spin-echo proton density weighted (SE(PDw)), and spin-echo T1 weighted (SE(T1w)). This method for image composition helps us identifying could reveal the location of the muscles in the laryngeal region *in vivo*. We have successfully identified the sternohyoid muscle, thyrohyoid muscle, lateral thyroarytenoid muscle, cricothyroid muscle, stenrocleidmustoid muscle, omohyoid muscle, and inferior constrictor muscles. The RGB-MRI could give further provide anatomical information of the larynx, especially about the muscle location and its deformation involved in on speech mechanisms, which will be useful for both research experts and beginners.

11:30

4aSC10. Motor control primitives arising from a dynamical systems model of vocal tract articulation. Vikram Ramanarayanan, Louis Goldstein, and Shrikanth Narayanan (Univ. of Southern California, 3740 McClintock Ave., EEB421, Los Angeles, CA 90089-2564, vramanar@usc.edu)

We have previously presented a computational approach to derive interpretable movement primitives from speech articulation data using a convolutive Nonnegative Matrix Factorization with sparseness constraints (cNMFsc) technique (Ramanarayanan *et al.*, Interspeech 2011; Ramanarayanan *et al.*, J. Acoust. Soc. Am. **134**(2), in press). However, it is not clear whether finding such a dictionary of primitives can be useful for speech motor control, particularly in finding a low-dimensional subspace for such control. In this paper, we examine this possibility in two steps. First, we use the iterative Linear Quadratic Gaussian (iLQG) algorithm to derive a set of control inputs to a dynamical systems model of the vocal tract that produces a desired movement sequence. Second, we use the cNMFsc algorithm to find a small dictionary of control input "primitives" that can be used to drive said dynamical systems model of the vocal tract to produce the desired range of articulatory movement. We show, using both qualitative and quantitative evaluation on synthetic data produced by an articulatory synthesizer, that such a method can be used to derive a small number of control primitives that produce linguistically interpretable and ecologically valid movements. Such a primitives-based framework could help inform theories of speech motor control and coordination.

11:45–12:00 General Discussion

Session 4aSP

Signal Processing in Acoustics: Alternative Array Spacing and Time Sampling Techniques

Andrew T. Pyzdek, Cochair
Appl. Res. Lab., The Penn State Univ., PO Box 30, State College, PA 16804

Jeffrey A. Ballard, Cochair
Appl. Res. Labs., The Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029

Invited Papers

9:55

4aSP1. Detection performance of coprime sensor arrays. Kaushallya Adhikari and John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts, Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, kadhikari@umassd.edu)

Coprime sensor arrays (CSAs) achieve the resolution of a fully populated uniform linear array (ULA) with the same aperture using fewer sensors. The CSA's reduced number of sensors diminishes its ability to attenuate white noise, and consequently also reduces its array gain. Assuming narrowband independent signal and white noise, when the signal and the noise are modeled by circular complex zero mean Gaussian distributions, the detection statistic distribution for the ULA simplifies to an exponential distribution with its parameter dependent on the input SNR. The detection statistic for the CSA is the product of the output of one subarray with the complex conjugate of the output of the second subarray. Being the product of two independent complex Gaussian random variables, the CSA detection statistic has a distribution proportional to a modified zeroth order Bessel function of the second kind [O'Donoughue and Moura, IEEE Signal Process. (2012)]. Manipulating the ULA and CSA detection statistic distributions provides analytical expressions for probabilities of detection and false alarm and also mean discriminating information. Evaluating these expressions confirms that a CSA pays a detection gain penalty of about 5 dB over a wide range of SNRs and array sizes. [Work supported by ONR.]

10:15

4aSP2. The application of compressive sensing to underwater acoustic array processing and design. Jeffrey S. Rogers, Charles F. Gaumond, and Geoffrey F. Edelmann (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375, jeff.rogers@nrl.navy.mil)

A brief overview of the application of compressive sensing to underwater acoustic array processing is given. Specific algorithms for estimating direction of arrival via L1 minimization are introduced. The Statistical Reduced Isometry Property (StRIP) is defined and used as a method to determine the ideal array designs for compressive beamforming. The tradeoff of using nested apertures and non-integer element spacing for improving StRIP over a broad range of frequencies is studied numerically. The array gain, which is a more directly useful performance metric, is also estimated using Monte Carlo methods. Compressive beamforming results from at-sea data taken on the Five Octave Research Array (FORA) will be presented. [Research funded by the Office of Naval Research.]

10:35

4aSP3. Coprime arrays in the context of compressive sensing. Andrew T. Pyzdek and R. Lee Culver (Appl. Res. Lab., The Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu)

Arising naturally from the formalism of coarrays, coprime arrays offer a means of achieving sparsity along an array while maintaining certain measures of performance. This configuration offers benefits over traditional configurations for large aperture towed arrays in the ocean environment, but shallow water propagation complicates the wide sense stationarity assumption that is essential to proper application of coarray concepts. Compressive sensing techniques allow for the reconstruction of generalized signals which are sparse on some known basis. Existing literature has shown the connection between compressive sensing and random arrays, drawing direct parallels between the results and methods that arose independently in both fields. Compressive sensing may offer an alternative understanding of coprime arrays, which is independent of wide sense stationarity, and thus applicable to shallow water environments. Coprime arrays will be considered in the context of compressive sensing. It will be discussed if the measurement matrices that arise from this array geometry might possess properties necessary for compressive sensing reconstruction to function, such as the Restricted Isometry Property. [This research was supported by the Applied Research Laboratory, at the Pennsylvania State University through the Eric Walker Graduate Assistantship Program.]

10:55

4aSP4. Near-field beamforming with augmentable non-uniform arrays for far-field interference suppression. Jonathan Odom and Jeffrey Krolik (Elec. and Comput. Eng., Duke Univ., PO Box 90291, Durham, NC 27708, jonathan.odom@duke.edu)

This paper addresses the problem of adaptive near-field beamforming in the presence of many far-field interferers. Minimally redundant and, more generally, augmentable non-uniform arrays span larger array apertures and permit adaptive discrimination of more sources than sensors but require spatial-stationarity and increased snapshot support, which precludes their use for dynamic, near-field sources. In this paper, adaptive beamforming with augmentable non-uniform arrays is proposed which nulls relatively static far-field interference while enabling the detection of dynamic near-field sources. The use of an augmentable under-sampled array permits formation of a covariance matrix that can be used to implement reduced-rank dominant rejection of more interferers than sensors. Meanwhile, near-field beamforming in bearing and range is achieved using only the physical non-uniformly spaced sensor locations. An interference dominated environment is simulated to demonstrate the increase of array gain achieved over current techniques in the low snapshot support regime. [Work sponsored by ONR.]

11:10

4aSP5. Coprime microphone arrays for direction-of-arrival estimation. John P. Nichols and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, jpnichols9@gmail.com)

Direction-of-arrival estimation using microphone arrays is relatively simple with low signal-to-noise ratio, but precise location estimation amongst noise requires many sensors to reduce beam width. Coprime linear microphone arrays allow for narrow beams with fewer sensors. A coprime microphone array is made up of two overlapping uniform linear arrays with M and N sensors, where M and N are coprime. By applying spatial filtering with both arrays and combining their outputs, $M + N$ sensors can yield MN directional bands. In this work, a coprime microphone array is used to estimate the location of multiple uncorrelated narrowband sources hidden in noise.

11:25

4aSP6. Beam-patterns of a collocated unit of three orthogonally oriented second-order directional sensors. Kainam T. Wong and Yang Song (Dept. of Electron. & Information Eng., Hong Kong Polytechnic Univ., DE 605, Hung Hom KLN, Hong Kong, ktwong@ieee.org)

A microphone or a hydrophone measures the incident acoustic wavefield as a scalar field of pressure. Underlying this pressure scalar is the particle-velocity vector, which constitutes the spatial gradient of the wavefield. Each Cartesian component of this gradient vector may be directly measured (without computing any spatial derivative) by a particle-velocity sensor aligned in parallel to that Cartesian coordinate. The entire 3×1 gradient vector may be measured at any point in space, using three such particle-velocity sensors, collocated but orthogonally oriented among themselves. Such a collocated triad has an array manifold independent of signal frequency, thus uncoupling azimuth-elevation beamforming from the frequency-dimension. This triad's beam-pattern has already been investigated by Wong and Chi ["Beam patterns of an underwater acoustic vector hydrophone located away from any reflecting boundary," IEEE J. Ocean. Eng. **27**(3), 628–637 (2002)]. While the particle-velocity sensor measures the first-order spatial derivative of the pressure field, a second-order spatial derivative of the pressure-field could likewise be defined and be measured by a "second-order directional sensor" that would have sharper directivity. This paper follows up that investigation, but now for second-order directional sensors.

11:40

4aSP7. Objective functions incorporating various norms for the three-dimensional sound manipulation. Yang-Hann Kim (Ctr. for Noise and Vib. Control, KAIST, Dept. of M.E., Sci. Town, Daejon-shi 305-703, South Korea, yanghann@kaist.ac.kr), Jung-Woo Choi, and Min-Ho Song (Ctr. for Noise and Vib. Control, KAIST, Daejeon, South Korea)

Using many array speakers to manipulate sound in space and time requires proper objective functions that can determine the array speakers' magnitude and phase relationships. Mathematically speaking, this means that magnitude and phase of each speaker have to be determined in such a way that certain designed objective functions can be minimized or maximized. The most popular way to define the objective function is to utilize the form of L2 norm, but it is also likely possible to utilize L1 norm to decide corresponding array speakers magnitudes and phases. Depending on what we would like to manipulate in space and time, the objective functions can be selected: they can be, for example, energy in a selected zone, or contrast of sound energy, for the case of L2 norm. In this paper, we study how those functions affect the performance of sound manipulation in terms of array parameters and effectiveness of 3D sound manipulation.

11:55

4aSP8. Modeling imaging performance of multistatic acoustic arrays of non-uniform geometries. Michael Lee, Michael Liebling, and Hua Lee (Elec. and Comput. Eng., Univ. of California, Santa Barbara, Harold Frank Hall, Rm. 4155, Santa Barbara, CA 93106-9560, michaellee@umail.ucsb.edu)

Unlike the most arrays in acoustical imaging systems, a system based on a dynamically reconfigurable multistatic array would be able to physically adapt to the volume of interest, improving angular coverage of the illuminating array aperture and thus resolution of target images. With such arbitrary distributions of array elements, the extent to which each element contributes to the image reconstruction of targets may change with each imaging cycle. For this reason, the acoustic beamspread and direction of each array element become dominant variables in the resolving capability and data processing of the system. In this paper, non-linear element distributions with specified beamspreads are simulated to image point targets for the purpose of analyzing this relationship and its relevance to spatial bandwidth. MATLAB simulations based on coherent illumination and image reconstruction provide the foundation for this work. By adding the capability to specify beamspread angles and direction for each element, imaging simulation results have more practical value. For any given array configuration, the region of interest can now be visualized in terms of favorability for acoustic illumination. Likewise, the beamspread specifications enable the image reconstruction algorithm to more reliably reconstruct targets.

Session 4aUW

Underwater Acoustics and Acoustical Oceanography: The Acoustics of Bubbles and Bubble Clouds in the Ocean

Grant B. Deane, Cochair
Marine Physical Lab., Univ. of California, San Diego, La Jolla, CA 92093-0238

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

Chair's Introduction—8:55

Invited Papers

9:00

4aUW1. What is the surface tension at a bubble wall? Tim Leighton, Mengyang Zhu (Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton, Hampshire SO17 1BJ, United Kingdom, tgl@soton.ac.uk), and Peter Birkin (School of Chemistry, Univ. of Southampton, Southampton, United Kingdom)

The surface tension on a bubble wall frequently arises in models of single bubbles and is an influential parameter when assessing many important processes. These range from the global transfer of mass, momentum, and energy between the oceans and the atmosphere, to the efficacy of microbubbles for clinical diagnosis and therapy. However, the simplicity by which surface tension appears in formulations belies the complexity of the process. For example, the distribution of chemical species or dopants may be uneven over the wall of a single bubble, the translation of a bubble relative to the bulk liquid may change the chemical loading over time, and the pulsation and shape changes of bubbles perturb the distribution of a dopant on the bubble wall. This paper reports on an experimental technique for estimating one measure of the surface tension at the gas/liquid boundary of a bubble.

9:20

4aUW2. The influence of temperature on bubble formation under breaking waves. Helen Czerski (Inst. for Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton SO17 1BJ, United Kingdom, h.czerski@soton.ac.uk)

The bubble plumes generated by breaking waves have a complex structure that is highly dependent on the local environment. These temporary bubble populations have a strong influence on many processes at the air-sea interface, for example, air-sea gas transfer, aerosol production, sound transmission, and optical absorption. To quantify the importance of the bubble plume, two things are required: the state of the initial bubble plume and an understanding of the longer-term plume evolution. This paper focuses on the formation of the initial bubble plume. I'll present evidence showing the effect of temperature on the fragmentation of single bubbles in the laboratory. These results imply that changing the water temperature has consequences for ocean bubble plume structure, bubble size distribution, and whole bubble plume acoustics, and these will be discussed.

Contributed Papers

9:40

4aUW3. Acoustic properties of oil/gas/sea water mixtures. R. Lee Culver (ARL, Penn State Univ., PO Box 30, State College, PA 16804, rlc5@psu.edu)

This paper presents an approach to calculating the acoustic properties, i.e., sound speed, and scattering cross-section, of oil, oil/gas, and oil/gas/sea water mixtures. The petroleum industry characterizes reservoir fluid in terms of the relationship between pressure, volume and temperature (PVT) of the oil utilizing the formation volume factor (FVF), which is defined as the ratio of the volume of oil at a pressure and temperature relative to a stock tank barrel (STB) at surface conditions. Another important oil reservoir parameter is the gas-oil ratio (GOR), which is the percent by volume of gas which is present in the oil at reservoir conditions (the gas is typically entirely dissolved—i.e., none is free—due to the great pressure). The reservoir fluid is typically under-saturated, meaning more gas could be dissolved in the oil, were it present. As the oil rises to the surface, the pressure and temperature drop. Once they drop below the bubble point, gas begins to come out of solution. Also, if fluids (oil and gas) are released into the ocean, sea water will become mixed in with the well fluids. The interest here is in the acoustic properties of oil only, oil/gas and oil/gas/sea water mixtures. The approach is to calculate the density and compressibility of the constituents and from them predict the aggregate acoustic properties using mixture theory.

9:55

4aUW4. Underwater sound radiated by bubbles released by melting glacier ice. Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Erin C. Pettit (Dept. of Glaciology and Geophys., The Univ. of Alaska Fairbanks, Fairbanks, AK)

Passive acoustics monitoring techniques have been examined as a method to remotely sense activity of glacier ice near the ice-ocean boundary [Ann. Glaciol. **53**, 113–121 (2012)]. Sound from glacier calving events and the resultant breaking and bobbing of the ice after impact with the water ranges from infrasound (<10 Hz), to low-frequency (100 Hz—1 kHz), to higher frequency sound (>10 kHz) generated by breaking tsunamis and seiches after impact. Bubbles are known to form within ice during glacier formation and can be released from glaciers as they undergo submarine melting. Due to the possibility of high internal bubble pressure, this release can occur in the form of jetting or squirting events. Signals hypothesized to be from bubbles being released from melting glacier ice were measured in Unakwik Inlet and Icy Bay, Alaska using passive autonomous hydrophone moorings and near-surface recordings in the 1 kHz—3 kHz frequency range. To temporally and spatially correlate such acoustic emissions with bubble activity, a set of laboratory measurements was performed using small samples of glacier ice, and acoustic emission was positively correlated with bubble release. Taken together, these measurements support the use of passive acoustics to monitor marine glacier ice melt.

10:10–10:30 Break

10:30

4aUW5. Low scatterer density limit of effective medium theory. Craig N. Dolder (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dolder@utexas.edu), Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Often effective medium theories assume that there are multiple scatterers per wavelength; however, practical limits for these models can be far more flexible. Commander and Prosperetti's model for sound propagation in bubbly liquids assumes that "an averaging volume must contain many bubbles" but does not explicitly assume that there must be many bubbles per wavelength, though that assumption might be considered implicit in the use of effective medium theory. The results of Foldy's exact multiple scattering theory are compared to the effective medium theory for very low scatterer densities in order to determine the low scatterer limit for which the effective medium theory holds. This is compared with experimental results with low scatterer densities. It is shown that effective medium models for bubbly liquids hold even when there are as little as one scatterer per wavelength. [This work was supported by the Office of Naval Research.]

10:45

4aUW6. Time and frequency analysis of air bubbles injected underwater. Hannan Lohrasbipeydeh (Dept. of Elec. Eng., Univ. of Victoria, EOW 448, 3800 Finnerty Rd., Victoria, B.C. V8P 5C2 Canada, lohrasbi@uvic.ca), Tom Dakin (Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and T. Aaron Gulliver (Elec. Eng., Univ. of Victoria, Victoria, BC, Canada)

The acoustic signal generated by vibrating air bubbles injected in water allows for the passive detection of underwater sources such as divers and gas leaking from pipes. In this paper, an experimental analysis of the time and frequency characteristics of these signals is presented. The decaying acoustic signal of a single bubble created by nozzles of different diameters is described and extended to a train of bubbles from an array of nozzles. The oscillatory motion of the bubble surface generates an acoustic signal that has a damped sinusoidal characteristic. The spectrum of this waveform has a fundamental frequency and several harmonics. Therefore, the effect of ambient noise can be suppressed using a suitable band pass filter with a center frequency equal to the bubble fundamental frequency. It is shown that the acoustic signal generated by these bubbles can be utilized for passive detection and remote sensing (including depth estimation). Experimental data from a Saanich inlet experiment is analyzed to obtain realistic results. These results are compared with the theoretical analysis.

11:00

4aUW7. The limits of using closed-cell foam as a pressure release condition when dealing with underwater acoustics. Craig N. Dolder and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dolder@utexas.edu)

Closed-cell foams are frequently used in both the research and industry communities to provide a pressure release condition for underwater sound. In practice, closed-cell foams can effectively imitate a pressure release condition even if the thickness of the foam is as small as 1/60th the wavelength of the sound in water, however these conditions start to break down as the sound speed in water is reduced. The presence of even a small volume fraction of bubbles can cause the sound speed to drop in such a way that the water can couple with the closed-cell foam and prevent the interface from providing a pressure release condition. Measurements made in an acoustic resonator are used to show how the idealized system breaks down. [This work was supported by the Office of Naval Research.]

Session 4pAA

Architectural Acoustics: The Enduring Contributions of Two Giants in Building Acoustics: Ronald L. McKay and Warren E. Blazier

David A. Conant, Cochair
MCH Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362

Joel A. Lewitz, Cochair
Rosen Goldberg Der & Lewitz, 1100 Larkspur Landing Cr., Larkspur, CA 94939

Chair's Introduction—1:00

Invited Papers

1:05

4pAA1. Ronald L. McKay, FASA: The Los Angeles years. David A. Conant (McKay Conant Hoover Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, dconant@MCHinc.com)

Ron McKay already had two very productive decades in various BBN field offices by the time we met in 1977, and was a high-level "generalist" with a growing portfolio of respected work in performing arts facilities. After 10 years together at BBN's Los Angeles office, we established our own consulting practice, McKay Conant Brook inc, with an emphasis on spaces for large public assembly. This paper focuses on Ron's contributions in the area of his special passion—performing arts—and various acknowledgments and awards leading to the prestigious AIA National Honor Award for Collaborative Achievement in 1999. His finest work was yet to come.

1:25

4pAA2. Growth in acoustics consulting at Bolt, Beranek and Newman with Ron and Warren. Carl Rosenberg (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, crosenberg@acentech.com), William J. Cavanuagh (Cavanaugh Tocci, Natick, MA), and Eric Wood (Acentech Inc., Cambridge, MA)

Acoustics consulting was a growing field in technology support to the architectural world in the second half of the 20th century, spearheaded by Bolt Beranek and Newman and other new firms. As part of this growth, BBN embarked on a program of establishing branch offices around the country. Ron McKay and Warren Blazier were outstanding technical leaders in their respective fields and also vanguards of these developments, although both found fulfillment later in their careers away from BBN. This paper shares the growth and evolution of the profession through these two paragons of consulting leadership, and acknowledges the profound contributions of their early years to the reputation of the acoustics consulting world.

1:45

4pAA3. On the shoulders of giants: Remembering the contributions of Warren Blazier. Robert F. Mahoney (Robert F Mahoney & Assoc., 310 Balsam Ave., Boulder, CO 80304-3238, rfm@rfma.com)

Gentleman, scholar, mentor, Warren Blazier exemplified all these roles in his career of six decades. Many of us who benefited from his research, his teaching, and especially from his unstinting and magnanimous instruction owe Warren a great deal. This brief talk will discuss some of the frontiers Warren established—and significantly advanced—as well as the lessons we beneficiaries learned from him. Many of these lessons extend well beyond the science of noise control and, we hope, can be passed on to our successors as well.

2:05

4pAA4. Warren Blazier's contributions to building acoustics, a manufacturer's perspective. Norman Mason (Mason Industries, Inc., 350 Rabro Dr., Hauppauge, NY 11788, nmason@mason-ind.com)

Warren Blazier and Norm Mason first met at York Borg-Warner's Engineering Offices over 50 years ago; Warren was running their Acoustics Department. Warren moved on to Bolt Beranek Newman and then physically to San Francisco and the two remained great friends. Norm Mason consulted Warren on his theoretical understanding of Noise and Vibration Control and Warren was influential in the development of Mason Industries Architectural products. This paper will attempt to capture the highlights of Warren Blazier's friendship with Mason Industries, Inc.

2:25–2:40 Break

2:40

4pAA5. Lessons from the dean of noise control. Jim X. Borzym (Borzym Acoust. LLC, 2221 Columbine Ave., Boulder, CO 80302, acoustics@columbine.net)

Professorial, collegial, passionate, and gentlemanly, Warren E. Blazier, Jr., was admired in our profession as a top-level consultant and collaborator, and sometimes referred to as the “Dean” of noise control engineering. This presentation will highlight several technical topics incorporated in project consultations by Warren Blazier. Topics include duct plena, fan efficiency, structural resonance, floated floors, and unusual projects. Comments and personal insights presented by a mentee and collaborator of Warren Blazer’s during the post-BBN decade 1986 through 1996.

3:00

4pAA6. Warren Blazier—The consultant’s consultant. Richard Talaske (TALASKE | Sound Thinking, 1033 South Bldg., Oak Park, IL 60302, rick@talaske.com)

Many acoustic consultants turned to Warren for advice or confirmation of design solutions to solve their toughest noise control design challenges, this consultant included. Spanning nearly three decades, Warren collaborated with TALASKE | Sound Thinking technically as Associated Acoustical Consultant on retainer and business-wise as a member of the Board of Directors. Insights into the expertise and personality of Warren will be provided, with added contributions by Jerry Hyde.

3:20

4pAA7. Warren Blazier and his contributions to noise rating criteria. Kenneth P. Roy (Bldg. Products Technol. Lab, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrong.com)

Warren Blazier was well known for his work with sound and vibration in mechanical systems, e.g., HVAC equipment. As such, he was particularly active at ASHRAE in addition to the ASA meetings. Although I was always focused more on the “room acoustics” side of the design, I none-the-less offered to help edit the 1999 ASHRAE Handbook chapter 43 with Warren and Chuck Ebbing. This section dealt with “Indoor Sound Criteria,” and what an eye opening event this turned out to be. You might say that I jumped into the fire as there was an ongoing “tug-of-war” over which rating system should be adopted to describe background noise in buildings from mechanical systems. NC, RC, RC mark II, NCB … several of these had been used up to this point and lots of controversy existed as to what to include in the re-write. A somewhat historic special meeting of ASHRAE TC2.6 was held in Boston on the evening of Sunday 27 June 1997 from 3:30 to 6:30 pm where this was discussed to resolve the issue. Both Warren and Leo Beranek were asked to make presentations on the merits of the competing approaches. And this is what was decided!!

3:40

4pAA8. Deep roots and spreading branches—A shared legacy of acoustic DNA. Larry Kirkegaard (Kirkegaard Associates, 801West Adams St., Chicago, IL 60607, lkirkegaard@kirkegaard.com) and Len Auerbach (Auerbach Pollock Friedlander, New York, NY)

Ron McKay and Warren Blazier were part of a generation of strong would-be consultants that were drawn to Bolt Beranek and Newman in the early to mid-sixties. They brought with them broad skills, great curiosity, and a contagious spirit of collaboration. Harvard and MIT were prime sources of talent, but BBN was magnetic to talent as well as project work from around the world. To know and appreciate Ron and Warren, you must know the richness of talent that surrounded and influenced them—their professional companions. Imagine a lively discussion with input from the likes of Leo Beranek, Dick Bolt, Bob Newman, Bill Cavanaugh, Ted Schultz, David Klepper, Bill Watters, Rein Pirn, George Kamperman, Bob Hoover, Carl Rosenberg, Layman Miller, Jacek Figwer, Jack Curtis, Russell Johnson, Bob Wolff, Tom DeGaetani, Len Auerbach, Dennis Paoletti, Joel Lewitz, and Dave Conant among many others. Arm-to-arm they could reach around the world. Their legacies enrich our profession in a myriad of ways. This paper shares memories and insights into both the genius and humanness of an important generation of our colleagues.

Session 4pAB

Animal Bioacoustics and Noise: Bioacoustic Contributions to Soundscapes II

John Hildebrand, Cochair
Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093

Simone Baumann-Pickering, Cochair
Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093

Invited Papers

1:30

4pAB1. Cyclical patterns in long-term bioacoustic data. Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., San Diego, CA), Ana Širović, and John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Cyclical patterns in biological systems occur from small to large temporal scales, and these cycles are also reflected in acoustic data. To interpret long-term patterns appropriately, knowledge about natural cycles is crucial. We will show examples of diel, lunar, and seasonal patterns for a variety of species from invertebrates to vertebrates. The observed patterns are likely driven by intra-specific communication, or prey behavior and availability, which in turn can be related to large scale and long-term environmental modulation. Absence of acoustic signals does not necessarily indicate absence of the caller but the calling behavior may be limited to a certain time of the cycle period. In addition, various acoustical sources can complicate results. Patterns from abiotic and anthropogenic sources may mask or alter natural biological acoustic cycles. Animals moving just outside of an acoustic recorder's detection range may lead to misinterpretation of diel behavior. Effects of multi-annual or multi-decadal cycles are extremely difficult to investigate due to the long time series needed to take them into consideration.

1:50

4pAB2. Patterns in bioacoustic activity observed in U. S. National Parks. Megan F. McKenna, Dan J. Mennitt, Emma Lynch, Damon Joyce, and Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO CO, megan_f_mckenna@nps.gov)

The Natural Sounds and Night Skies Division of the U.S. National Park Service has collected month-long acoustic recordings at more than 300 sites in 73 park units located throughout the United States, dating back to 2000. Each monitoring session lasted 25 days or more; some sites were monitored more than once. At all sites a calibrated Sound Level Meter recorded acoustic data in one-second, one-third octave band resolution; at many sites, simultaneous continuous acoustic recordings were collected using a digital audio recorder. These data were analyzed to identify broad patterns in bioacoustic activity within one-third octave bands. These bioacoustical patterns were analyzed in relation to site characteristics, seasons, and anthropogenic noise levels to identify significant associations. The resultant model could be used to produce a map predicting bioacoustical activity throughout the coterminous United States.

2:10

4pAB3. Baleen whale calls in the Southern California Bight from 2009 to 2012. Ana Sirovic (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., MC 0205, La Jolla, CA 92093-0205, asirovic@ucsd.edu), Marie A. Roch (San Diego State Univ., San Diego, CA), Simone Baumann-Pickering, Jasmine Buccowich, Amanda Debich, Sarah C. Johnson, Sara M. Kerosky, Lauren K. Roche, Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA)

Baleen whales are an important contributor to the Southern California Bight soundscape. Calls from some species only contribute seasonally, while others are part of the soundscape year-round, albeit at varying levels. Passive acoustic monitoring has been conducted at the U.S. Navy's Southern California Offshore Range (SCORE) since 2009. Data were collected at two sites concurrently using High-frequency Acoustic Recording Packages (HARPs). Analysis in the low frequency band (10 Hz—1 kHz) has yielded results on the seasonal and interannual variation in the presence of calling blue (*Balaenoptera musculus*), fin (*B. physalus*), Bryde's (*B. edeni*), and humpback whales (*Megaptera novaeangliae*). Calls of most species (blue, Bryde's, and humpback whales) were only present during part of the year, indicating seasonal migration common for these species. On the contrary, fin whale calls were prevalent year-round, although the abundance of their 20 Hz calls tended to decrease in the summer. The link between the relative changes in interannual call abundance (i.e., soundscape contribution) and prevailing environmental conditions, such as the sea surface temperature and chlorophyll a, was investigated using the Tethys spatial-temporal database framework. These links can be important for understanding the year-to-year variation in soundscape.

Contributed Papers

2:30

4pAB4. Deep-diving cetaceans and the Deepwater Horizon oil spill. Karlina Merkens (UC San Diego, SIO, 9500 Gilman Dr., MC 0205, La Jolla, CA 92093-0205, kmerkens@ucsd.edu), Mark McDonald (Whale Acoust., Bellvue, CO), Simone Baumann-Pickering, Kaitlin Frasier, Sean Wiggins, and Hildebrand John (UC San Diego, SIO, La Jolla, CA)

The Gulf of Mexico is home to at least six species of deep-diving cetaceans, including beaked whales, sperm whales, and dwarf and pygmy sperm whales. These species are all found in the region that was impacted by the Deepwater Horizon oil spill. Using High-frequency Acoustic Recording Packages (HARPs), we monitored for their presence at three deep-water sites. From over two years of wideband (10 Hz—100 kHz) recordings, the detections of deep-diving cetacean sounds were related to environmental and anthropogenic factors using Generalized Additive Models to identify relevant features. The modeling showed that the significance of habitat parameters varies by species and site, although lunar illumination and sea surface height anomaly were significant for most species at all sites. The relationships between the acoustic presence of the cetaceans and their environment help provide an understanding of the ecology of these species as well as the potential impact of the oil spill on their habitat. This material is based upon work supported by BP and NOAA under Award Number 20105138. Any opinions, findings, and conclusions or recommendations expressed in this publication are those of the author(s) and do not necessarily reflect the views of the BP and/or any State or Federal Natural Resource Trustee.

2:45

4pAB5. Prospects for short-term and long-term passive acoustic monitoring of environmental change impact on marine mammals. Natalia Sidorovskaia (Phys., Univ. of Louisiana at Lafayette, UL BOX 44210, Lafayette, LA 70504-4210, nas@louisiana.edu), Azmy S. Ackleh (Mathematics, Univ. of Louisiana at Lafayette, Lafayette, LA), Christopher O. Tiemann (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Juliette W. Ioup, and George E. Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

The Littoral Acoustic Demonstration Center, LADC, a consortium of scientists from four Gulf state universities and the U.S. Navy, was begun in 2001 to study underwater noise and acoustic propagation, and the impact of human activities in the ocean on marine mammals, with emphasis on the Gulf of Mexico (GoM) region. LADC has a library of broadband passive acoustic data, collected by autonomous bottom-moored buoys, which sampled the GOM region ambient noise state, seismic airgun array emissions, and/or marine mammal activities six times during the last decade. LADC acoustic data represent an opportunity to study the short- and long-term effects of environmental changes on the marine mammal population. Environmental factors include baseline anthropogenic noise levels, passages of tropical storms, seismic exploration surveys in the area, and the 2010 Deepwater Horizon oil spill accident. The talk summarizes recent findings on the relationship between regional population dynamics of sperm and beaked whales and abrupt environmental changes with emphasis on the recent GoM oil spill. Statistically significant results of the study suggest a need for establishing consistent acoustic monitoring protocols in the oceanic areas of current or potential industrial activities. [Past data acquisitions were supported by ONR, SPAWAR, JIP, NSF, and Greenpeace.]

3:00

4pAB6. Tethys: A workbench for bioacoustic measurements and environmental data. Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720, marie.roch@sdsu.edu), Simone Baumann-Pickering, Daniel Hwang, Heidi Batchelor (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), Catherine L. Berchok (Fisheries Sci. Centers, NOAA, Seattle, Washington), Danielle Cholewiak (Fisheries Sci. Centers, NOAA, Woods Hole, Massachusetts), John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), Lisa M. Munger, Erin M. Oleson (Fisheries Sci. Centers, NOAA, Honolulu, Hawaii), Shannon Rankin (Fisheries Sci. Centers, NOAA, San Diego, California), Denise Risch (Fisheries Sci. Centers, NOAA, Woods Hole, Massachusetts), Ana Širović (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), Melissa S. Soldevilla (Fisheries Sci. Centers, NOAA, Miami, Florida), and Sofie M. Van Parijs (Fisheries Sci. Centers, NOAA, Woods Hole, Massachusetts)

A growing number of passive acoustic monitoring systems have resulted in a wealth of annotation information, or metadata, for recordings. These metadata are semi-structured. Some parameters are essentially mandatory (e.g., time of detection and what was detected) while others are highly dependent upon the question that a researcher is asking. Tethys is a metadata system for spatial-temporal acoustic data that provides structure where it is appropriate and flexibility where it is needed. Networked metadata are stored in an extended markup language (XML) database, and served to workstations over a network. The ability to export summary data to OBIS-SEAMAP is in development. The second purpose of Tethys is to serve as a scientific workbench. Interfaces are provided to networked databases, permitting the import of data from a wide variety of sources, such as lunar illumination or sea ice coverage. Interfaces currently exist for MATLAB, JAVA, and PYTHON. Writing data driven queries using a single interface enables quick data gathering from multivariate sources to address hypotheses. Examples showing the results of analysis of acoustic data from acoustic deployment from 26 sites across the Northern Pacific will be shown.

3:15–3:40 Break

Invited Papers

3:40

4pAB7. Long-term bioacoustic monitoring for gauging animal populations: An approach involving flight calls of night migrating songbirds. William Evans (Old Bird Inc., 605 W. State St., Ithaca, NY 14850, admin@oldbird.org)

At times during the course of a year, the airspace over most locations in North America contains flight calls of night migrating songbirds. The calls typically have an audio frequency between 2 and 10 kHz and are 0.03-0.4 s in duration. Documenting such calls in a consistent manner enables temporal and quantitative calling patterns to be determined. Theoretically, such acoustic data gathered over time could be used as an index to population change as well as for documenting shifting migration routes. This presentation will discuss the development of a multi-sensor system designed to synchronously sample nocturnal flight calls of migrating songbirds across eastern North America. We will review the decisions involved with the system design that minimize non-target aspects of the soundscape and that help standardize monitoring over time. We will then illustrate the monitoring power of this application with flight call data from ten fall migrations at one monitoring station and two fall migrations from a transect of ten monitoring stations across eastern North America.

4:00

4pAB8. Acoustic monitoring of breeding amphibians at Yosemite National Park and Point Reyes National Seashore. Patrick Kleeman, Gary Fellers (US Geological Survey, 1 Bear Valley Rd., Point Reyes, CA 94956, pkleeman@usgs.gov), and Brian Halstead (US Geological Survey, Dixon, CA)

The calling behavior of frogs and toads at breeding sites lends itself to acoustic monitoring of these amphibian populations. We are using Automated Recording Devices (ARD) at two National Park units in California, Yosemite National Park and Point Reyes National Seashore, to monitor the breeding efforts of amphibians by recording their calls. We are monitoring both common species (Pacific chorus frogs, *Pseudacris regilla*, occurs in both parks) and imperiled species (Yosemite toad, *Anaxyrus canorus*, Yosemite; California red-legged frog, *Rana draytonii*, Point Reyes) to investigate whether breeding phenology will shift with changing climatic conditions. The use of ARD is also providing a more complete picture of the diel calling patterns of these species, and how some species are partitioning their acoustical environment by frequency and time in order to breed successfully while surrounded by noisy neighbors. Information gathered through acoustic monitoring is very valuable for conserving rare amphibians and ensuring that common species remain common.

Contributed Papers

4:20

4pAB9. Determining the contribution of cetacean noise to the marine soundscape of Australia's Northwest Shelf. Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au)

The continental shelf off Western Australia is 500 km wide and has an annual mean sea surface temperature of 28 degrees Celsius. Biodiversity is rich and includes at least 45 cetacean species (whales and dolphins). A catalog of sounds produced by these animals was established based on a literature review and recordings obtained by the Centre for Marine Science and Technology at Curtin University. An automatic detector for these sounds was developed and includes the following steps: (1) Fourier transformation of the recorded time series, (2) spectrogram normalization, (3) computation of information entropy of the spectrogram, (4) investigation of entropy distribution and thresholding, and (5) removal of detections with fewer than a predefined number of spectrogram pixels. Detector performance was assessed by comparison to manual detections. The total energy of the biological sounds detected was computed to determine the contribution of cetaceans to the underwater noise budget at several locations and times of year. [Work supported by Chevron Australia.]

4:35

4pAB10. Spatiotemporal variability in coral reef soundscapes in St. John, U.S. Virgin Islands. Maxwell B. Kaplan, T. Aran Mooney (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS50, Woods Hole, MA 02543, mkaplan@whoi.edu), and Jim Partan (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Passive acoustic measurements of coral reef "soundscapes" can be an effective way of tracking biological activity and may help assess community-level biotic diversity. While a reef soundscape may vary both temporally and spatially, this variability is often not well understood. To investigate this, we deployed multiple digital acoustic recorders (DMONs) for both short- (24-h) and long-term (4 months) investigations at three patch reefs that varied in coral cover (low, intermediate, and high levels) in the U.S. Virgin Islands National Park (sample rate: 120 kHz). The short-term investigation consisted of four continuously recording instruments spaced at ~20 m intervals. Long-term measures included two recorders per reef on a duty cycle of 2.5 min/2 h. Fish and coral diversity, ambient light intensity, temperature, and salinity were also measured. Results indicate diel patterns in snapping shrimp signals (dominant energy between 2.5 and 20 kHz) with peaks at dusk and dawn. Sound pressure level (SPL) of the snapping shrimp band varied spatially within and among reefs, with higher maximum SPL at reefs with low and intermediate coral cover. However, within-reef SPL variability was lowest at the site with high coral cover. Temporal patterns in snapping shrimp acoustic activity were correlated within and among all three reefs.

4:50

4pAB11. The ambient acoustic environments at two locations in Laguna San Ignacio, Baja, Mexico. Kerri Seger (Scripps Inst. of Oceanogr., Univ. of California San Diego, 4090 Rosenda Ct, Unit 199, San Diego, CA 92122, kseger@ucsd.edu), Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Steven Swartz (Laguna San Ignacio Ecosystem Sci. Program, Darnestown, MD), and Jorge Urban (Laboratorio de Mamíferos Marinos, Universidad Autonoma de Baja California Sur, La Paz, Mexico)

Each winter gray whales (*Eschrichtius robustus*) breed and calve in Laguna San Ignacio, with the lagoon's northern section more heavily used by mothers rearing calves. The southern section of the lagoon is open to milling and ecotourism traffic, while the northern section is restricted to vessel transits only. Ambient acoustic data from autonomous underwater recorders have been collected between 2008 and 2013 at Punta Piedra (southern section) and Camp Kuyima (northern section). Multiple sources of acoustic sound exist in the lagoon, including tidal flows, fish chorusing, gray whale vocalizations, snapping shrimp, daily land/sea breezes, and panga activity. Here the cumulative distributions of rms sound pressure levels from all deployments during daytime and nighttime are presented for several frequency bands that represent contributions from the varying source mechanisms. Since concurrent data from both restricted (northern) and unrestricted (southern) sections exist, comparing sound level distributions between sites can provide insight into the relative contributions of various mechanisms to the overall ambient noise environment. These data have established a baseline for monitoring trends and changes in acoustic environments of the lagoon in anticipation of future tourist development. [Work sponsored by LSIESP and Ocean Foundation.]

Session 4pAO

Acoustical Oceanography and Animal Bioacoustics: Properties, Trends, and Utilization of Ocean Noise II

Jennifer L. Miksis-Olds, Cochair
Appl. Res. Lab., Penn State, PO Box 30, M.S. 3510D, State College, PA 16804

Zoi-Heleni Michalopoulou, Cochair
Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102

Chair's Introduction—1:55

Invited Paper

2:00

4pAO1. Soundscapes from hydrophone stations in the comprehensive nuclear-test-ban treaty organisation's international monitoring system hydroacoustic network. Mark K. Prior and David Brown (IDC/SA, Comprehensive Nuclear-Test-Ban Treaty Organisation, PO Box 1200, Vienna 1400, Austria, mark.prior@ctbto.org)

The Comprehensive Nuclear-Test-Ban Treaty Organisation operates a global network of sensors that includes cabled sound-channel hydrophones in the Atlantic, Pacific and Indian Oceans. Hydrophones are deployed in groups of three, known as triads, so that the arrival times and azimuths of signals can be obtained. Data are recorded at frequencies up to 100 Hz with continuous acquisition and data relay via satellite connection to CTBTO's International Data Centre. Signals from distant earthquakes, underwater explosions, marine mammals and ice-breaking are routinely detected and an extensive archive has been built up over the last decade. To understand sensor detection performance, high-level summaries of noise properties are required to establish the "acoustic context' for each station. These "soundscapes" allow the identification of source types that dominate in specific frequency bands. Examples signals are illustrated and information regarding the sources of persistent signals is extracted.

Contributed Papers

2:20

4pAO2. The marine soundscape of the Fram Strait. Hanne Sagen, Hans Kristian Tengesdal, Mohamed Babiker (Nansen Environ. and Remote Sensing Ctr. (NERSC), Bergen, Norway), and Dag Tollefsen (Norwegian Defence Res. Establishment (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no)

The marine soundscape of the Fram Strait has been subject to investigations since the mid 1980's. Increasing interest in Arctic operations has initiated a recent series of acoustic experiments that includes synoptic ambient noise measurements in the Marginal Ice Zone conducted over the years 2010–2012. This presentation will give an overview of these experiments, then focus on results from measurements made with sonobuoys under varying ice and environmental conditions in the MIZ. Noise spectra (20 Hz—2 kHz) are presented, discussed, and compared with historical data from 1985 to 1987. Spectra are categorized by environmental parameters including wind force and direction as derived from numerical models, ice concentration derived from satellite images, ocean wave properties from a coupled ice-ocean prediction model, and sound propagation conditions inferred from the ice-ocean model. The contributions to this soundscape that will be quantified and discussed include open-ocean wind-generated noise, ice floe collision, marine mammals, and seismic exploration activity.

2:35

4pAO3. Using an autonomous underwater vehicle to track the changing arctic ambient noise field in real time. Stephanie Fried and Henrik Schmidt (Massachusetts Inst. of Technol., 77 Massachusetts Ave, Rm 5-204, Cambridge, MA 02139, eowyn@mit.edu)

The ambient noise field, particularly the directionality of the noise, can provide a wealth of information about the local environment. Changes in the ambient noise field often reflect changes in the physical environment. Accurate calculation of the noise field, though, can be a challenge. Because of their maneuverability autonomous underwater vehicles (AUVs) provide novel capabilities not only for measuring and analyzing the local noise field, but also for continuous tracking of changes to the noise field and thus the environment. Of interest is the measurement and analysis of the ambient noise in arctic environments. By integrating models for arctic ambient noise into an AUV simulation, this paper analyzes the use of AUVs in real-time autonomous tracking of the three-dimensional changing arctic ambient noise field.

2:50

4pAO4. Interdecadal trends in ocean ambient sound at seven sites in the northern Pacific Ocean basin. Rex K. Andrew (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98040, rex@apl.washington.edu), Bruce M. Howe (Ocean & Res. Eng., Univ. of Hawaii-Manoa, Honolulu, HI), and James A. Mercer (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

A long-term observation program begun in approximately 1994 has amassed time-series of ambient sound short-time spectra from omni-directional hydrophones deployed on the ocean floor at seven locations in the northern Pacific Ocean. Each time-series consists of spectra estimated every 5 min over a useful band of 10–500 Hz, thereby encompassing the vocalizations of baleen whales, the anthropogenic contribution of ship traffic noise, and wind/wave noise due to sea surface processes. Simple linear trend lines show that traffic noise in the northern and northwestern reaches of the Pacific has increased by as much as 3 to 4 dB during this program. In the eastern North Pacific, however, the ambient sound shows a decrease of about 3 dB. The number of ships in the world merchant fleet increased by approximately 25% over this period. This change is insufficient to explain the increases in noise levels along northern and northwestern regions, and provides no explanation for the decreases observed in the north-east. The traffic noise field is evidently dependent on more complex temporal and geographical patterns of shipping traffic. [Work supported by ONR.]

Invited Paper

3:05

4pAO5. Quantifying ocean noise and its spatiotemporal variability on Australia's Northwest Shelf. Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au), Alexander Gavrilov, and Robert McCauley (Ctr. for Marine Sci. & Technol., Curtin Univ., Perth, WA, Australia)

The Northwest Shelf is an extensive oil and gas region off Western Australia. The Centre for Marine Science and Technology at Curtin University has recorded underwater noise in this region for 14 years on behalf of industry and government. Under the Collaborative Environmental Research Initiative (CERI), this data is being shared and synthesized to quantify the marine soundscape and to describe spatiotemporal variability. Automatic software analysis tools were developed to process the data. Power spectrum density percentiles were computed for all sites on a monthly basis, and compared. Distinct spectral features were identified. Factors contributing to the observed spatiotemporal variability ranged from long-term offshore oil and gas installations to fish choruses. [Work supported by Chevron Australia.]

Contributed Papers

3:25

4pAO6. Wind dependence of shallow water ambient noise in a biologically rich temperate coastal area. Delphine Mathias (GIPSA-Lab, 11 rue des Mathématiques, Saint Martin d'Hères 38402, France, delphine.mathias@gmail.com), Cedric Gervaise, and Lucia Di Iorio (GIPSA-Lab, Chair Chorus Grenoble INP Foundation, Saint Martin d'Hères, France)

The Iroise Marine Natural Park, created in 2007, is the first French natural marine park. This archipelago located in Western Brittany is a shallow water area that comprises 11 islands and hosts a rich variety of marine life, including seaweed fields, benthic organisms, endangered seals, and cetaceans. Three underwater autonomous recorders were moored at 10-m depth and sampled at 32 kHz from June 2011 to November 2011. Here we report on the dependency of shallow water ambient noise level on wind speed in a biologically rich environment. First we extract the ambient noise level in presence of transient sounds produced by benthic organisms by removing instantaneous sound pressure levels higher than a threshold computed using the kurtosis of the raw 10 sec time series. We then show that the ambient noise level allows to extract environmental information such as wind speed and biological rhythms, and that both are explaining 90% of its variance. Dependence of ambient noise and ocean noise level to wind speed at several frequencies are compared to reference work by Wenz (1962) and previous shallow water studies. Finally, we discuss how data assimilation coupling measured ambient noise and environmental parameters can help monitor marine ecosystems.

3:40

4pAO7. Correlating acoustical with physical and biological oceanography. Iain Parnum (Ctr. for Marine Sci. & Technol., Curtin Univ., Perth, WA, Australia), Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au), and Arti Verma (Ctr. for Marine Sci. & Technol., Curtin Univ., Perth, WA, Australia)

The Centre for Marine Science & Technology at Curtin University built and maintains the underwater acoustic recorders of Australia's Integrated Marine Observing System (IMOS; http://IMOS.org.au). Recordings have been obtained at four locations (off Western Australia, Victoria, and New South Wales) since 2011. IMOS includes a multitude of oceanographic and remote sensors, contributed by various institutions, which are also responsible for data management. Data are shared and publicly available encouraging collaboration and syntheses. This study has compiled time series of weather data, tides, current data (from Acoustic Doppler Current Profilers, ADCP), and wind (from radar measurements), and established correlations with underwater noise is a series of one-third octave bands between 10 Hz and 3 kHz from the Perth Canyon. Our results further demonstrate that ocean noise in certain frequency bands can be used to estimate aspects of physical and biological oceanography.

3:55

4pAO8. Undersea noise characterization using vector sensors and adaptive particle filters. Dennis Lindwall (Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, lindwall@bellsouth.net), Don DelBalzo (Marine Information Resource Corp., Ellicot City, MD), Dimitrios Charalampidis (Elec. Eng., Univ. of New Orleans, New Orleans, LA), Jim Leclere, E. J. Yoerger, and George Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

Acoustic noise in the ocean is spatially complex and dynamic. It results from a composite of moving discrete and distributed sources with narrow and broadband signatures from near and distant locations over complex propagation paths. Whether the goal of noise characterization is to improve understanding of the sources and environmental aspects of noise or to find and describe weak signals, it can be better achieved with vector acoustic data and vector-based analysis rather than with a pure scalar-pressure approach. We will show analytically and with realistic acoustic noise simulations that the vector characterization of noise parameters such as directional distribution, frequency content, and time variation is more accurate and can be done with simpler instrumentation than what is commonly done with pressure data. We employ adaptive particle filters to model and estimate the temporal aspects of the noise fields. The filter's state and observation vectors consist of signal directions and magnitudes, respectively. As a result, the noise fields are directly associated with the filter's process and observation noise.

4:10

4pAO9. Ocean wave seismic and acoustic noise detected with distributed fiber optic sensor array. Kent K. Hathaway (Coastal & Hydraulics Lab., US Army Engineer Res. & Development Ctr., 1261 DC Rd., Kitty Hawk, NC 27949, Kent.K.Hathaway@usace.army.mil), Richard D. Costley, Eric Smith, Troy Milburn, and Jennifer R. Picucci (GeoTech. & Structures Lab., U.S. Army Engineer Res. & Development Ctr., Vicksburg, MS)

The U.S. Army Engineer Research & Development Center installed a Fiber Optic Sensor System (FOSS) at the Field Research Facility (FRF) in Duck, NC as a test-bed for evaluating FOSS performance in measuring ambient acoustic and seismic noise in a coastal environment. The sensor system consists of a buried fiber optic cable with a length of approximately 3 km, which wends its way through the sand dunes and along the beach. An optical interrogator contains a pulsed laser which injects light into an optical fiber within the cable and receives Rayleigh backscattered signals from it. The system, which is able to detect and locate seismic activity along the entire length with a 10 m resolution, is being used to study ocean wave generated noise. The FRF also maintains and operates a real-time cross-shore directional wave array, consisting of five bottom-mounted acoustic wave gauges installed at 2 to 11 m depths. The main focus of the work presented here compares FOSS data with directional wave measurements made with the cross-shore array under a variety of conditions. Data collected with other sensor systems (e.g., vertical long-period seismometer, infrasound microphone, anemometers, rain gauges, high resolution video) were also used in this comparison.

THURSDAY AFTERNOON, 5 DECEMBER 2013 GOLDEN GATE 2/3, 1:00 P.M. TO 6:00 P.M.

Session 4pBA

Biomedical Acoustics: Recent Advances in Therapeutic Ultrasound II

Tatiana D. Khokhlova, Cochair
Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Oleg Sapozhnikov, Cochair
Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation

Contributed Papers

1:00

4pBA1. Laparoscopic high intensity focused ultrasound for the treatment of soft tissue. Narendra T. Sanghvi and Adam Morris (R & D, SonaCare Medical, 4000 Pendleton Way, Indianapolis, IN 46226, narensanghvi@sonacaremedical.com)

There is a growing demand to perform focal surgery, particularly for prostate and kidney tumors. RF and Cryo ablation are used for treatment of tumors; however, there are reported complications mainly skipping of cancer cells and bleeding. It has been shown that HIFU can provide acoustic hemostasis to overcome such complications. However, HIFU must compete with these modalities with improved treatment efficiency and the size of the applicator must fit in a 12 mm trocar/port used during the laparoscopic surgery. We modified the Sonatherm-600i applicator and transducer that generates a split beam HIFU at 4 MHz with a center element for imaging operating at 6.5 MHz. The transducer is integrated in a robotic probe that renders bi-plane images for tissue localization and ablation. Based on selected tissue ablation volume, treatment trajectory is generated by the computer to provide optimum thermal dose to result in complete coagulative necrosis. The treatment is conducted using continuous HIFU with an interlaced imaging for tissue change monitoring. The ablation efficiency of this device is @ 1 cc/min that is better than Cryo and RF ablation. Results will be presented from recent animal studies.

1:15

4pBA2. Model-based feasibility assessment and evaluation of prostate hyperthermia with a commercial MRI-guided endorectal high intensity focused ultrasound ablation array. Vasant A. Salgaonkar (Radiation Oncology, Univ. of California San Francisco, 396 Ano Nuevo Ave., Apt. 106, Sunnyvale, CA 94085, salgaonkarv@radonc.ucsf.edu), Viola Rieke, Eugene Ozhinsky (Radiology and Biomedical Imaging, Univ. of California San Francisco, San Francisco, CA), Punit Prakash (Elec. and Comput. Eng., Kansas State Univ., Manhattan, KS), Juan Plata (Radiology, Stanford Univ., Stanford, CA), John Kurhanewicz (Radiology and Biomedical Imaging, Univ. of California San Francisco, San Francisco, CA), I-C (Joe) Hsu, and Chris J. Diederich (Radiation Oncology, Univ. of California San Francisco, San Francisco, CA)

Numerical simulations were conducted to devise methods for targeted and protracted hyperthermia (40–46 °C, 30–60 min) to the prostate with a commercial MR-guided endorectal ultrasound phased array (2.3 MHz, ExAblate, InSightec). The intention is to fast-track clinical implementation of this FDA approved ablation system for delivering targeted hyperthermia in conjunction with radiation or chemotherapy. Conformable hyperthermia to focal tumors in posterior and hemi-gland prostate was simulated through 3D patient-specific biothermal models and beamformed acoustic patterns that incorporated the specific constraints imposed on the ExAblate array:

irregular element spacing, switching speeds, operating power and short pulse duration. Simulations indicated that diverging and iso-phase sonications could treat ($T > 41\,^\circ$C, $_{max} < 46\,^\circ$C) 13–23 cm^3 with ~1.1 W/cm^2, multi-focused patterns could treat 4.0 cm^3 with 3.4 W/cm^2, and curvilinear patterns could treat 6.5 cm^3 with 0.8 W/cm^2 while avoiding rectum, urethra, pubic bone, *etc*. Custom beamforming identified through simulations was implemented on the ExAblate system and sonications were performed in tissue mimicking phantom material. ExAblate delivered long duration sonications (0.86 W/cm^2) with these customized beamforming patterns and generated diffuse hyperthermia ($\Delta T = 4$–$8\,^\circ$C) in phantom, monitored with real-time multi-slice MR temperature imaging (3T). [NIH R01CA122276, Focused Ultrasound Foundation.]

1:30

4pBA3. Ultrasound-mediated remote actuation of implantable devices for localized drug delivery. Parag V. Chitnis (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, pchitnis@riversideresearch.org), Olga Ordeig, and Samuel K. Sia (Dept. of Biomedical Eng., Columbia Univ., New York, NY)

Direct local delivery of therapeutics can significantly improve long-term outcome and quality of life for cancer patients. We present a drug-delivery approach that employs focused ultrasound (FUS) for remotely actuating drug-loaded, biocompatible implants consisting of porous NiPAAm hydrogels encapsulated in a PDMS disk. The hydrogels were 1 mm thick and 6 mm in diameter. The NiPAAm formulation was designed to contract to 30% of original size when heated to 45°C. The capsule was loaded with 20-kDa TRITC-Dextran and placed in a custom-designed PDMS chamber containing de-ionized water. A thermocouple was embedded in the NiPAAm gel to monitor local temperature. TRTIC-dextran released to the surrounding media was quantified by absorbance at 540/580 nm. Maintaining the capsules at 37°C for two days using a hot plate did not trigger release. A 1.5-MHz FUS transducer operating at low intensities (<500 W/cm^2) elevated the gel temperature to 45°C in 32.6 ± 19 s ($N = 10$). Thermocouple was employed as a feedback to modulate (on/off) FUS in real-time to maintain gels at 45°C for 10~min, which resulted in a release of 10.6 ± 0.3~μg of dextran. Capsules were then implanted in eight mice; four mice were subjected to FUS. FUS actuated release *in vivo* as evidenced by fluorescence imaging.

1:45

4pBA4. High intensity focused ultrasound laparoscopic instrument for partial nephrectomy. Stuart Mitchell, Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, sbmitch@apl.washington.edu), Jonathan Harper, Ryan Hsi (Urology, Univ. of Washington Medical Ctr., Seattle, WA), and Lawrence Crum (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Partial nephrectomy (PN) is the gold standard for small clinically localized renal masses because of equal oncologic outcomes and greater preservation of renal function compared with radical nephrectomy (RN). However, it is a complex operation due to the challenges of cutting into a well-vascularized organ and the need for reconstruction of the remaining kidney following excision. PN is associated with higher blood loss, risk of transfusion, and longer operative time compared to RN. High intensity focused ultrasound (HIFU) affords the ability to ablate tissue and perform hemostasis, thus potentially mitigating some of the challenges associated with PN. The purpose of this paper is to introduce a new HIFU clamp as an adjunctive tool for PN. A HIFU device was created to conform to the shape of a commonly used laparoscopic instrument. Characterization studies were conducted using *ex vivo* tissue. Histology was performed to evaluate thermal damage. *Ex vivo* studies indicated that complete ablation planes could be achieved at temperatures sufficient for thermal tissue necrosis. Gross parenchymal changes were observed with clear demarcation between treated and untreated regions. Histological evaluations revealed that there were no viable cells in ablated regions. [This work was funded by NIH (EB013365).]

2:00

4pBA5. Pulsed focused ultrasound treatment of muscle mitigates paralysis-induced bone loss in the adjacent bone: A study in a mouse model. Sandra L. Poliachik (Dept. of Radiology, Seattle Children's Hospital, Seattle, WA), Tatiana D. Khokhlova, Yak-Nam Wang, Julianna C. Simon (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, tanyak@apl.washington.edu), Ted S. Gross (Dept. of Orthopaedics, Univ. of Washington, Seattle, WA), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA)

Bone loss can occur following bed rest, space flight, spinal cord injury, or age-related hormonal changes. The treatment methods for this condition include pharmaceutical interventions and exercise, neither of which is particularly effective. Other technologies include low intensity pulsed ultrasound targeted to the bone, used previously to enhance fracture healing, and whole body vibration. This study attempted to mitigate paralysis-induced bone loss indirectly, by applying pulsed focused ultrasound (pFUS) to the midbelly of a paralyzed muscle. We employed a mouse model of disuse that utilizes onabotulinumtoxin A, which induces rapid bone loss in 5 days. The pFUS treatments were performed daily for four consecutive days following paralysis. A spherically focused 2-MHz transducer produced 5-microsecond pulses at pulse repetition frequency mimicking motor neuron firing rates during walking (80 Hz) or standing (20 Hz). Two different power levels were used corresponding to peak positive focal pressures of 30 and 18 MPa. The trabecular bone changes were characterized using micro computed tomography. Our results indicated that application of pFUS at pulse repetition frequency of 20 Hz and lower amplitude setting successfully mitigated paralysis-induced bone loss. The targeted muscle tissue did not display any sign of injury. [Work supported by CDMRP SCIRP (SC090510).]

2:15

4pBA6. Optimization of parameters for therapeutic applications of high intensity myocardial contrast echocardiography. Douglas Miller, Chunyan Dou (Radiology, Univ. of Michigan, 3240A Medical Sci. I, 1301 Catherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu), Gabe E. Owens (Pediatric Cardiology, Univ. of Michigan, Ann Arbor, MI), and Oliver D. Kripfgans (Radiology, Univ. of Michigan, Ann Arbor, MI)

High intensity myocardial contrast echocardiography (HI-MCE) can lethally injure cardiomyocytes leaving scattered microlesions. This cavitational bioeffect may be of value for graded tissue-reduction therapy for conditions such as hypertrophic cardiomyopathy. Anesthetized rats in a heated water bath were treated with 1.5 MHz focused ultrasound, which was guided by an 8 MHz diagnostic imaging probe. Eight-pulse bursts were triggered intermittently over 5 min at approximately end systole during contrast microbubble infusion. The relative efficacy between 2 MPa (~173 W/cm^2 I_{PA}) or 4 MPa (~892 W/cm^2) pulses, 1:4 or 1:8 trigger intervals, and 5 or 10 cycle pulses was explored in 6 groups. ECG premature complexes (PCs) induced by the triggered pulse bursts were counted, and microlesions assessed in Evans blue-stained cardiomyocyte scores (SCSs). The increase from 2 to 4 MPa produced significant increases in PCs and SCSs. In addition, the higher pressure eliminated the decline in the rate of PC induction over time, which was seen at 2 MPa and likely hindered therapeutic efficacy. Neither increased trigger intervals nor pulse durations yielded significant increases in therapeutic effects. High concentrations of microlesions were readily produced, which suggest that HI-MCE can be refined into a clinically robust method for therapeutic myocardial reduction.

2:30

4pBA7. Delivery of different-size molecules by ultrasound-induced blood-brain barrier opening and its correlation with acoustic emission. Hong Chen, Anushree Srivastava, Tao Sun, Oluyemi Olumolade, and Elisa E. Konofagou (Dept. of Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, hc2666@columbia.edu)

Focused ultrasound (FUS) in combination with microbubbles (MBs) has been successfully used in the delivery of therapeutic agents of various sizes through the blood-brain barrier (BBB) in preclinical studies. However, the dependence of delivery efficiency on the drug molecular size calls for further exploration. Fluorescence-labeled dextrans of molecular weights of 3, 70, 150, and 2000 kDa were used as model therapeutic compounds. Dextrans were mixed with MBs and injected intravenously to mice immediately after the onset of FUS sonication. The acoustic emission from ultrasound-activated MBs was acquired passively using a 10 MHz transducer and quantified for stable, inertial, and total cavitation doses. The drug delivery efficiency was quantified by the relative delivery amount and volume estimated based on fluorescent images of the brains. It was found that dextran of 3 kDa can be delivered trans-BBB at a pressure level below the inertial cavitation threshold; however, dextrans of 70, 150, and 2000 kDa were delivered only when the pressure was above the inertial cavitation threshold. At the same pressure level, the amount and volume of dextrans delivered decreased as the dextran size increased. A linear correlation of total cavitation dose and the fluorescence enhancement was found for each size dextran.

2:45

4pBA8. Synergy between high intensity focused ultrasound and ethanol injection in ablation of thyroid cancer cells. Hakm Murad (Biomedical Eng., TuLn. Univ., 108 Cottonwood Dr., Gretna, LA 70056, hmurad@tulane.edu), Nguyen H. Hoang, Sithira H. Ratnayaka (Biomedical Eng., TuLn. Univ., New Orleans, LA), Koji Tsumagari, Emad Kandil (Surgery, TuLn. Univ. School of Medicine, New Orleans, LA), and Damir Khismatullin (Biomedical Eng., TuLn. Univ., New Orleans, LA)

We have investigated the combination of high intensity focused ultrasound (HIFU) and ethanol injection for ablation treatment of anaplastic thyroid cancer, a highly aggressive form of thyroid cancer characterized by >80% mortality within months. The suspension of FB1 anaplastic thyroid cancer cells (100 μl, 2.7 million cells/ml) were placed in a 0.2 ml thin-wall PCR tube and then exposed to HIFU alone, ethanol alone, or ethanol + HIFU. The focused ultrasound signal was generated by a 1.1 MHz transducer with acoustic power ranged from 4.1 W to 12.0 W. Ethanol was diluted in the FB1 cell growth medium to the concentration of 2%, 4%, or 10% (v/v) and applied to the cells before HIFU exposure. The viability of the cells was measured by flow cytometry and trypan blue exclusion 2, 24, and 72 h post-treatment. The exposure of FB1 cells to HIFU alone greatly reduced the number of viable cells immediately after treatment; however, their proliferation rate remained high. On the other hand, both the viability and proliferation rate significantly decreased in the cells treated with both ethanol and HIFU. In conclusion, percutaneous ethanol injection (PEI) and HIFU have a synergistic effect on anaplastic thyroid cancer ablation.

3:00

4pBA9. High intensity focused ultrasound ablation of ethanol-treated liver tissues and cancer cells. Nguyen H. Hoang (Biomedical Eng., TulaneUniv., 5243 Beaucaire St., New Orleans, LA 70129, nhoang@tulane.edu), Hakm Y. Murad (Biomedical Eng., TulaneUniv., Gretna, LA), Sithira H. Ratnayaka, Chong Chen, and Damir B. Khismatullin (Biomedical Eng., TulaneUniv., New Orleans, LA)

We have investigated the combined effect of HIFU and ethanol injection on the temperature rise and cavitation in porcine liver tissues and on the viability and proliferation rate of HepG2 liver cancer cells. Tissues were injected with 95% ethanol before being subjected to the HIFU beam generated by a 1.1 MHz transducer with acoustic power ranged from 1.17 W to 20.52 W. Cavitation events and the temperature in and around the focal zone were measured by a passive cavitation detector and type K thermocouples, respectively. In the cell study, 100 μl of HepG2 cell suspension (2.7 million cells/ml) was placed in a 0.2 ml thin-wall PCR tube. Ethanol 2% or 4% in the cell growth medium was added to the cell suspension, and the cells were then exposed to HIFU for 30 s. The data of these experiments show that the pre-treatment of tissues with ethanol reduces the threshold power for inertial cavitation and increases the temperature rise. Both the viability and proliferation rate were significantly decreased in cells treated with ethanol and HIFU, as compared to individual treatments. The results of our study indicate that ethanol injection and HIFU have a synergistic effect on liver cancer ablation.

3:15–3:30 Break

3:30

4pBA10. High intensity focused ultrasound for *Enterococcus faecalis* biofilm. Siew-Wan Ohl (Fluid Dynam., Inst. of High Performance Computing, 1 Fusionopolis Way, #16-16 Connexis North, Singapore 138632, Singapore, ohlsw@ihpc.a-star.edu.sg), Kulsum Iqbal (Discipline of Prosthodontics, Operative Dentistry and Endodontics, National Univ. of Singapore, Singapore, Singapore), Boo Cheong Khoo (Dept. of Mech. Eng., National Univ. of Singapore, Singapore, Singapore), Jennifer Neo (Discipline of Prosthodontics, Operative Dentistry and Endodontics, National Univ. of Singapore, Singapore, Singapore), and Amr Sherif Fawzy (Discipline of Oral Sci., National Univ. of Singapore, Singapore, Singapore)

High intensity focused ultrasound (HIFU) is used to removal *Enterococcus faecalis* (*E. faecalis*) in planktonic suspension and dental biofilm. The bacteria *E. faecalis* is commonly found in secondary dental infection after root canal treatment. Sealed petri dish with *E. faecalis* planktonic suspension is placed at the focal region of the bowl-shaped HIFU transducer of 250 kHz in a water bath. It is subjected to sonification of 30 to 120 s. It is found that the HIFU successfully lysed and removed the bacteria from counting its colony forming units (CFU), performing scanning electron microscopy (SEM) and confocal microscopy. Also, *E. faecalis* biofilms in human teeth are subjected to the same HIFU treatment. Similar analysis is performed with SEM and confocal microscopy. It is found that after 60 s of sonification, most of the biofilm is either removed or lysed. In conclusion, this study highlights the potential of using HIFU as non-destructive dental root canal disinfection treatment.

3:45

4pBA11. Accurate quantification and delivery of thermal dose to cells in culture. Elly Martin, Adam Shaw, Nilofar Faruqui, and Michael Shaw (Acoust. and Ionizing Radiation Div., National Physical Lab., Hampton Rd., Teddington, Middlesex TW11 0LW, United Kingdom, adam.shaw@npl.co.uk)

HIFU treatments involve raising the temperature of target tissue above 60°C in short (~2 s) bursts. At higher temperatures, shorter times are required to induce a given deleterious effect: the Sapareto-Dewey thermal dose equation is often used to relate the time to produce a biological effect at one temperature to the time to produce equivalent effects at another. A heating chamber was developed to deliver controlled thermal doses to cells in culture under continual observation by differential interference contrast microscopy. The system comprised of a cell culture well and cover slip coated with a transparent electrode inserted into a microscope stage with electrical contacts. Thermal doses were delivered by applying programmed current-time profiles and using a PID controller to rapidly raise and maintain the temperature of the chamber above 37°C while monitoring with fine wire thermocouples. Initially, HeLa cells in monolayer culture were imaged before, during, and after heating. Visible changes in cell shape and adhesion began shortly after raising the temperature by 8°C and progressed during a heating period of 20 min, continuing for more than 12 h after the cells were returned to 37 °C. No such changes were observed in control cells. Results will be presented exploring the validity of the S-D relationship for shorter, higher temperature exposures.

4:00

4pBA12. Thermal lesion imaging using Lorentz force: Proofs of concept. Pol Grasland-Mongrain, Stefan Catheline (LabTAU, INSERM U1032, 151 Cours Albert Thomas, Lyon 69424, France, jean-yves.chapelon@inserm.fr), Jean-Martial Mari (Imperial College, London, France), Rémi Souchon, Ali Zorgani, Alexandre Petit, Florian Cartellier, Cyril Lafon, and Jean-Yves Chapelon (LabTAU, INSERM U1032, Lyon, France)

The Lorentz force can be used by different means to image thermal lesions in biological tissue. In the first method presented here, so-called magneto-acoustical electrical tomography, a tissue sample is held in a magnetic field and is subsequently exposed to a focused ultrasound beam. The displacement within the magnetic field caused by this ultrasound beam results in an electrical current due to the Lorentz force. In this way, the change in electrical conductivity due to the presence of thermal lesions can then be observed. Conversely, when an electrical current is applied to tissue placed in a magnetic field, a shear wave is induced by the Lorentz force.

Elastography images can be reconstructed from this shear wave, revealing thermal lesions by the change in elastic modulus. The first method was tested on *ex-vivo* chicken breast sample with a 500 kHz transducer and a 300 mT magnetic field. The second method was tested on gelatin phantom with a 100 mA current and 300 mT magnetic field. Images and results will be presented for both methods. These techniques could be used for the monitoring of thermal lesion formation in high intensity focused ultrasound treatment.

4:15

4pBA13. Prediction of the reversibility of the ultrasound-induced blood-brain barrier opening using passive cavitation detection with magnetic resonance imaging validation. Tao Sun, Gesthimani Samiotaki, and Elisa E. Konofagou (Dept. of Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, ts2765@columbia.edu)

Various molecules have been shown to cross the blood-brain barrier (BBB) upon exposure to focused ultrasound combined with microbubbles and exhibit therapeutic effects. Real-time monitoring, thereof, remains one of the key elements before clinical translation of this technique. The dependence of acoustic emissions on the closing timelines of the BBB opening volume and its permeability was investigated under different pressures (0.30, 0.45, and 0.60 MPa) and microbubble sizes (diameters: 1–2, 4–5, or 6–8 μm). A 10-MHz passive cavitation detector was used to acquire cavitation signals during sonication at the mouse right hippocampus ($n = 45$). Contrast-enhanced dynamic and T1-weighted MR scans were performed immediately after sonication and up to 6 days thereafter. Contrast-enhanced volumes and diffusion rates of the contrast agent were quantified as indicators for the BBB opening. It was found that the stable cavitation dose increased with the number of days required for closing while it reached a plateau after day 4. However, the inertial cavitation dose exhibited an exponential increase with the duration of the opening. A linear correlation between the total cavitation dose and BBB opening days was found. Moreover, the volume and permeability indicator K_{trans} were found to be both pressure- and bubble size-dependent. The dependence on the bubble-diameter and pressure allows us to predict and control the safety profile of this technique.

4:30

4pBA14. Rapid aberration correction for transcranial magnetic resonance-guided focused ultrasound surgery using a hybrid simulation and magnetic resonance-acoustic radiation force imaging method. Urvi Vyas and Kim Butts Pauly (Radiology, Stanford Univ., 1420 Guerrero St., San Francisco, CA 94110, urvivyas@stanford.edu)

Transcranial magnetic resonance-guided focused ultrasound surgery is a technique for causing tissue necrosis in the brain though the intact skull. Skull spatial and acoustic heterogeneities cause changes in the location, shape, and intensity of the focus. Current techniques use computed tomography (CT) imaging or MR-acoustic radiation force images (MR-ARFI) to correct these aberrations. CT-based techniques approximate acoustic parameters from Hounsfield units but suffer from co-registration concerns. MR-ARFI-based techniques use MR images as feedback to manipulate transducer phases, but require many image acquisitions (~4000) for one correction [Herbert, IEEE-TUFFC **56**(11)2388–2399]. We demonstrate here a hybrid technique that uses one MR-ARFI image to improve the focal intensity. The hybrid simulation-MR-ARFI technique used an optimization routine to iteratively modify the simulation aberrations to minimize the difference between simulated and experimental radiation force patterns. Experiments were conducted by applying skull-based aberrations to a 1024-element, 550 kHz phased-array transducer. The experimental MR-ARFI image of the aberrated focus was used with the simulation pattern from the hybrid angular spectrum [Vyas, IEEE-TUFFC **59**(6)1093–1100] beam propagation technique to estimate aberrations. The experiment was repeated three times. The hybrid simulation-MR-ARFI technique resulted in an average increase in focal MR-ARFI phase of 44%, and recovered 83% of the ideal MR-ARFI phase.

4:45

4pBA15. A non-axisymmetric, elongated pressure distribution in the lithotripter focal plane enhances stone comminution *in vitro* during simulated respiratory motion. Jaclyn M. Lautz, Georgy Sankin, Joseph Kleinhenz, and Pei Zhong (Mech. Eng. & Mater. Sci., Duke Univ., Sci. Dr., Durham, NC 27708, jaclyn.lautz@duke.edu)

A challenge in clinical shock wave lithotripsy (SWL) is stone translation due to a patient's respiratory motion, in a direction perpendicular to shock-wave propagation, which may negatively affect stone comminution while increasing the risk of tissue injury. We have developed a method using external masks and a modified lens geometry to transform the axisymmetric pressure distribution in the focal plane of an electromagnetic lithotripter into a non-axisymmetric elliptical distribution. At equivalent acoustic pulse energy (46 mJ), the peak pressure was reduced from 44 MPa to 38 MPa while the −6 dB focal width was increased from 7.4 mm for the original to 11.7 mm (major axis) and 7.9 mm (minor axis) of the modified field. *In vitro* stone comminution was performed in a tube holder ($d = 14$ mm) using a translation pattern with 12 breaths per minute and 15 mm in excursion distance. Stone comminution after 1000 shocks are $71.2 \pm 4.4\%$ and $65.2 \pm 8.3\%$ ($p < 0.05$) along the major- and minor-axis of the modified field, respectively, compared to $62.6 \pm 7.2\%$ for the original axisymmetric field. These results suggest that an elongated pressure field aligned along the direction of stone motion may enhance stone comminution in SWL. [Work supported by the NIH and the NSF GRFP.]

5:00

4pBA16. Shockwave tensile phase transmission depends on the gas concentration of the coupling medium. Spencer T. Frank (Mech. Eng., Univ. of California Berkeley, 1849 Cedar St., Apt. B, Berkeley, CA 94703, spencerfrank@berkeley.edu), Jaclyn Lautz, Georgy N. Sankin, Pei Zhong (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), and Andrew Szeri (Mech. Eng., Univ. of California Berkeley, Berkeley, CA)

Previous research shows that a shockwave's tensile phase can be strongly attenuated as a function of gas concentration in the coupling medium. Here, we seek to elucidate the relationship between tensile attenuation and gas concentration via pressure measurements at the focus and highspeed imaging. By performing *in vitro* experiments with water of varying gas concentrations (2.05 mg/L, 4.30 mg/L, and 6.50 mg/L), the negative impulsive pressure is correlated to the density of the bubble cloud that occurs in the beampath. It is found that for gas contents below 4 mg/L the bubble cloud remains sparse and the shockwave's tensile phase is successfully transmitted with no loss in impulsive pressure. For gas contents 4 mg/L and above the bubble cloud becomes highly dense and prevents transmission with up to a 75% loss in impulsive pressure. Corresponding stone comminution experiments show that the treatment efficiency sharply decreases with increasing gas concentration. These results underlie the importance of degassing the water used in the coupling medium before treatment.

5:15

4pBA17. Fragmentation of kidney stones *in vitro* by focused ultrasound bursts without shock waves. Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, 1013 NE 40th St., Seattle, WA 98105, amax38@u.washington.edu), Bryan W. Cunitz, Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Ryan S. Hsi, Mathew D. Sorensen, Jonathan D. Harper (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Shock wave lithotripsy (SWL) is the most common procedure for treatment of kidney stones. SWL noninvasively delivers high-energy focused shocks to fracture stones into passable fragments. We have recently observed that lower-amplitude, sinusoidal bursts of ultrasound can generate similar fracture of stones. This work investigated the characteristics of stone fragmentation for natural (uric acid, struvite, calcium oxalate, and cystine) and artificial stones treated by ultrasound bursts. Stones were fixed in position in a degassed water tank and exposed to 10-cycle bursts from a

200-kHz transducer with a pressure amplitude of $p \leq 6.5$ MPa, delivered at a rate of 40–200 Hz. Exposures caused progressive fractures in the stone surface leading to fragments up to 3 mm. Treatment of artificial stones at different frequencies exhibited an inverse relationship between the resulting fragment sizes and ultrasound frequency. All artificial and natural types of stones tested could be fragmented, but the comminution rate varied significantly with stone composition over a range of 12–630 mg/min. These data suggest that stones can be controllably fragmented by sinusoidal ultrasound bursts, which may offer an alternative treatment strategy to SWL. [Work supported by NIH 2T32DK007779-11A1, R01 EB007643, P01 DK043881, R01 DK092197, NSBRI through NASA NCC 9-58.]

5:30

4pBA18. Kidney stone fracture by surface waves generated with focused ultrasound tone bursts. Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, olegs@apl.washington.edu), Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Wayne Kreider, Bryan W. Cunitz, and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Previous studies have provided insight into the physical mechanisms of stone fracture in shock wave lithotripsy. Broadly focused shocks efficiently generate shear waves in the stone leading to internal tensile stresses, which in concert with cavitation at the stone surface, cause cracks to form and propagate. Here, we propose a separate mechanism by which stones may fragment from sinusoidal ultrasound bursts without shocks. A numerical elastic wave model was used to simulate propagation of tone bursts through a cylindrical stone at a frequency between 0.15 and 2 MHz. Results suggest that bursts undergo mode conversion into surface waves on the stone that continually create significant stresses well after the exposure is terminated. Experimental exposures of artificial cylindrical stones to focused burst waves *in vitro* produced periodic fractures along the stone surface. The fracture spacing and resulting fragment sizes corresponded well with the spacing of stresses caused by surface waves in simulation at different frequencies. These results indicate surface waves may be an important factor in fragmentation of stones by focused tone bursts and suggest that the resulting stone fragment sizes may be controlled by ultrasound frequency. [Work supported by NIH 2T32DK007779-11A1, R01 EB007643, P01 DK043881, R01 DK092197, NSBRI through NASA NCC 9-58.]

5:45

4pBA19. Histotripsy beyond the "intrinsic" cavitation threshold using very short ultrasound pulses: "Microtripsy". Kuang-Wei Lin, Yohan Kim (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., Gerstacker, Rm. 1107, Ann Arbor, MI 48109, kwlin@umich.edu), Adam D. Maxwell (Urology, Univ. of Washington, School of Medicine, Seattle, WA), Tzu-Yin Wang (Radiology, Stanford Univ., Stanford, CA), Timothy L. Hall, Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), Brian Fowlkes (Radiology, Univ. of Michigan, Ann Arbor, MI), and Charles A. Cain (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Conventional histotripsy uses pulses with $\geq$3 cycles wherein the bubble cloud formation relies on the pressure-release scattering of the positive shock fronts from sparsely distributed cavitation bubbles. In a recent work, the peak negative pressure (P(-)) threshold for the generation of dense bubble clouds directly by a negative half cycle were measured, and this threshold has been called the "intrinsic threshold." In this work, characteristics of lesions generated with this intrinsic threshold mechanism were investigated using RBC phantoms and excised canine tissues. A 32-element, PZT-8, 500 kHz therapy transducer was used to generate short (<2 cycles) histotripsy pulses at PRF = 1Hz and P(-) = 24.5–80.7 MPa. The results showed that the spatial extent of the histotripsy-induced lesions increased as the applied P(-) increased, and the lesion sizes corresponded well to the estimates of the focal regions above the intrinsic threshold. The sizes for the smallest reproducible lesions averaged 0.9×1.7mm (lateral $\times$ axial), significantly smaller than -6 dB beamwidth of the transducer (1.8×4.0 mm). These results suggest that predictable, well-confined and microscopic lesions can be precisely generated using the intrinsic threshold mechanism. Since the supra-threshold portion of the negative half cycle can be precisely controlled, lesions considerably less than a wavelength are easily produced ("microtripsy").

THURSDAY AFTERNOON, 5 DECEMBER 2013 — MASON, 1:30 P.M. TO 4:00 P.M.

Session 4pEA

Engineering Acoustics: Beam Control of Microphone and Transducer Arrays

Michael Zarnetski, Chair
NUWC, Newport, RI 02841

Contributed Papers

1:30

4pEA1. Microphone array exploratory study. Marc Messier (Univ. of Miami, 1527 Albenga Ave., University Village Bldg. 4 Rm. 401A, Coral Gables, FL 33146, m.messier@umiami.edu)

In the field of acoustic signal processing, one of the most popular areas of research is that of microphone arrays. As a small research project in this field, a microphone array system with adjustable polar response will be developed. It is worth noting that this submission is more a research proposal than an abstract. This research will take part as a means of combining coursework and research for courses in engineering acoustics and real-time digital signal processing at the University of Miami. Before any arrays are physically constructed or any code written on a DSP, simulations will be performed in MATLAB to determine optimum array configurations and to analyze various DSP algorithms for adjusting polar responses. Then, using a Texas Instruments TMS320C6713 DSP on a DSK audio target board, and writing code in C, Assembly, and MATLAB, physical tests will be conducted to analyze performance of design. A MEMS microphone model will likely be used for its small size and low cost. From there, results will be analyzed and further research in the area proposed.

1:45

4pEA2. Prediction of the spatial response of a microphone capsule using scattering and lumped-element simulations. Douglas Rollow (Technol. and Innovation, Sennheiser Electronic Corp, 550 15th St., Ste. 37, San Francisco, CA 94103, tad.rollow@sennheiser.com), Vladimir Gorelik, Meike Wulkau (Technol. and Innovation, Sennheiser Electronic GmbH & Co. KG, Wedemark, Germany), and Sebastian Chafe (Technol. and Innovation, Sennheiser Electronic Corp., San Francisco, CA)

When a microphone capsule is placed in an environment with surrounding structures, the electrical response of the capsule will be dictated by the

capsule design and by acoustic scattering of the structures surrounding it. Lumped element models of the transducer are typically used in predicting the electrical response to the field, but when the transducer is operating at a frequency where the spatial variation of the field is significant, the simplifying assumptions used in these models no longer hold. In this work, a finite element-based scattering simulation provides blocked-port field quantities to drive a lumped element circuit model, predicting the electrical output as a function of the frequency and incident angle of an incoming plane wave. The scattering simulation allows for the inclusion of mechanical supporting structures and protective screens, and their influence on the field at the rear port of a gradient transducer. Simulated results are shown for the simplified capsule and compared to measurements of a real capsule in free field conditions as well as with surrounding structures.

2:00

4pEA3. Microphone array with computer vision based directivity. Marc Messier and Jordan Reimers (Univ. of Miami, 1527 Albenga Ave., University Village Bldg. 4 Rm. 401A, Coral Gables, FL 33146, m.messier@umiami.edu)

In the field of acoustic signal processing, one of the most popular areas of research is that of microphone arrays. As a small research project in this field, a microphone array system with adjustable polar response will be developed. To make research more innovative and multi-disciplinary, the polar response of the array will be controlled by a facial tracking system implemented with computer vision techniques. This research will take part as a means of combining coursework and research for courses in engineering acoustics, computer vision, and real-time digital signal processing at the University of Miami. Before any physical testing, simulations will be performed in MATLAB to determine optimum array configurations and to analyze various facial tracking and dsp algorithms. Then, using a Texas Instruments TMS320C6713 DSP on a DSK audio target board, and writing code in C, ASSEMBLY, and MATLAB, physical tests will be conducted to analyze performance of design. A MEMS microphone model will likely be used for its small size and low cost. A Microsoft Kinect and compatible desktop computer will be used for the computer vision interface. From there, results will be analyzed and further research in the area proposed.

2:15

4pEA4. A simple adaptive cardioid direction finding algorithm. Gary W. Elko (mh Acoustics LLC, 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com) and Jens Meyer (mh Acoustics LLC, Fairfax, Vermont)

A simple direction-finding algorithm using three or four omnidirectional microphones is described. The algorithm is based on the minimization of the output power of a generally steerable cardioid microphone. It is shown that the algorithm can be reduced to running three independent single-tap LMS filters for the general 3D case and two independent single-tap LMS adaptive filters for the 2D (null constrained to lie in a plane). Results will also be shown for a 32-element spherical microphone array for sources corrupted with microphone self-noise and reverberation.

2:30

4pEA5. *In situ* evaluation of surround sound system performance. Eric M. Benjamin (Surround Res., 1229 Springwood Way, Pacifica, CA 94044, ebenj@pacbell.net), Aaron J. Heller (Artificial Intelligence Ctr., SRI Int., Menlo Park, CA), and Fernando Lopez-Lezcano (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., Palo Alto, CA)

Surround sound systems are produced with the intention of reproducing the spatial aspects of sound, such as localization and envelopment. As part of his work on Ambisonics, Gerzon developed two metrics, the velocity and energy localization vectors, which are intended to predict the localization performance of a system. These are used during the design process to optimize the decoder that supplies signals to the loudspeaker array. At best, subjective listening tests are conducted on the finished system, but no objective assessments of the spatial qualities are made to verify that the realized performance correlates the predictions. In the present work, binaural recordings were made of a 3-D 24-loudspeaker installation at Stanford's Bing Studio. Test signals were used to acquire the binaural impulse response of each loudspeaker in the array and of Ambisonic reproduction using the loudspeaker array. The measurements were repeated at several locations within the hall. Subsequent analysis calculated the ITDs and ILDs for all cases. Initial results from the analysis of the ITDs and ILDs for the center listening position show ITDs, which correspond very closely to what is expected in natural hearing, and ILDs, which are similar to natural hearing.

2:45

4pEA6. Network modeling of multiple-port transducers across multiple modes of vibration. Michael L. Kuntzman, Nishshanka N. Hewa-Kasakarage, Donghwan Kim, and Neal A. Hall (Elec. and Comput. Eng., The Univ. of Texas at Austin, 10100 Burnet Rd., Bldg. 160, MER 1.108, Austin, TX 78752, mlkuntzman@gmail.com)

A network modeling procedure is presented that is capable of modeling transducers across a broad frequency regime with multiple coupling ports. The model is based on modal superposition, and a separate network is crafted for each vibration mode of the device. Modal velocity, rather than a particular physical velocity on the vibrating transducer, is chosen as the flow variable in each network. Multiple ports are modeled with the use of multiple transformers in series. A procedure for performing system identification to complete the network parameters is also presented, which can be performed experimentally, analytically, or through use of a finite element model in the design stage. Application of the procedure to a multiple port piezoelectric microphone is presented.

3:00

4pEA7. Environment mapping and localization with an uncontrolled swarm of ultrasound sensor motes. Erik Duisterwinkel (INCAS3, Dr. Nassaulaan 9, Assen 9401 HJ, Netherlands, erikduisterwinkel@incas3.eu), Libertario Demi, Gijs Dubbelman (Dept. of Elec. Eng., Eindhoven Univ. of Technol., Eindhoven, Netherlands), Elena Talnishnikh, Heinrich J. Wörtche (INCAS3, Assen, Netherlands), and Jan W. Bergmans (Dept. of Elec. Eng., Eindhoven Univ. of Technol., Eindhoven, Netherlands)

A method is presented in which a (large) swarm of sensor motes perform simple ultrasonic ranging measurements. The method allows to localize the motes within the swarm, and at the same time, map the environment which the swarm has traversed. The motes float passively uncontrolled through the environment and do not need any other sensor information or external reference other than a start and end point. Once the motes are retrieved, the stored data can be converted into the motes relative positions and a map describing the geometry of the environment. This method provides the possibility to map inaccessible or unknown environments where electro-magnetic signals, such as GPS or radio, cannot be used and where placing beacon points is very hard. An example is underground piping systems transporting liquids. Size and energy constraints together with the occurrence of reverberations pose challenges in the way the motes perform their measurements and collect their data. A minimalistic approach in the use of ultrasound is pursued, using an orthogonal frequency division multiplexing technique for the identification of motes. Simulations and scaled air-coupled 45–65 kHz experimental measurements have been performed and show feasibility of the concept.

3:15

4pEA8. Simulations about non-Doppler continuous wave usage of ultrasonic transducers. Emre İkizler (Informatics and Information Security Res. Ctr. (BILGEM), The Sci. and Technolog. Res. Council of Turkey (TUBITAK), TUBITAK Yerleskesi BILGEM Binasi, Gebze 41400, Turkey, emre.ikizler@tubitak.gov.tr) and Hulya Sahinturk (Dept. of Mathematical Eng., Yildiz Tech. Univ., Istanbul, Turkey)

In this work, computer simulations are executed in order to detect the presence of an object which passes through the area illuminated by ultrasonic transmitter. In simulations, object which has a constant speed can be detected by only observing received signal level on ultrasonic receiver, not by using Doppler Effect or Fast Fourier Transform. Additionally, alterations on speed and distance of the object causes sensible alterations on received signal level on ultrasonic receiver. According to simulation results, a simple way for detecting speed of objects is able to be introduced by using more simple signal processing techniques compared to Doppler Effect or Fast Fourier Transform related techniques.

3:30

4pEA9. Development of a multi-resonance transducer for highly directional underwater communication. Yonghwan Hwang, Yub Je (Mech. Eng., Postech, PIRO416, Postech, Hyo-ja dong, Nam gu, Po hang KS010, South Korea, serenius@postech.ac.kr), Jaeil Lee, Jonghyeon Lee (Ocean System Eng., Jeju Univ., Jeju, South Korea), Wonho Kim, Heesun Seo (ADD, Jin hae, South Korea), and Wonkyu Moon (Mech. Eng., Postech, Po hang, South Korea)

The parametric array is a nonlinear conversion process that generates a narrow beam of low-frequency sound using small aperture. It can be applied to underwater communication between two nodes with known locations, since the highly directional sound beam may provide such benefits as privacy, no interference due to the multi-path. The difference frequency wave (DFW) from the parametric array shows small side lobes and extraordinary directivity. The shortcoming of the DFW generated by the parametric array may be its low sound pressure level relative to that of the directly generated sound beams. In this study, we designed and fabricated a multi-resonance transducer as a parametric array source and evaluated its feasibility as a transmitter. For that purpose, we determined the proper design parameters for midrange communication. We selected 10 kHz as the communication frequency and then determined the primary frequencies as 100 and 110 kHz. We composed the source transducer using the two kinds of unit transducers. The fabricated transducer array and the developed operating techniques enabled us to successfully transmit letters, words, and drawings inside the water tank. By testing the characteristics, we confirmed that the developed operating scheme and transducer can be used for underwater communication. [Work supported by ADD (UD130007DD).]

3:45

4pEA10. A basic study on frequency characteristics compensation of sound at ear drum when using hearing aids. Hitoshi Iseda (Dept. of Mech. and Aerosp. Eng., Tokyo Inst. of Technol., Chofugaoka 3-66-1,801, Chofu, Tokyo 182-0021, Japan, iseda.h.aa@m.titech.ac.jp) and Masaaki Okuma (Dept. of Mech. and Aerosp. Eng., Tokyo Inst. of Technol., Kawagoe, Japan)

Wearing the acoustic equipment such as hearing aids (HAs) and earphones changes the frequency characteristics of sound at the ear drum because the equipment influences acoustic condition of outer ear canal as a partition wall in the canal. This influence should be compensated in order not to degrade the quality of auditory perception. To achieve this, the identification of acoustic property in the canal is required for both cases of with and without HAs. The final goal of this research is to develop a universal method for the frequency characteristics compensation of sound at the ear drum when wearing HAs and incorporating the algorithm of the method into such equipment. In this paper, as the first phase of the research, the authors present the results of an experimental identification technique using simple and large scale models of canal and HAs based on the theory of the transfer matrix method. The result of a compensation experiment using digital filter processing is also presented.

THURSDAY AFTERNOON, 5 DECEMBER 2013 — CONTINENTAL 1, 2:00 P.M. TO 4:30 P.M.

Session 4pMUa

Musical Acoustics and Structural Acoustics and Vibration: Acoustics of Percussion Instruments II

Thomas D. Rossing, Chair
Stanford Univ., Stanford, CA 94022

Invited Papers

2:00

4pMUa1. Modeling orchestral crotales as thin plates. Thomas R. Moore, Daniel W. Zietlow, and Donald C. Griffin (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu)

Orchestral crotales are designed in such a way that the overtones become less harmonic as the fundamental pitch increases. Deutsch, *et al.* used Kirchhoff-Love theory to show that by eliminating the central mass and choosing the correct ratio of inner to outer radius the overtones can be harmonically related (J. Acoust. Soc. Am. **116**, 2427 (2004)). However, when a crotale was constructed using this design, the overtones were not harmonically related. We show that this lack of agreement between theory and experiment is due to the fact that shear motion of the inner boundary, which is neglected in classical thin-plate theory and thought to be unimportant, can significantly affect the resonance frequencies of plates even when they are extremely thin.

2:20

4pMUa2. Acoustics of Western and Eastern bells, old and new. Thomas D. Rossing (CCRMA, Stanford Univ., 26464 Taaffe Rd., Los Altos Hills, CA 94022, rossing@ccrma.stanford.edu)

The modes of vibration and sound radiation from tuned church bells, carillon bells, handbells, ancient Chinese two-tone bells, and temple bells will be compared. Most bells have a circular cross section, but many ancient Chinese bells do not, and thus they have two different strike notes, depending upon where they are struck. The musical interval between these two strike notes is often near a minor third or a major third.

Contributed Papers

2:40

4pMUa3. Evolution of the Hang percussion instrument and associated performance practices. David Wessel (Music CNMAT, Univ. of California Berkeley, 1750 Arch St., Berkeley, CA 94709, davidwessel@me.com) and Thomas Rossing (Thomas Rossing, CCRMA, Stanford, CA)

It is now over 10 years since the widespread adoption of the Hang percussion instrument. It has evolved considerably since its invention in 2000 by Felix Rohner and Sabina Scharer in Bern, Switzerland. We present acoustical analyses of variations of the instrument in conjunction with an evolving performance technique. The talk will include demonstrations of performance practices and their acoustical consequences.

3:00

4pMUa4. Characterization of the mechanical properties of the steelpan. April Bryan, Marc Gobin, Akill Griffith, Dillon Frederick (Dept. of Mech. and Manufacturing Eng., The Univ. of the West Indies, St. Augustine, Trinidad and Tobago, aprilbr@gmail.com), Brian Copeland (Elec. and Comput. Eng., The Univ. of the West Indies, St Augustine, Trinidad and Tobago), and Clement Imbert (Mech. and Manufacturing Eng., The Univ. of the West Indies, St Augustine, Trinidad and Tobago)

The steelpan is a struck idiophone whose playing surface is constructed by forming the top of a fifty-five gallon steel oil drum into a sunken, nearly hemispherical surface and then raising smaller shells on the hemisphere to form notes. The completed instrument resembles an inverted turtle shell and is played by striking the notes with sticks. Although it is understood that variations in note geometry and material properties are mainly responsible for the characteristic sounds generated when the notes are struck, few studies have investigated these relationships. Previous research efforts have explored the metallurgical properties and the characteristic vibrations of the notes. Less emphasis has been placed on the relationship between the mechanical properties of the steelpan and its acoustic behavior. In this research, the variation in the mechanical properties across the playing surface of a tenor steelpan is characterized. Of the instruments in the steelpan family, this instrument has the greatest deformation and the most notes. More specifically, the variation in the residual stress, strength, Young's Modulus, and Poisson's ratio are determined and compared among octave sets. This characterization is important for the development of models that relate the mechanical properties to the acoustical behavior of the steelpan.

Invited Papers

3:20

4pMUa5. Rhythmic techniques and psychoacoustic effects of the percussion music of Steve Reich. Garry M. Kvistad (Woodstock Percussion, Inc., 167 DuBois Rd., Shokan, NY 12481, garry@chimes.com)

Beginning in the 1960s, composer Steve Reich began to experiment with rhythmic devises. "Phase shifting" is where unison melodies become canons as one voice speeds up slightly. This was done first with tape loops and later with live performers. In 1967, he applied this technique to "Piano Phase" in which two pianists on two pianos start a pattern in unison. One player slowly speeds up, creating canons which lock in and out of rhythmic "unisons" eventually ending back in melodic unison. This process continues two more times with different patterns. This rhythmic technique was used in several other early works of Reich such as his 1970 composition, "Drumming." NEXUS member Garry Kvistad has built a multi-element set of percussion instruments tuned in just intonation to perform Piano Phase. The first section is played on a large wooden bar instrument similar to an African Amadinda. The second section is played on a set of thick aluminum tubes. The last section is played on an aerophone activated by slapping the ends of closed tubes. In this presentation, Mr. Kvistad will demonstrate this difficult performance technique using the newly built instruments.

3:40

4pMUa6. A first look into the caxirola—Official music instrument of the Soccer World Cup 2014. Talita Pozzer and Stephan Paul (UFSM, Undergraduate Program in Acoust. Eng., DECC-CT-UFSM, Av. Roraima 1000, Camobi, Santa Maria, RS 97105-900, Brazil, talita.pozzer@eac.ufsm.br)

For the 2014 Soccer World Cup Brazilian Musician Carlinhos Brown created the caxirola as the official music instrument, adapting an old African instrument—the caxixi. Both caxixi and caxirola generate sound by hard particles impacting on the walls of a closed basket. While the basket of the caxixi is made of natural materials the basket of the caxirola is made of environmentally friendly polymer. Remembering the acoustical impact of the vuvuzela in 2010 as quite negative it was decided to study the acoustics of the caxirola. First, the way people will use the caxirola in terms of arm's excursion and velocity of shaking was studied. It was found that the caxirola was used moving it longitudinally and perpendicularly to its main axis, both in vertical direction. Spectra of the sound emitted by the caxirola and caxixi moving perpendicularly are quite similar with slightly more low frequency energy. However, when subjected to longitudinal movements, spectra are different. For the caxirola, the particle impact sound on the walls is the same for all impacted walls but for the caxixi, where lateral walls are of different material than the bottom, the spectra shows more energy for impacts at bottom.

Contributed Papers

4:00

4pMUa7. Javanese gong wave signals. Matias H. Budhiantho and Gunawan Dewantoro (Electronics and Comput. Eng., Satya Wacana Christian Univ., Jln. Diponegoro 52-60, Salatiga, Jawa Tengah 50711, Indonesia, mhwb@Gmail.com)

In Central Java, the Gong is one of eminent gamelan instrument, an ensemble of predominantly struck instruments that has deep philosophical meaning for Javanese. However, there lack of studies concerning on this particular instrument as a bridging means between scientific description and human artistic perception. This study aims to investigate the spectral and temporal properties as well as particularly look into the typical wave-like sound of the Gong. Acoustic measurements were conducted and analyzed using ARTA. Both frequency and time domain analyses were explored to better understand the nature of the Gong wave signals. The fundamental frequency which decays much more slowly than the other harmonic started with lower increasing frequency. The wave-like sound of the Gong maybe due to signal behavior the resemblance beat phenomenon between early and later development of the fundamental sustaining frequency.

4:15

4pMUa8. Percussion, via transducers. Alex Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

The sounds within a music recording are necessarily mediated by the microphones and loudspeakers associated with their recording and playback. The effect of microphones—type and placement—invites a unique view of musical acoustics. The necessity of loudspeaker playback motivates a range of signal processing strategies, using equalization, compression, reverberation, distortion, and more to further reshape the sound. This paper reviews the sonic influence of contemporary audio engineering craft on the sound of percussion instruments as realized in music recordings.

THURSDAY AFTERNOON, 5 DECEMBER 2013 CONTINENTAL 1, 5:00 P.M. TO 6:00 P.M.

Session 4pMUb

Musical Acoustics: Percussion Concert

Thomas D. Rossing, Chair
Stanford Univ., Stanford, CA 94022

A mini-concert will be held following session 4pMUa featuring Gary Kvistad, David Wessel, Punita Singh, Rohan Krishna Murthy and others.

THURSDAY AFTERNOON, 5 DECEMBER 2013 CONTINENTAL 9, 1:00 P.M. TO 5:25 P.M.

Session 4pNS

Noise and Structural Acoustics and Vibration: Active Control of Sound and Vibration

Scott D. Sommerfeldt, Cochair
Dept. of Physics, Brigham Young Univ., N181 ESC, Provo, UT 84602

Kenneth Cunefare, Cochair
Georgia Technol., Mech. Eng., Atlanta, GA 30332-0405

Invited Papers

1:00

4pNS1. Active noise control: Eight decades of research and applications. Kenneth Cunefare (School of Mech. Eng., The Georgia Inst. of Technol., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

On January 27, 1933, Paul Lueg submitted his patent application for an active noise control system. In the intervening 80 years, the concept has progressed from simple single-input single-output feedback control systems through to complex MIMO approaches implementing a seemingly endless variety of control algorithms. Academic publications appear to continue at approximately a constant pace over the past decade, with much attention paid to algorithm refinement and development of applications. Commercial successes of active noise control include ANC headset in the aviation and consumer markets, as well as systems installed in passenger vehicle (some of which also incorporating active vibration control systems). This paper will provide a brief historical retrospective on the development of active noise control, and survey the current state of academic research as well as existing and proposed production applications (consumer, aviation, defense, etc.).

1:20

4pNS2. Strategies for improving speech intelligibility and warning signal detection in communication headsets/hearing protectors. Anthony J. Brammer, Eric R. Bernstein, and Gongqiang Yu (Ergonomics Technol. Ctr., UConn Health Ctr., 263 Farmington Ave., Farmington, CT 06030-2017, brammer@uchc.edu)

Strategies for improving speech understanding and warning signal detection when wearing communication headsets/hearing protectors (HPDs) in environmental noise must accommodate sounds from different sources at different times. A subband signal processing approach would appear desirable, with a delayless structure essential for active noise reduction (ANR). The requirements for communication channel and ANR controllers differ, owing to the different bandwidths required for speech and warning signals, and for ANR. Subbands for optimizing speech signal-to-noise ratios are commonly fractional-octave bandwidth, while computational efficiency favors linear subbands for ANR. Increasing the number of subbands reduces computational cost for the latter but the advantage is less apparent for communication signal control. Possibilities exist for harmonizing subband filter structures by constructing models of speech intelligibility using computationally efficient bandwidths. In contrast, algorithms for detecting warning sounds tend to be governed more by audibility than bandwidth considerations. The issues will be discussed using a simulation of a circumaural HPD that can replicate word scores obtained by a subject when subject-specific transfer functions are employed. Where available, results from physical devices and subjects will be included, as well as the consequences of differences in individual auditory abilities. [Work supported by NIOSH grant R01 OH008669.]

1:40

4pNS3. A perceptually motivated active noise cancellation system for hearing impaired listeners: An overview. Buye Xu, Jinjun Xiao, and Tao Zhang (Signal Processing Res., Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344, buye_xu@starkey.com)

Nowadays most hearing impaired (HI) patients are fitted with either open-fitting or vented-fitting hearing aids (HAs) to reduce the occlusion effect. In an environment where strong low-frequency ambient noise is present, significant noise energy can directly leak into the ear canal bypassing noise reduction algorithms in the HAs and may reduce speech intelligibility and listening comfort for HA users. One way to mitigate such an issue without occluding the ear canal is to implement an active noise cancellation (ANC) system inside the ear canal. Traditional ANC systems are designed to minimize the total sound pressure level in the ear canal. However, this may not necessarily lead to an optimal solution from the perceptual perspective (e.g., loudness may be reduced instead of being minimized as a result). In this paper, a perceptually motivated feedback ANC system is presented: a spectral shaping filter is applied to the residual error signal to minimize the loudness for HI listeners. In addition, implications of the practical constraints introduced by the HAs are discussed based on acoustic simulations and experimental results.

2:00

4pNS4. A better frequency domain adaptive algorithm for active noise control in a short duct. Jing Lu and Xiaojun Qiu (Inst. of Acoust., Nanjing Univ., Hankou Rd. 22th, Nanjing 210093, China, lujing@nju.edu.cn)

For the feedforward active noise control in a short duct, the noncausality of the whole system is often inevitable, since the reference sensor needs to be placed very close to the control source, and the acoustic transmission delay between them is often surpassed by the inherent AD/DA and anti-aliasing filters latency in the controller. The commonly used normalized frequency domain LMS algorithm possesses the benefit of fast convergence speed, but suffers from deteriorated steady-state performance when used in non-causal circumstances. In this paper, an efficient modification of the normalized frequency domain LMS algorithm is proposed, which can significantly improve wide-band noise reduction level in non-causal circumstances. Both simulations and experiments demonstrate the superiority of the proposed algorithm.

2:20

4pNS5. Local and global active noise control using a parametric array loudspeaker. Nobuo Tanaka (Tokyo Metropolitan Univ., 6-6 Asahigaoka, Hino-shi, Tokyo 191-0065, Japan, ntanaka@sd.tmu.ac.jp)

This paper deals with local as well as global active noise control (ANC) using a parametric array loudspeaker (PAL) possessing intriguing properties: sharp directivity, low sound pressure decay by distance, capability of steering directivity, etc. After briefing some properties of a PAL necessary for ANC, this paper presents pinpoint control using a PAL for suppressing the sound pressure at a designated location, hence local control. It is shown that unlike conventional ANC in which a voice coil loudspeaker is used, the pinpoint control may achieve the sound pressure suppression without causing spillover. Using the same sound control source, PAL, this paper then refers to global control, termed trivial control in the art, enabling one to generate a global zone of quiet. The trivial control strategy falls into a category of acoustic power control based on a trivial condition requiring the collocation of a primary source and a control source. The trivial condition formerly avoided because of literally trivial may be implemented due to the property of a PAL, thereby enhancing the control effect, theoretically infinity. Two kinds of control strategy are then demonstrated with a view to fulfilling the trivial condition for global control.

2:40

4pNS6. Virtual mechanical impedance approach for the active structural acoustic control of panels. Alain Berry, Marc Michau, Philippe Micheau (Mech. Eng., Université de Sherbrooke, 5907 Laurent, Sherbrooke, QC J1N 3Z2, Canada, alain.berry@usherbrooke.ca), and Philippe Herzog (Laboratoire de Mécanique et d'Acoustique, Marseille, France)

This work investigates harmonic Active Structural Acoustic Control of flexural panels using structural, collocated and dual actuator-sensor pairs. Two types of transducer technologies are envisioned: (1) thin piezoelectric actuators and sensors; (2) electrodynamic inertial actuators and transverse velocity sensors. The control strategy is to locally impose a complex, virtual mechanical impedance to the structure via a linear relation between the actuator input and sensor output of each pair, at each frequency of interest. This virtual

impedance is optimized to minimize the sound radiation of the structure at the corresponding frequency. The approach is implemented as a two-step process: (1) the optimal virtual impedance matrix is derived from identification of the primary sound and transfer functions between the control actuators, structural sensors and far-field acoustic sensors; (2) the optimal virtual impedance matrix is imposed using a real-time, iterative controller. Numerical and experimental results are discussed to highlight the underlying physical interpretation of the virtual impedances for sound radiation or sound transmission control. The implication of the different actuator and sensor technologies in terms of sensing the global vibration and acoustic response is also discussed.

3:00–3:15 Break

3:15

4pNS7. Active structural acoustic control using a sum of weighted spatial gradients control metric. William R. Johnson, Daniel R. Hendricks, Monty J. Anderson, Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT 84602, will.johnson@byu.edu), and Scott D. Sommerfeldt (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Active structural acoustic control (ASAC) is an active noise control technique, which provides global control by targeting and minimizing the structural vibrations which contribute to radiated sound power. The majority of research in ASAC has focused on validating various proposed concepts on flat rectangular plates, an important but not comprehensive class of structures. To extend the body of knowledge, ASAC has been investigated on finite ribbed plates under a variety of boundary conditions. Simulated results have shown that two different approaches, minimizing the volume velocity and minimizing the weighted sum of spatial gradients (WSSG) provide comparable average attenuation of radiated sound power on ribbed plates. With regards to sensing, minimizing WSSG has several advantages over minimizing volume velocity. In particular, WSSG has been formulated to be easier to measure than volume velocity, without requiring a priori information about the structure or its modes. WSSG has also been shown to be relatively uniform spatially and relatively insensitive to boundary conditions, while also providing improved control over volume velocity at structural modes higher than the first mode. These results suggest that more practical, complex vibrating structures can be effectively controlled for the reduction of radiated sound power using the WSSG approach.

3:35

4pNS8. Convergence analysis of filtered-x least mean squares algorithm for active control of repetitive impact noise. Guohua Sun, Tao Feng, Mingfeng Li, and Teik C. Lim (College of Eng. and Appl. Sci., Univ. of Cincinnati, 801 ERC, P.O. Box 210018, 2901 Woodside Dr., Cincinnati, OH 45221, teik.lim@uc.edu)

The prevalent adaptive active noise control (ANC) algorithm, namely, the filtered-x least mean squares (FXLMS), exhibits a critical challenge for treating transient impact noise. This is because the FXLMS algorithm requires certain adaptation time to converge satisfactorily. However, the FXLMS algorithm may have its learning capacity when the transient noise shows certain repeatability. In this paper, a distinctive theoretical analysis of the convergence behavior of ANC system with the standard FXLMS algorithm is conducted for repetitive impulse-induced transient noise control. To simplify the derivation, the secondary path is assumed to be a pure delay model. Through this analysis, a step size bound condition is derived, and an optimal step size that leads to the fastest convergence is determined. To validate the analysis, extensive numerical simulations are performed considering various pure delay secondary path models. Calculations are in very good agreement with the theoretical analysis. Finally, a more general secondary path is considered to further demonstrate the effectiveness of the FXLMS algorithm for repetitive impulse noise control. The results indicate that ANC system with the FXLMS algorithm can be a very promising technique for repetitive transient noise typically seen in industrial facilities such as punching machines.

Contributed Papers

3:55

4pNS9. Active control of sound transmission through soft-cored sandwich panels using volume velocity cancellation. Kiran C. Sahu and Prof. Jukka Tuhkuri (Dept. of Appl. Mech., Aalto Univ., Puumiehenkuja 5 A, Espoo 02150, Finland, kiran.sahu@aalto.fi)

In this paper, the active control of harmonic sound transmitted through soft-cored sandwich panels into a rectangular enclosure is studied. As it has already been shown that in the low frequency region, the noise transmission through soft-cored sandwich panels mainly occurs due to flexural and dilatational modes [Rimas Vaicaitis, NASA Technical Note, NASA TN D-8516, 1977]; therefore, in this study, volume velocity cancellation control strategy is used to control these modes, and achieve sound attenuation in a broad frequency range. Point force and uniformly distributed force actuators are used as the secondary source to cancel the volume velocity of the inner surface of the sandwich panel which is open to cavity. Cancelling the net volume velocity of this is compared not only in terms of the reduction in sound power in the enclosure but also in terms of the plate velocities. Numerical studies indicate that the active control method controls both the flexural and dilatational modes and therefore, attenuates significant amount of sound power inside the cavity irrespective of the isotropic loss factors of the viscoelastic core. Also a finite element study has been done in the commercially available COMSOLMultiphysics software to compare with the analytical result.

4:10

4pNS10. Effect of modeling errors on virtual sensing systems for active noise control. Luis Vicente (Aragon Inst. of Eng. Res., Univ. of Zaragoza, Maria de Luna, 1, Zaragoza E50018, Spain, lvicente@unizar.es)

In the active noise control literature, there are a number of algorithms based on the virtual sensing approach. In all of them, the aim is to control the noise at a position apart from the physical sensors, by somehow estimating the actual signal at that point. The benefit of such an arrangement is evident. However, the practical difficulties found when trying to achieve a properly working virtual sensing system surely are the main reason for a reduced number of successful applications reported. In this paper, we analyze and quantify those difficulties for several virtual sensing algorithms. All of them are based on some modeling that needs to be made previous to the actual control. We focus on the effect that errors on these models have on the stability of the whole system, the cancellation capability and the convergence rate. We check that the sensitivity of the cancellation capability to modeling errors is much higher in the virtual sensing case, when compared to that of systems with physical sensors on the desired points of cancellation.

4:25

4pNS11. Current developments in practical active damping systems for vibration-isolated platforms. Vyacheslav Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA 92606, vyacheslav.ryaboy@newport.com)

The paper discusses recent developments in practical implementation of the active damping solution for vibration isolated platforms (optical tables) known as Smart Table. The system implements a decentralized velocity feedback with sensors and actuators integrated into the platform. It proved effective in creating vibration-free environments for sensitive experiments and precision manufacturing processes in life sciences, nanotechnologies and other areas. The paper describes expansion of the technology to larger platforms characterized by lower resonance frequencies. The technical difficulties related to interference between the resonance properties of the structure and the resonance of the electromagnetic actuator (see Elliott and Baumann, J. Acoust. Soc. Am. **121**(5), 2007) were addressed successfully. Other developments required by expansion of the area of applications included introduction of dampers acting in all directions, as well as a portable modular version of the active dampers.

4:40

4pNS12. Development of radiation mode shapes for cylindrical shells. Pegah Aslani, Scott Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N203 ESC, Provo, UT 84602-4673, pegah.aslani@gmail.com), and Jonathan Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

For many acoustical applications, it is desirable to evaluate the radiated power. About two decades ago, a set of formulations were developed to represent the acoustic radiation in terms of radiation mode shapes. A convenient method for determining these radiation modes involves representing the radiating structure as a set of elementary simple radiators, from which the radiation can be decomposed into the set of orthogonal radiation modes. Radiation mode shapes are very useful not only for calculating the power, but also to determine which modes are the most efficient radiators. This generally allows one to achieve a rather accurate estimate of the radiated power by including only a relatively small number of the most efficient radiating modes. This concept has significant implications for an efficient strategy for implementing an active noise control system. Previous work reported in the literature has primarily focused on evaluating the radiation mode shapes of flat structures, such as beams and plates. There has not been as much reported on the radiation mode shapes for cylindrical shells. This paper focuses on implementing these concepts to determine the radiation mode shapes of cylindrical shells and using them to determine the radiated acoustic power.

4:55

4pNS13. Echo removal in tubular acoustic systems: Passive and active techniques. Keir H. Groves and Barry Lennox (Elec. and Electron. Eng., The Univ. of Manchester, Sackville St. Bldg., Granby row, Manchester M17AY, United Kingdom, keir.groves@manchester.ac.uk)

Acoustic pulse reflectometry (APR) has been shown to be a very capable means of identifying features in tubular objects. APR systems excite a test object with a sound wave and listen for reflections, indicating the presence of features in the test object. An undesirable effect of this process is that the returning sound wave is re-reflected by the loudspeaker and re-enters the system. This paper presents two complimentary techniques that may be used to remove unwanted echoes in APR systems. The first approach uses two axially separated microphones to separate forward and backward propagating waves. This passive technique is shown to be highly capable of cancelling undesired echoes in the system. The second approach actively cancels unwanted echoes by introducing a phase inverted version of the wave that is incident on the loudspeaker. The active cancellation operates in real-time using the measured backwards propagating wave. As a consequence of the proposed techniques, the effectiveness of APR when applied to detecting features within tubular systems is improved considerably. The empirical results presented at the conference will demonstrate that corrosion effects, such as holes and pits, located in short lengths of pipes, can be detected clearly within seconds.

5:10

4pNS14. The effects of the orifice plate structure on the aerodynamic noise in the high parameter pressure reducing valve. Lin Wei and Zhijiang Jin (Inst. of Chemical Machinery and Process Equipment, No. 38, Zheda Rd., Hangzhou, Zhejiang 310007, China, linweily@163.com)

The high velocity steam flow in the high pressure reducing valve can cause loud noise, which is harmful to the operators and the relevant devices. The orifice plate is used to reduce the noise in the pressure reducing valve. The main objective is to study the relationship between the orifice plate structure and the aerodynamic noise in the high pressure reducing valve. Based on computational fluid dynamics hybrid approach was used to simulate the flow and the acoustic field in the valve. The thickness of the plate, the length between the plate and the plug, the bore diameter and its distribution were changed to analyze their effects on aerodynamic noise.

Session 4pPA

Physical Acoustics: Advances in Infrasound Research II

Roger M. Waxler, Cochair
NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677

John Heffington, Cochair
NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677

Invited Paper

1:30

4pPA1. Partitioning of seismo-acoustic motions for near-surface explosions and yield estimation. Arthur Rodgers (LLNL, 7000 East Ave., L-046, Livermore, CA 94551, rodgers7@llnl.gov)

Explosions near the Earth's surface excite both atmospheric overpressure and seismic ground motions. The amplitudes of air-blast (and hence acoustic/infrasound) overpressures and seismic motions depend on the explosive yield as well as the height-of-burst (HOB, for above ground emplacement) or depth-of-burial (DOB, for buried emplacement). We present analysis of air-blast overpressures and seismic motions with the goal of developing methods for robust yield estimation for near-surface blasts. Our investigations are based on the HUMBLE REDWOOD set of chemical high-explosive tests at Kirkland Air Force Base in Albuquerque, NM. We find that the air-blast positive phase impulse and seismic P-wave zero-to-peak displacement amplitude are robust estimators of yield. An empirical model for the amplitudes as a function of yield, range and HOB/DOB is presented and allows estimation of yield and HOB/DOB given a set of air-blast and seismic measurements. We find that yield and HOB/DOB can be estimated simultaneously by combining air-blast and seismic measurements. Strong trade-offs between the amplitudes and the yield and HOB/DOB for a single measurement type inhibit accurate estimates. However, simultaneous inversion of both overpressure and seismic measurements improve estimates, justifying combined seismo-acoustic analysis.

Contributed Papers

1:50

4pPA2. Observations on geomagnetic auroral infrasound waves 2003—2013. Justin J. Oldham, Charles R. Wilson, John V. Olson, Curt Szuberla, and Hans Nielsen (Phys., Univ. of Alaska Fairbanks Geophysical Inst., PO Box 750972, Fairbanks, AK 99775, joldham6@alaska.edu)

Persistent, high trace velocity infrasound activity, associated with auroral events, has been routinely observed from the CTBT/IMS I53US infrasound station in Fairbanks. Comparisons of the infrasound data with data from the Geophysical Institute Magnetometer Array, the Poker Digital All-Sky Camera, and historic data from the Poker Flat Imaging Riometer show that the observed infrasound is correlated with periods of heightened geomagnetic activity and is produced in the lower Ionosphere. With the infrasound array operating near-continuously now for roughly one full period of the solar cycle, we have systematically isolated all such geomagnetic auroral infrasound events to form a data set suitable for statistical analysis. We note a relationship between the occurrence of geomagnetic infrasound waves (GAIW) and the recovery phase of geomagnetic storms when the geomagnetic H component has a peak-to-peak amplitude of ~1500 gamma during the local time period from 5 to 10 h at the CIGO Magnetic Observatory in Fairbanks. During this time interval I53US is under the auroral oval when there are large pulsating aurora events that produce infrasonic waves. These observations restrict the apparent source geometry and generating phenomena of the infrasound, as well as providing a basis for comparison with idealized models of GAIW generation.

2:05

4pPA3. Pneumatic infrasound source: Model experiment and theory. Justin D. Gorhum, Thomas G. Muir, Charles M. Slack III, Timothy W. Hawkins, Yurii A. Ilinskii, and Mark F. Hamilton (Appl. Res. Lab., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713, jgorhum@arlut.utexas.edu)

In a previous presentation [J. Acoust. Soc. Am. **133**, 3327 (2013)], an experimental model study of a pneumatic infrasound source that utilizes the pulsation of compressed air was discussed. The present paper discusses new measurements and theoretical modeling efforts that are currently underway. Measurements of the source level, directivity patterns, propagation loss, and frequency response are presented and analyzed. Acoustic and aerodynamic models are presented and discussed with a focus on modeling and predicting nearfield system performance using multipole (monopole, dipole, and quadrupole) representations of the sound source. Measurement techniques and engineering considerations are addressed, as are physical interpretations of the process. [Work supported by ARL:UT Austin.]

2:20

4pPA4. Pneumatic infrasound source: Expanded model development and tests. Thomas Muir, Charles M. Slack, Justin D. Gorhum, and Timothy M. Hawkins (ARL UT Austin, P.O. Box 8029, Austin, TX 78713, tgmuir@earthlink.net)

A model experiment discussed in a companion paper is expanded through the engineering development of a larger scale system to provide concept evaluation of portable infrasound generation, for calibration and

tests of receiver array stations. This system utilizes an industrial compressor producing 350 cubic feet per minute of air at pressures up to 150 pounds per square inch, which is stored in two 500 gallon tanks. Air streams from each tank are released through two synchronized, rotating, 2 in. diameter ball valves, producing modulated pulse jets into the atmosphere, which then create infrasonic tone bursts. Measurements on the propagation and frequency response of infrasound so produced are compared and modeled with a view toward assessment of practical utility. [Work supported by Applied Research Laboratories, University of Texas at Austin.]

2:35

4pPA5. Acoustic signals and directivity for explosive sources in complex environments. Roger M. Waxler (NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu), Doru Velea (SAIC, Reston, VA), Jessie Bonner (Weston Geophysical Corp., Boston, MA), and Carrick Talmadge (NCPA, Univ. of Mississippi, University, MS)

Much work has gone into characterizing the blast wave, and ultimate acoustic pulse, produced by an explosion in flat, open land. Recently, an experiment was performed to study signals produced by explosions in more complex environments, both above and below ground and in the vicinity of mountainous terrain. Explosive charges, ranging in weight from 200 to 2000 lbs, were detonated in a variety of configurations in and around tubes and culverts as well as buried in alluvium and limestone. A large number of acoustic sensors were deployed to capture the directivity of the signals in the near-field and to characterize the propagation of the signal to the far field. Significant directivity was observed in the near field signals from many of the shots. The influences of both meteorology and topography were evident.

2:50

4pPA6. Infrasound from buried seismic sources in the presence of surface topography. Arthur Rodgers (LLNL, 7000 East Ave., L-046, Livermore, CA 94551, rodgers7@llnl.gov)

Buried seismic sources (such as explosions and earthquakes) can generate acoustic motions in the atmosphere through coupling along the solid-fluid boundary. Infrasound overpressures from such sources have been computed using the Rayleigh Integral where acceleration time-histories along the boundary are inversely weighted by distance, delayed by travel time and summed at an observation point in the far-field. Typically, these calculations assume the Earth's surface is flat; however, topography can result in variations of the overpressure signals due to amplitude differences at the surface and phase differences along the direction of propagation. This study considers the Rayleigh Integral to compute far-field overpressure using seismic ground motion simulations that include accurate representation of surface topography. Through a series of numerical experiments we attempt to quantify the effect of surface topography on overpressure signals.

3:05

4pPA7. An empirical study of acoustic/infrasonic source and propagation effects using a large dataset of explosions. Emily A. Morton (Geophys., Los Alamos National Lab., 4129 S Meadows Rd., Apt. 2121, Santa Fe, NM 87507, emorton@lanl.gov) and Stephen J. Arrowsmith (Geophys., Los Alamos National Lab., Los Alamos, NM)

In May 2013 we performed a series of seventy explosion tests, varying the mass, shape, and height of the explosives. Shots were comprised of 11.6 kg, 4.9 kg, and 1.7 kg cylinders and 14.9 kg spheres, all of Comp-B. Explosive heights varied between 4, 2, 1, and 1/2 m above the surface, at the surface, and buried 1 m below the surface. Explosives above the surface were suspended by rope between two concrete pillars. In addition, ground surfaces were altered between dry sand, chicken wire, and concrete blocks. We monitored the explosions on 13 acoustic stations. Four temporary stations were deployed surrounding the shot site at less than 1 km distance. Eight additional stations were at distances of 1 to less than 9 km, and one at ~23 km from the shot site, 4 of which were temporary stations, and 5 are part of the Los Alamos Seismo-acoustic Network. We report on a detailed analysis of signal differences related to explosive and meteorological variations. The large quantity of data from repeating shots enables us to formally characterize the relative importance of source and path variations.

3:20

4pPA8. Comparison of primary and secondary calibrations of infrasound microphones from 0.01 to 20 Hz. Thomas B. Gabrielson (Appl. Res. Lab., Penn State Univ., PO Box 30, State College, PA 16804, tbg3@psu.edu)

Secondary calibration of microphones at infrasonic frequencies by comparison to a reference pressure transducer in a piston-driven chamber is straightforward as long as the two transducers can be located much closer than a wavelength or a correction for their separation can be determined accurately. If the response of the reference transducer is flat to zero frequency, the reference can be calibrated statically. For comparison calibration, the uncertainty is dominated by the uncertainty in the reference. In this investigation, a calibration chamber that is normally used for comparison calibration has been analyzed for primary calibration. In the primary mode, the calibration depends on chamber dimensions, piston displacement, temperature, barometric pressure, leak rate, and a thermo-viscous acoustic model. The primary and secondary calibrations are performed simultaneously; however, the two calibration modes produce almost entirely independent response estimates of both magnitude and phase. The calibrations extend well below the nominal low-frequency roll-off of the microphone and allow identification of the characteristics of the pressure-equalization leak. In addition to the linear analysis, the effects of nonlinearity and convection are explored.

3:35

4pPA9. Lightning characterization through acoustic measurements. Louis-Jonardan Gallin (DAM, DIF, CEA, CEA/DAM Ile de France, Arpajon 91297, France, gallin@dalembert.upmc.fr), Mathieu Rénier, Éric Gaudard (Institut Jean le Rond d'Alembert, UPMC, Paris, France), Thomas Farges (Institut Jean le Rond d'Alembert, UPMC, Arpajon, France), Régis Marchiano (Institut Jean le Rond d'Alembert, UPMC, Paris, France), and François Coulouvrat (Institut Jean le Rond d'Alembert, CNRS, Paris, France)

Lightning generated acoustic shock waves are the most frequent natural explosions: they are good candidates to probe meteorological local properties of the acoustic propagation medium over distances of less than 100 km. The goal of the Ph.D. is to study the transformation the thunder undergoes (amplitude, spectrum) during its travel from the lightning channel towards a detector (microphone, microbarograph), the work is based on two complementary approaches: first the Flhoward software (UPMC) designed to simulate the propagation of acoustic shock waves through a realistic atmosphere model (including temperature gradients, rigid ground, and winds) will help us studying the traveling waveforms. And in second the analysis of the acoustic records (audible and infrasounds) obtained during the PEACH campaign (Autumn 2012) will provide data to which simulations will be confronted.

Session 4pPP

Psychological and Physiological Acoustics: The Ear Club: Honoring Ervin R. Hafter and His Contributions to the Study of Binaural Processing and Auditory Cognition II

Brent Edwards, Chair
Starkey Hearing Res. Ctr., 2150 Shattuck Ave., Berkeley, CA 94704

Invited Papers

2:00

4pPP1. Lateralization of simulated sources and echoes on the basis of interaural differences of level. Raymond H. Dye, Jacquelyn P. Hill, Leslie M. Ryan, Alexander E. Cupler, and Kevin M. Bannon (Psych., Loyola Univ. Chicago, 1032 W. Sheridan Rd., Chicago, IL 60201, rdye@luc.edu)

This experiment assessed the relative weights given to source and echo pulses lateralized on the basis of interaural differences of level (IDLs). Separate conditions were run in which the to-be-judged target was the first (source) or second (echo) pulse. Each trial consisted of two intervals; the first presented a 3000-Hz diotic pulse that marked the intracranial midline and the pitch of the target frequency. The second presented the sequence of a source followed by an echo. Target frequency was always 3000 Hz, while the non-target pulse was presented at 1500, 3000, or 5000 Hz. Delays between the source and echo were varied from 8 to 128 ms. IDL's were chosen for both pulses from Gaussian distributions with $\mu = 0$ dB and $\sigma = 4$ dB. Dependent variables included normalized target weight, proportion correct, and the proportion of responses predicted from the weights. Although target weight and proportion correct generally increased with increasing non-target frequency and echo delay for both target conditions, the effects were always larger when the echo served as the target. The superiority of performance when judging echoes vs sources will be discussed in terms of recency effects in binaural hearing.

2:20

4pPP2. Within- and across-channel integration of information for the precedence effect. Bernhard U. Seeber (Audio Information Processing, Technische Universität München, Arcisstrasse 21, Munich 80333, Germany, seeber@tum.de)

Erv Hafter has with his team investigated the spectral and temporal integration of binaural information to learn about binaural hearing in situations with multiple sounds and reflections. For these studies, the Simulated Open Field Environment (SOFE), a loudspeaker setup in an anechoic chamber, was created. Beginning with an overview of the SOFE, I will present results on the spectral density and bandwidth of long-duration complex tones needed for the precedence effect to occur, the ability to correctly locate a sound in the presence its delayed copy. The hypothesis is that the precedence effect cannot be evoked with a single low frequency tone, because the addition of its delayed copy alters the interaural phase and thus its location. A larger bandwidth is needed to stabilize localization at the lead either through integrating binaural information across frequency or through extracting information from the temporal envelope. Results show that a stable precedence effect could not be obtained at or below 1 Bark bandwidth, and that at least two tones per critical band over 2 Bark are required. The echo threshold increases with increasing bandwidth or spectral density, suggesting that within and across-channel information is combined.

2:40

4pPP3. Where am I, where is the sound source? Yost A. William, Xuan Zhong, Anbar Najam (Speech and Hearing Sci., ASU, PO Box 870102, Tempe, AZ 85257, william.yost@asu.edu),

Locating sound sources in the everyday world often involves listeners and sources who change location. Since sound source localization cues are relative changing when the listener or source moves, veridical and accurate sound source localization when the listener moves requires that the auditory brain "knows" where the listener is in 3-dimensional space. Experiments were conducted in a sound-deadened room with 36 loudspeakers located on a 5-foot radius sphere and a rotating chair. The listener was rotated in accelerating, decelerating, and constant velocity conditions and was either sighted (eyes open) or blind folded (eyes closed). The sound source was either fixed at one location, or the sound (100-ms, broadband noise) changed position from one loudspeaker to the next (along a 24-loudspeaker azimuth circle) in an accelerating, decelerating, or constant velocity manner. In all cases but one, listeners were able to perceive the loudspeaker presenting the sound in the same way they did when the listener was stationary. When the eyes were closed (no visual cues) and the chair was rotated at constant velocity (no semicircular canal vestibular cues), listeners badly misperceived sound source locations. The results indicate that veridical sound source localization requires visual and/or vestibular information.

3:00–3:15 Break

3:15

4pPP4. Revisiting the loudness of sounds with asymmetric attack and decay. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

Stecker and Hafter [J. Acoust. Soc. Am. **107**, 3358–3368 (2000)] compared the loudness of sounds whose envelopes had a fast attack and a slow decay (designated F-S) and a slow attack and a fast decay (designated S-F). They found that, for sinusoidal and broadband noise carriers, S-F stimuli were louder than F-S stimuli of equal energy. They argued that this effect could not be explained by current models of loudness and that the loudness effect may be related to the parsing of auditory input into direct and reverberant sound. Subsequent work has shown that the differences in loudness between F-S and S-F stimuli can be partially accounted for by the loudness model of Glasberg and Moore [J. Audio Eng. Soc. **50**, 331–342 (2002)], which incorporates a form of temporal averaging that is asymmetric in time. However, the model does not account for the context effect found by Stecker and Hafter. The largest differences in loudness between F- S and S-F stimuli occurred after pre-exposure to a F-S stimulus. This may happen because, when successive sounds have similarly slow decays, the decaying part is attributed to room reverberation and contributes less to loudness.

3:35

4pPP5. Attention and the refinement of auditory expectations. Psyche Loui (Psych. and Neurosci. and Behavior, Wesleyan Univ., 12 Chestnut St., Cambridge, Massachusetts 02139, ploui@wesleyan.edu)

Although traditional approaches in psychoacoustics emphasize bottom-up processes of hearing, a lasting approach of the Hafter lab has been to merge the bottom-up view with top-down influence of cognitive and training-related factors. Much of this is embodied in a research program on auditory attention: the listener's ability to extract relevant features of the auditory scene (Hafter *et al.*, 2007). I joined the Hafter lab with interests in auditory attention and music perception. In work conducted in the lab we observed effects of training on attentive processing of musical harmony—musicians are slower to respond to musically unexpected harmonies but faster to respond to expected harmonies, suggesting that long-term training refines the expectations that are built up from lifelong exposure to music in one's culture. Armed with this knowledge, we further asked if these expectations could be learned. Using a non-Western musical scale, we showed rapid learning of perceptual patterns, which can be modeled as a reduction in uncertainty and an increase in correlation with the auditory environment. In subsequent work we combined electrophysiology, neuropsychology, and neuroimaging to show that the learning mechanisms that sharpen auditory expectations are rapid, flexible, and depend on neural connectivity that also subserves linguistic processes.

3:55

4pPP6. Attention and effort during speech processing. Anastasios Sarampalis (Univ. of Groningen, Grote Kruisstraat 2/1, Groningen 9712TS, Netherlands, a.sarampalis@rug.nl)

The concepts of attention and effort are not new in auditory science, yet it is only recently that we have started systematically studying their involvement in speech processing. The task of deciphering speech can vary in its cognitive demands, depending on a number of factors, such as sound quality, the state of the auditory and cognitive systems, room acoustics, and the semantic complexity of the signal itself. Understanding these interactions does not only shed light on the functions supporting speech processing, but is also critical when it comes to evaluating new hearing aids and cochlear implant strategies. This presentation will describe work that is either based on Erv Hafter's ideas while I was at UC Berkeley or inspired by discussions with him in subsequent years. Its central theme is the measuring of listening effort and its implications to digital signal processing, cochlear implants, aging, or understanding non-native languages.

4:15

4pPP7. Spatial release of the cognitive effort of understanding speech in multi-talker environments. Sridhar Kalluri, Jing Xia, Nazanin Nooraei, and Brent Edwards (Starkey Hearing Res. Ctr., Starkey Hearing Technologies, 2150 Shattuck Ave., Ste. 408, Berkeley, CA 94704, sridhar_kalluri@starkey.com)

Ervin Hafter was prescient in recognizing the need for taking into account top-down cognitive processing for understanding the interaction between auditory perception and cognition. His insight, that signal processing such as noise reduction may modify cognitive demands without changing auditory performance, led to a seminal study in collaboration with the Starkey Hearing Research Center which showed that hearing aid technology can reduce the cognitive effort of understanding noisy speech. Inspired by Erv's insight, we are studying if increasing the spatial separation between competing talkers reduces the cognitive effort needed to listen in multi-talker environments. Following the lead of Erv's seminal study, performance on a simultaneous secondary task, in our case visual tracking, is a measure of the cognitive effort consumed by the primary task of understanding target speech. Our results show that spatial separation can reduce cognitive effort even when it does not give further improvement in speech intelligibility over existing segregation cues. These results suggest that a measure of cognitive effort is useful for assessing the benefit of hearing technology that improves spatial segregation. This is an important finding because the measure addresses benefit along a dimension that is not captured in standard assessment of speech reception performance.

4:35–4:45 Concluding Remarks

Session 4pSA

Structural Acoustics and Vibration and Architectural Acoustics: Structural and Acoustic Response Due to Impulsive Excitation

Marcel Remillieux, Chair
Los Alamos National Lab., Geophysics Group (EES-17), M.S.: D446, Los Alamos, NM 87545

Chair's Introduction—1:30

Invited Papers

1:35

4pSA1. Inelastic deformation and failure of partially strengthened profiled blast walls. Arash Soleiman-Fallah, Ebuka Nwankwo (Civil Eng., Imperial College London, Skempton Bldg., Exhibition Rd., South Kensington Campus, London SW18 4GR, United Kingdom, as3@imperial.ac.uk), Genevieve S. Langdon (Mech. Eng., Univ. of Cape Town, Cape Town, South Africa), and Luke A. Louca (Civil Eng., Imperial College London, London, United Kingdom)

Blast walls that separate the potentially hazardous regions of the topside on an offshore platform were designed to resist lower loads than those envisaged today thus it is desirable to upgrade their blast resistance in a cost-effective and non-intrusive manner. One proposal is to retrofit the existing blast walls partially with centrally located composite patches. This study presents an assessment tool, which provides understanding of the effect of a composite patch on the blast resistance of blast walls. Numerical simulations of a proposed retrofitted wall are performed to gain insight into the failure progression of the wall *ab initio*. Damage in the composite patch was considered, and the numerical simulations showed that fiber fracture did not occur thus there was no significant loss of in-plane stiffness and strength. Based on these observations, the rapid assessment tool, analytically formulated to incorporate the effect of the composite patch which strengthens the wall and moves the plastic hinge locations away from the wall centre to the composite-steel edge, is deemed a suitable tool. The assessment tool and the numerical simulations are partially validated by the experimental results. The tool runs quickly and provides reasonable accurate predictions for the deformation response of the walls.

1:55

4pSA2. Predicting the response of structures to transient shock loading. Mauro Caresta, Robin S. Langley, and Jim Woodhouse (Eng., Univ. of Cambridge, Trumpington St., Cambridge CB21PZ, United Kingdom, maurorestaca@yahoo.it)

This work concerns the prediction of the response of an uncertain structure to a load of short duration. Assuming an ensemble of structures with small random variations about a nominal form, a mean impulse response can be found using only the modal density of the structure. The mean impulse response turns out to be the same as the response of an infinite structure: the response is calculated by taking into account the direct field only, without reflections. Considering the short duration of an impulsive loading, the approach is reasonable before the effect of the reverberant field becomes important. The convolution between the mean impulse response and the shock loading is solved in discrete time to calculate the response at the driving point and at remote points. Experimental and numerical examples are presented to validate the theory presented for simple structures such as beams, plates, and cylinders.

2:15

4pSA3. Characterization of a spark source focused by an ellipsoidal reflector. Xiaowei Dai, Yi-Te Tsai (Civil, Architectural, and Environ. Eng., The Univ. of Texas at Austin, 301 E. Dean Keeton St., M.S. C1747, Austin, TX 78712, jyzhu@mail.utexas.edu), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Jinying Zhu (Civil, Architectural, and Environ. Eng., The Univ. of Texas at Austin, Austin, TX)

Air-coupled ultrasonic non-destructive testing (NDT) provides an ideal solution for rapid scanning of large specimens. Unfortunately, despite decades of research, many challenges remain to render air-coupled ultrasonic methods a broadly effective sensing modality for high impedance materials due to low energy transmission between air and the solids being inspected. In this paper, we present experimental results and theoretical analysis of an electrical spark source focused by an ellipsoidal reflector. This acoustic source, which generates a short duration, high amplitude signal in air, is of high interest for air-coupled NDT for high impedance materials and has been shown to excite wave motion in concrete without contact. Theoretical modeling using weak shock theory and the KZK equation is used to predict the temporal and spatial features of the pressure field in the region of the geometric focus. We also present a series of experimental studies that characterize the spark generated acoustic wave in both free-field and the focused conditions. The bandwidth and directivity of the focused spark source are shown to be adjustable by changing the spark gap size and the reflector geometry. Finally, experimental results from three reflectors made of different material and geometries are presented.

2:35

4pSA4. Dynamic acousto-elasticity in Berea sandstone: Influence of the strain rate. Jacques Riviere (EES-17, Los Alamos National Lab., MS D446, Los Alamos, NM 87545, riviere_jacques@yahoo.fr), Thibault Candela, Marco Scuderi, Chris Marone (GeoSci., Penn State Univ., University Park, PA), Robert Guyer, and Paul A. Johnson (EES-17, Los Alamos National Lab., Los Alamos, NM)

In comparison with standard nonlinear ultrasonic methods such as frequency mixing or resonance based measurements that allow one to extract average, bulk variations of modulus and attenuation versus strain level, dynamic acousto-elasticity (DAE) allows to obtain the elastic behavior over the entire dynamic cycle, detailing the full nonlinear behavior under tension and compression, including hysteresis and memory effects. To improve our understanding of these phenomena, this work aims at comparing static and dynamic acousto-elasticity to evaluate the influence of strain rate. To this purpose, we perform acousto-elasticity on a sample of Berea sandstone and a glass beads pack, oscillating them from 0.001 to 10 Hz. These results are then compared to DAE measurements made in the kHz range. We observe that the average decrease in modulus increases with frequency, meaning that conditioning effects are higher at high strain rate, when relaxation characteristic time is higher than the oscillation period. This result, together with previous quasi-static measurements (Claytor *et al*., GRL 2009) showing that the hysteretic behavior disappears when the protocol is performed at a very low strain-rate, confirms that a rate dependent nonlinear elastic model has to considered for a more complete description (Gusev *et al*., PRB 2004).

2:55–3:10 Break

3:10

4pSA5. Semi-analytical study of interfacial stresses in adhesively bonded single lap joints subject to transverse shock loading. Ebuka Nwankwo, Arash Soleiman-Fallah, and Luke A. Louca (Civil Eng., Imperial College London, Skempton Bldg., Exhibition Rd., South Kensington Campus, London sw7 2az, United Kingdom, en208@imperial.ac.uk)

Debonding in adhesively bonded lap joints is a detrimental failure mode contingent upon the level of stresses developed in the adhesive. A semi-analytical model is developed to estimate the peel and shear stresses in an isotropic elastic adhesive in a single lap joint subjected to transverse shock loads. The proposed semi-analytical model is an extension of existing mathematical models to study the coupled transverse and longitudinal vibrations of a bonded lap joint system. The adhesive is modeled as an isotropic material in ABAQUS. The interfacial stresses obtained by finite element simulations were used to validate the analytical model. The maximum peel and shear stresses predicted by the analytical model in the adhesive were found to correlate well with the maximum stresses predicted by the corresponding numerical models. The peel stresses in the adhesive were found to be higher than shear stresses, a result which is consistent with intuition for transversally pulse loaded joints. The semi-analytical model is able to predict the maxim stresses in the edges where debonding initiates due to the highly asymmetrical stress distribution as observed in the finite element simulations and experiment. The stress distribution under uniformly distributed transverse pulse loading was observed to be similarly asymmetric.

Contributed Papers

3:30

4pSA6. Structural infrasound from a barge collision with the Mississippi River Bridge. Anna M. Miller, Richard D. Costley, Henry Diaz-Alvarez, Mihan H. McKenna, and Christopher P. Simpson (GeoTech. & Structures Lab., US Army Engineer Res. & Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, anna.m.millter@usace.army.mil)

The Mississippi River Bridge in Vicksburg MS is a 7 span cantilever bridge 3389 feet long by 68.5 ft wide and is part of the Interstate-20 corridor. On 23 March 2011 around 1:30 pm, a barge moving downstream struck a pier of the bridge. Infrasound stations located at the Waterways Experiment Station (WES) detected the impact. There are indications from the infrasound signatures that infrasound was radiated from the bridge and also from disturbances on the surface of the water (waves or eddies) that resulted from the collision. Finite Element (FE) models of the bridge and pier were developed to simulate the response of the bridge due to the barge impact. It was possible to identify those portions of the infrasound signature produced by vibration of the bridge deck. A synopsis of the accident will be presented along with the recorded infrasound signatures. Results from the dynamic structural model of the bridge will be discussed and related to the infrasound signature.

3:45

4pSA7. A hybrid numerical model for the exterior-to-interior transmission of impulsive sound through three-dimensional, thin-walled elastic structures. Marcel C. Remillieux (Geophys. Group (EES-17), Los Alamos National Lab., Los Alamos, NM 87545, mcr1@lanl.gov), Stephanie M. Pasareanu (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA), and U. Peter Svensson (Dept. of Electronics and Telecommunications, Norwegian Univ. of Sci. and Technol., Trondheim, Norway)

Exterior propagation of impulsive sound and its transmission through three-dimensional, thin-walled elastic structures, into enclosed cavities, are investigated numerically in the framework of linear dynamics. A hybrid model was developed in the time domain by combining the advantages of two existing numerical tools: (i) exterior sound propagation and induced structural (façade) loading are computed using the image-source method for the reflected field (specular reflections) combined with an extension of the Biot-Tolstoy-Medwin method for the diffracted field, (ii) the fully coupled vibro-acoustic response of the interior fluid-structure system is computed using a truncated modal-decomposition approach. In the model for exterior sound propagation, it is assumed that all surfaces are acoustically rigid. Since coupling between the structure and the exterior fluid is not enforced, the model is applicable to the case of a light exterior fluid and arbitrary interior fluid(s). The structural modes are computed with the finite element method using shell elements. Acoustic modes are computed analytically assuming acoustically rigid boundaries and rectangular geometries of the enclosed cavities. This model is verified against finite-element solutions computed with a commercial software package for the cases of rectangular structures containing one and two cavities, respectively.

4:00

4pSA8. Theoretical and experimental analysis of shock isolation using non linear stiffness. Diego Ledezma, Jose de Jesus Villalobos-Luna (Facultad de Ingenieria Mecanica y Electrica, Universidad Autonoma de Nuevo Leon, Av Universidad sn, San Nicolas de los Garza, Nuevo Leon 66456, Mexico, diego.ledezmard@uanl.edu.mx), Neil Ferguson (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom), and Michael Brennan (Departamento de Engenharia Mecanica, UNESP, Ilha Solteira, Brazil)

Shock vibration is a common problem involving large forces and accelerations, usually resulting in nonlinear behavior. Normally shock isolation systems are modeled after linear passive stiffness elements intended to absorb the energy from the shock, and viscous damping in order to dissipate the energy. An experimental system with low dynamic stiffness is proposed and presented in this work, using a combination of positive stiffness and negative stiffness given by magnetic forces. The experimental prototype is

based on a theoretical model involving a cubic restoring force. The results presented shown how such an isolator provides improved shock isolation, pointing out advantages and disadvantages.

4:15

4pSA9. A study on structural vibration of washing machine with gyroscope. Gyu Sung Na, Young Jin Park, Yoon Sik Park (Mech. Eng., KAIST, 291 Daehak-ro, Yuseong-gu, Daejeon 305-701, South Korea, joycap01@kaist.ac.kr), and Jeong Hoon Kang (Digital Appliances, Samsung Electronics, Suwon, South Korea)

This paper is proposed about reducing the transient vibration of drum type washing machine. The vibration of washing machine is caused by unbalanced cloths in high spinning drum, and the displacement of tub is maximized at transient range about 3 Hz (180 rpm). The dynamic model of washing machine is include a diaphragm. In this study, the displacement of tub is decreased by using gyroscope system. Multibody dynamic model of washing machine include gyroscope is designed and the vibration of tub have been reduced by the gyroscope system.

THURSDAY AFTERNOON, 5 DECEMBER 2013 PLAZA B, 1:30 P.M. TO 5:00 P.M.

Session 4pSCa

Speech Communication: Language Description

Natasha L. Warner, Chair
Dept. of Linguist., Univ. of Arizona, Box 210028, Tucson, AZ 85721-0028

Contributed Papers

1:30

4pSCa1. Best practices in measuring vowel merger. Jennifer Nycz (Dept. of Linguist., Georgetown Univ., 1437 37th St. NW, Washington, DC 20057, jn621@georgetown.edu) and Lauren Hall-Lew (Linguist. and English Lang., The Univ. of Edinburgh, Edinburgh, United Kingdom)

Vowel mergers are some of the most well-studied sound change phenomena, particularly in varieties of English. But although sociolinguists, dialectologists, and phoneticians are all interested in providing accurate and precise descriptions of an individual speaker's participation in a near-merger (or near-split), the methods for doing so vary widely, especially for researchers analyzing naturalistic corpora. In this paper, we consider four current methodological approaches to representing and assessing vowel distance and overlap: Euclidean distances between averages, Pillai-Bartlett trace (Hay *et al.*, 2006), mixed effects regression modeling (Nycz 2013), and the spectral overlap assessment metric (Wassink 2006). We compare the advantages and disadvantages of each by applying all four methods to three separate data sets. These represent low vowel realizations by speakers from three different studies of English variation: one undergoing merger (COT and CAUGHT in San Francisco, California), one undergoing split, for the same contrast (COT and CAUGHT among Canadians in New York City), and one undergoing split, but for a different contrast (TRAP and BATH among Scots in England). By comparing the similarities and differences between the data sets themselves, as well as the differing analytic motivations for quantifying speaker-specific vowel overlap, we conclude with practical recommendations.

1:45

4pSCa2. The use of high rise terminals in Southern Californian English. Amanda Ritchart (Linguisit., UCSD, 1 Miramar St. #929004, La Jolla, CA 92092, aritchart@ucsd.edu) and Amalia Arvaniti (English Lang. & Linguist., Univ. of Kent, Kent, United Kingdom)

This study investigates High Rise Terminals (HRTs), i.e., utterance-final rising pitch movements, as used in Southern Californian English (SoCalE), examining the phonetics and phonology of HRTs and their relation to pragmatic functions. Twelve female and 11 male speakers were recorded during a map task and in the retelling of a sitcom scene. HRTs were coded for discourse function (statement, question, confirmation request, floor holding) based on context. The alignment of the pitch rise start was measured from the onset of the utterance's last stressed vowel, and the rise's final Hz value was recorded. In HRTs used for statements, the rise started within the stressed vowel, while in questions it started after vowel offset. Together with the low F0 on the stressed syllable, this pattern suggests that statements have a L*L-H% melody while questions have L*H-H%. Confirmation requests and floor holding were more variable in alignment. Consistent differences in pitch scaling were found in the order: questions, confirmation requests > floor holding > statements. Females used HRTs more often than males, and their HRTs showed greater pitch excursion and later alignment. In conclusion, SoCalE uses different HRT melodies than other varieties and maintains a distinction between HRTs for statements and questions.

2:00

4pSCa3. Phonetic shift across narrative and quoted speech styles. Paul De Decker (Dept. of Linguist., Memorial Univ. of Newfoundland, St. John's, NF A1B 3X8, Canada, pauldd@mun.ca)

Qualitative descriptions of speech accompanying verbs of quotation (e.g., "She was like, 'I'm not going in there!'") characterize quoted speech as a mimetic performance (Buchstaller 2003, Winter 2002) with "selective depictions" of the quotees words (Clark and Gerrig 1990, 1). The current study aims to quantify the performative and mimetic nature of quoted speech by comparing acoustic measurements of 539 vowel productions obtained through narratives of personal experience (Labov and Waletzky 1967) as told between friends. First and second formant frequencies were measured at the temporal midpoint of each vowel using PRAAT5.3 (Boersma and Weenink 2012), normalized using the BARK method and compared in a one way ANOVA in SPSS. The dependent variables were F0, F1, F2, and duration while gender of speaker and lexical set key word (Wells 1982) served as the independent variables. Results indicate that mainly female speakers showed phonetically shifted vowel quality features when moving from narrative style speech to quoting voices for characters in their stories. This specific type of phonetic alteration across speech styles is examined as a type of "speech play" (Sherzer 2002) and its role in story-telling is examined further.

2:15

4pSCa4. Mon voice registers: Acoustics and laryngeal control. Arthur S. Abramsonn, Mark K. Tiede (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, arthur.abramson@uconn.edu), and Theraphan Luangthongkum (Linguist., Chulalongkorn Univ., Bangkok, Thailand)

Mon is spoken in many villages in Thailand and Myanmar. The dialect of Ban Nakhonchum, Ratchaburi Province, Thailand, has two voice registers, modal and breathy, phonation types that, along with other phonetic properties, commonly distinguish registers. Four native speakers recorded several repetitions of 14 randomized words (seven minimal pairs) for acoustic analysis. We used a subset of these pairs for listening tests to verify the perceptual robustness of the distinction. Four speakers, three of the original ones and one new one, were also recorded using electroglottography (EGG) while repeating the word set several times. The listening tests showed the distinction to be robust. Acoustic analysis of both sets of recordings was done using the UCLA VoiceSauce program. Differences in noise component (ratio of harmonics to noise and cepstral peak prominence), spectral slope, fundamental frequency, and formant frequencies all differ across the registers. For analysis of the EGG data we used the UCLA EGGWorks program to obtain closure quotient (CQ) measures. CQ was significantly different for all four speakers with higher values for the modal register. The salience of these cues in maintaining the register distinction will be discussed. [Work supported by NIH grants and the Thailand Research Fund.]

2:30

4pSCa5. The case for strident vowels. Matthew Faytak (Linguist., Univ. of California Berkeley, 2632 San Pablo Ave., Apt. A, Berkeley, CA 94702, mf@berkeley.edu)

I present evidence for a natural class of strident vowels characterized by significant high-frequency energy caused by turbulent airflow. This turbulent airflow is not incidental to a narrow articulatory "tube," as is common for high vowels (Klatt 1975, Ohala and Solé 2010). Rather, all share an acoustic signal consistent with turbulence produced by a jet of air angled so as to strike an obstacle anterior to the jet, as seen in strident fricatives (Shadle 1990). The Mandarin words "four" [sz□] and "ten" [ʂʐ̩□] provide examples of these vowels at alveolar and retroflexed places of articulation; I provide further examples, including labiodentals, from my research on the Kom language of Cameroon. Vowels are essential for clear and reliable perception of speech, as their low spectral center of gravity, high intensity, and open articulatory configuration allow for the realization of cues to perception of neighboring, less intrinsically perceptible consonantal segments (Liberman *et al*., 1954). Strident vowels, with their higher center of gravity, lower intensity, and consonant-like articulation, call into question the nature of this modulation, suggesting the utility of broader definitions for a sufficiently perceptible modulation in the speech signal (Kawasaki-Fukumori and Ohala 1997).

2:45–3:00 General Discussion

3:00–3:30 Break

3:30

4pSCa6. Falling diphthongs have a dynamic target while rising diphthongs have two targets: Acoustics and articulation of the diphthong production in Ningbo Chinese. Fang Hu (Inst. of Linguist., Chinese Acad. of Social Sci., 5 Jian Guo Men Nei St., Beijing 100732, China, hufang@cass.org.cn)

It is controversial whether diphthongs are phonologically vowel sequences and thus phonetically have two targets or diphthongs are phonologically vowel phonemes that contrast with monophthongs and thus phonetically have one dynamic target. Chinese dialects are generally known as having a rich inventory of diphthongs, and typically there are both falling and rising diphthongs. This paper is an acoustic and articulatory study on the diphthongs in Ningbo Chinese. The acoustic data are from 20 speakers and the lingual kinematic data are collected from 6 speakers by using EMA. The acoustic results show that both the onset and offset elements have comparable formant frequency patterns to their corresponding target citation vowels in a rising diphthong, but in a falling diphthong, only the onset element has a comparable formant frequency pattern to its corresponding target citation vowel whereas the offset element is highly variable. The articulatory results further reveal that diphthong onset is better controlled than diphthong offset, and more importantly, diphthong production is constrained by the general articulatory-to-acoustic relations. It is generally concluded that in Ningbo Chinese, rising diphthongs have two targets and can thus be understood as vowel sequences while falling diphthongs have only one dynamic target and should be treated as a single vowel phoneme.

3:45

4pSCa7. Regional effects on Indian English sound and timing patterns. Hema Sirsa (Linguist., Univ. of Oregon, 179 NW 207th Ave., Beaverton, OR 97006, hsirsa@uoregon.edu)

English, spoken as second/third language by millions of speakers of India (IE), differs from other varieties of English in terms of sound patterns. Most descriptions of IE have focused on the influence of native language on IE (Wiltshire and Harnsberger, 2006; Sirsa and Redford, submitted). Some studies have also pointed out that IE may be evolving into multiple varieties due to social and political pressures (Wiltshire, 2005), but so far dialectal differences have not been explored independently from L1 influences. The current study aimed to do just this. Regionally based segmental and suprasegmental differences were investigated in IE spoken by Hindi and Telugu speakers, with equal numbers of speakers of each L1 recruited from two geographical sites (Delhi and Hyderabad). Analysis of IE sound patterns indicated that speakers from Hyderabad had more fronted /u/ than Delhi speakers, whereas Delhi speakers had longer phrase-final lengthening then Hyderabad speakers. Speakers from the two sites also had different rhythm structures and speech rates. These results support the suggestion that IE is evolving into multiple varieties, and that these varieties are not simply a function of different L1s.

4:00

4pSCa8. Tonal alignment in Deori. Shakuntala Mahanta (Dept. of Humanities and Social Sci., Indian Inst. of Technol. Guwahati, Guwahati, Assam 781039, India, shakunmahanta@gmail.com), Indranil Dutta (Dept. of Computational Linguist., English and Foreign Lang. Univ., Hyderabad, Andhra Pradesh, India), and Prarthana Acharyya (Dept. of Humanities and Social Sci., Indian Inst. of Technol. Guwahati, Guwahati, Assam, India)

This paper reports on the results from an experiment on tone in Deori, a language spoken by about 20,000 people in Assam (India). Data from 10 speakers where the target word bearing the tonal contrast appeared in the sentence medial position is presented. Time-normalized pitch of different words shows that words may have a lexically specified high or low tone. A high tone may contrast with a low tone, but its phonetic implementation of rise or fall in a disyllabic word depends on whether the syllable on which the contrast appears is initial or final. A tonal contrast on the first syllable leads to a falling contour, but when the contrastive tone appears on the second syllable of a disyllabic word then the tonal contour is falling. Exceptions to this pattern appear in closed disyllables where a steep rise in either the low or high tone is not observed. A high or a low tone may also contrast with a word which is not specified with any tone, in which case there is no rise or fall. Statistical analyses show that Deuri tones exhibit phonetic properties that are dependent on contextual factors like syllable position and segmental properties.

4:15

4pSCa9. An acoustic description of Chemehuevi. Benjamin V. Tucker (Linguist., Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca)

Chemehuevi, a member of the Uto-Aztecan language family, is spoken along the Colorado River in both Arizona and California. The language is extremely endangered with fewer than five known speakers, all over the age of 50. Chemehuevi is classified following Miller *et al*. (2005) as a dialect of Colorado River Numic along with Southern Paiute and Ute. The present work offers a general description of the acoustic characteristics of the Chemehuevi phoneme inventory based on both an analysis of archival (3 female speakers recorded by: Major, 1969; Tyler, 1972; Press, 1973–1974) and current field recordings (1 male speaker recorded by: Penfield, Serratos, and Tucker, 2005–2006, 2010) of the language. To date, there is little acoustic analysis of Numic languages available. Vowel characteristics are analyzed by extracting duration and the first three formant frequencies. Consonants are also investigated using relevant acoustic measures (such as voice-onset time and centroid frequency). Additionally, the present acoustic analysis is compared to early descriptions of the phoneme inventory and provides evidence regarding the nature of the vowel inventory (is /e/ a phoneme), location of stress, idiolectal differences, and word final voiceless vowels.

4:30

4pSCa10. Acoustic features of upper necaxa totonac ejective fricatives. Rebekka Puderbaugh and Benjamin V. Tucker (Dept. of Linguist., Univ. of AB, 2-40 Assiniboia Hall, University of AB, Edmonton, AB T6G 2E7, Canada, puderbau@ualberta.ca)

The purpose of this study is to investigate the acoustic properties of a class of sounds known as ejective fricatives in Upper Necaxa Totonac (UNT), a Totonac-Tepehua language of northern Puebla, Mexico, and to relate these sounds to those in other languages. Ejective fricatives are an exceedingly rare class of sounds found in only a relatively small number of the world's languages. This study attempts to clarify the nature of the acoustic signal of these sounds in UNT, whose historical origins have been reconstructed as former fricative plus glottal stop clusters [Beck, 2006, Univ. of Alberta Working Papers, 1], use the acoustic data to verify whether these segments are in fact canonical ejectives and propose future directions for further research. Analyses of the segments in question include duration and center of gravity of the fricative portions, presence or absence of any periods of silence surrounding the segments, durations of such silences, and effects on pitch, amplitude, duration, and formants of neighboring vowels. Due to the variable nature of the realization of laryngeal phonemes in UNT, pitch, amplitude, and voice quality of both preceding and following vowels were analyzed as well.

4:45–5:00 General Discussion

THURSDAY AFTERNOON, 5 DECEMBER 2013 PLAZA A, 1:00 P.M. TO 5:00 P.M.

Session 4pSCb

Speech Communication: Speech Production II (Poster Session)

Jelena Krivokapic, Chair
Linguist., Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109-1220

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors the opportunity to view other posters authors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

4pSCb1. Phonological encoding and articulatory duration in spontaneous speech. Melinda Fricke (Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720-2650, melindafricke@berkeley.edu)

Many studies have found that word duration is correlated with a word's contextual predictability in conversational speech. Lindblom (1990)'s Hypo/Hyperarticulation Theory, Jurafsky *et al.* (2001)'s Probabilistic Reduction Hypothesis, and Aylett and Turk (2006)'s Smooth Signal Redundancy Hypothesis all suggest that such differences in duration are due to processes occurring primarily at the lexical level. The present study, however, suggests that these differences may be attributable to processes occurring at the phonological level. In this study, mixed modeling is used to examine the voice onset time and rime duration of monosyllabic /p t k/ words in spontaneous, connected speech (the Buckeye Corpus; Pitt *et al.*, 2007). Higher contextual predictability given the previous word is found to be associated with shorter VOT, while higher contextual predictability given the following word is associated with shorter rime duration. VOT also varies according to the number and type of a word's phonological neighbors; words with more neighbors overlapping in the rime have significantly longer VOT, while words with more neighbors overlapping in the initial CV have significantly shorter VOT. These results motivate a model of speech production that assumes both lexical-phonological feedback and positional encoding of segments (e.g., Sevald and Dell, 1994).

4pSCb2. Syntactic probability affects morpheme durations. Clara Cohen (Linguist., Univ. of California at Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, cpccohen@berkeley.edu)

This project investigates the role of syntactic predictability on the duration of morphemes. Previous research has found that contextually predictable speech units tend to be shorter in duration. Usually, such research focuses on the duration of words or syllables, and context is defined in terms of n-gram strings. This project extends such research by investigating the role of syntactic context on the production of morphemes. Are more probable morphemes also reduced when they are more probable in a given syntactic context? Russian sentences with quantified subject noun phrases (e.g., "three chairs") allow both singular and plural verb agreement suffixes, but the probability of observing one or the other is variable. In this study, Russian speakers produced sentences with either singular or plural agreement of varying probability. The lists were counterbalanced so that each sentence was produced with both the singular and plural suffix. Although there was no difference in duration for singular suffixes, high-probability plural suffixes were shorter than low-probability plural suffixes. Differences in whole-word durations cannot account for these differences in suffix durations. These results suggest that contextual predictability in the form of agreement relations can affect the phonetic production of the morphemes that encode those relations.

4pSCb3. Reduction and frequency analyses of vowels and consonants in the Buckeye speech corpus. Byunggon Yang (English Education, Pusan National Univ., 30 Changjundong Keumjunggu, Pusan 609-735, South Korea, bgyang@pusan.ac.kr)

In a casual conversation American speakers tend to talk fast and to reduce or change sounds of phonetic symbols defined in an English dictionary which we would find in citation speech style. This study examined how much reduction of pronunciation Americans make from the dictionary prescribed symbols to the real speech ones and how frequently Americans use vowels and consonants in the Buckeye speech corpus. The corpus was recorded by 40 American male and female subjects for an hour per each subject. Results were as follows: First, the Americans produced a reduced number of vowels and consonants in daily conversation. The reduction rate from the dictionary transcriptions to the real transcriptions was around 38.2%. There was not much difference between the vowels and consonants in the reduction. Second, the Americans used more front high and back low vowels while 78.7% of the consonants accounted for stops, fricatives, and nasals. This indicates that the segmental inventory has nonlinear distribution in the speech corpus. From those results we conclude that there is a substantial reduction in the real speech from the dictionary symbols and suggest that English educators consider pronunciation education reflecting the real speech data.

4pSCb4. The effect of high and low variability conditions on phonetic convergence. Grant McGuire (Linguist., Univ. of California, Santa Cruz, Stevenson Faculty Services, Santa Cruz, CA, gmcguir1@ucsc.edu), Molly E. Babel, and Jamie Russell (Linguist., Univ. of Br. Columbia, Vancouver, BC, Canada)

Studies of phonetic convergence using single-word auditory naming tasks offer insight into how variability in stimuli affect the translation from speech perception to speech production. In this paper, we report on an experiment which compares phonetic convergence in single-word production between high variability (mixed talker condition) or low variability (blocked talker condition) using five female model talkers' voices for the task. Twenty female participants participated in a production task where they produced baseline tokens and shadowed model talker productions in either the high or low variability condition. Phonetic imitation was quantified using listener judgments in an AXB similarity rating task where a model token was compared to a shadower's baseline and shadowed token. The results indicate a trend towards more convergence in the low variability condition, but this was highly affected by model voice; one model voice was spontaneously imitated more in the high variability condition than the low variability condition. Several socio-cognitive tests were administered to shadowers, and continued analyses of the data will explore whether these individual socio-cognitive measures predict shadowers' predispositions toward phonetic convergence.

4pSCb5. The effect of task difficulty on phonetic convergence. Jennifer Abel (Linguist., Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, jennifer.abel@alumni.ubc.ca)

Cognitive workload is the information processing load a person experiences when performing a task; the more difficult the task, the greater the cognitive workload. Increased task difficulty/cognitive workload has been shown to have an effect on several acoustic measures of speech such as amplitude, word/syllable/utterance duration, and f0. To date, the task difficulty-speech production link has only been studied in individuals. This study examines the effect of different levels of task difficulty on phonetic convergence within dyads collaborating on a task. Dyad members had to build identical LEGO® constructions without being able to see each other's construction, and with each member having half of the picture-based instructions required to complete the construction. Three levels of task difficulty were created, based on the number of pieces in each step of the construction—easy (2 pieces/step), medium (3 pieces/step), and hard (4 pieces/step)—with five dyads at each level (30 participants total). Dyads were audio- and video-recorded, and completed working memory and mental rotation tests and personality questionnaires prior to the task. Acoustic analysis and AXB perception studies are underway to examine the amount and type of convergence in each dyad.

4pSCb6. Word-internal ambisyllabic consonants are codas. Karthik Durvasula, Ho-Hsin Huang, and Rose Merrill (Michigan State Univ., B330 Wells Hall, East Lansing, MI 48824, durvasul@msu.edu)

The syllabic affiliation of ambisyllabic consonants (e.g., the word-medial consonants in happy and Danny) is unclear. Research on ambisyllabic consonants has revealed an inconsistent set of phonetic correlates (Krakow, 1989; Turk, 1994; Gick, 2004). While some suggest they behave as onsets or codas (but not both simultaneously), others suggest their gestural durations are intermediate between onsets/codas. At least some of the research is based on comparisons of the ambisyllabic consonants to word-edge onsets/codas. However, comparisons to word-edges are confounded by the fact that such consonants undergo domain-edge related changes (Fougeron, 2001; Keating *et al.*, 2003b). Here, we control for this confound, and compare ambisyllabic consonants to word-medial onsets and codas. We conducted an experiment on 10 native English speakers, who produced 15 repetitions at three different speech rates of 16 English words (8 test, 8 filler) that consisted of the nasal consonants [n or m] in one of four positions: word-medial onset, word-medial coda, word-final coda, and ambisyllabic context (e.g., gamete, gamble, gam, and gamma). The results suggest: (1) Consistent with previous research, there are durational differences between word-medial and word-final nasal codas; (2) Ambisyllabic consonants clearly pattern with the word-medial nasal codas and are significantly different from the nasal onsets.

4pSCb7. The articulation of derived affrication in American English. Jae-Hyun Sung (Linguist,, Univ. of Arizona, 814 E 9th St., Apt. 14, Tucson, AZ 85719, jhsung@email.arizona.edu)

Affrication of coronal stops before /ɹ/ is commonly observed in English. For instance, /t/ in "tree" and /d/ in "dream", in which coronal stops precede /ɹ/, are often realized as affricated stops (i.e., [ʧɹi] instead of [tɹi]; [ʤɹim] instead of [dɹim]). Given that morphological structures and frequency of words play a critical role in many coarticulatory processes (Bush, 2001; Ernestus *et al.,* 2006; Myers and Li, 2009), the present study investigates whether the degree of derived affrication before /ɹ/ is influenced by different morphological structures and frequency of words and phrases. This study uses ultrasound imaging and audio recordings of seven native speakers of American English to examine the articulatory aspect of derived affrication. Comparisons of the degree of affrication show significant differences among words in various environments, in which tautomorphemic words and high-frequency words and phrases lead to greater degree of affrication. Furthermore, the gestural patterns of various morphological and frequency conditions are highly individualized.

4pSCb8. Temporal coordination of sibilants in Polish onset clusters. Manfred Pastätter and Marianne Pouplier (Inst. of Phonet. and Speech Processing, LMU, Schellingstraße 3, Munich 80799, Germany, manfred@phonetik.uni-muenchen.de)

In this study we employ the gestural syllable model to examine cluster-vowel timing in Polish sibilant initial (SI = {/ʃm-, ʃp-, sp-, sk-/}) and sibilant final (SF = {/mʃ-, pʃ-, ps-, ks-/}) onset clusters. In this model, the timing of a complex onset is evaluated relative to a simplex onset and it is predicted that timing relations between onset and vowel should be invariant independently of onset complexity (Browman and Goldstein, 2000). Articulatory data of three speakers show that SI clusters conform to the predicted timing pattern in terms of a globally organized onset cluster relative to the vowel ("c-center"). This is compatible with previous findings for several languages. For SF clusters, however, there are considerable timing differences between complex and corresponding simplex onsets. This suggests that SF clusters are coordinated differently (and inconsistently) to the following vowel compared to SI clusters, as also reported previously for Romanian (Marin, 2013). We investigate to which extent this difference between SI and SF clusters is related to sibilants' high coarticulatory resistance preventing a close C-V coordination. We will present an analysis of jaw movement data to consider possible effects of jaw position constraints on the temporal coordination of clusters.

4pSCb9. Compensatory vowel shortening before complex coda clusters in the production and perception of German monosyllables. Sandra Peters and Felicitas Kleber (Inst. of Phonet. and Speech Processing, LMU, Schellingstr. 3, Munich 80799, Germany, sandra@phonetik.uni-muenchen.de)

The main aim of the present study was to investigate incremental coda compensatory shortening in the production and perception of German monosyllables including factors such as accentuation (i.e., accented vs deaccented) and codas' manner of articulation (i.e., sonorant vs obstruent). Ten speakers produced real German words like /klɪŋ/ and /klɪŋt/. We measured the duration of the vowel and the first coda consonant (C1). Overall there was no significant vowel shortening effect. However, some speakers did show vowel shortening and even more so in accented tokens with sonorant codas. Additionally, all speakers tended to shorten C1. In a subsequent experiment, we tested whether listeners compensate for different degrees of vowel and C1 shortening. 21 subjects judged which vowel in selected pairs such as /klɪŋ/—/klɪŋt/ was longer. In two thirds of all pairs, listeners perceived vowels before simplex codas as longer—even in pairs with equal segment durations. While this overall bias indicates perceptual vowel shortening before complex codas, listeners nevertheless show tendencies to compensate for non-shortened vowels before complex sonorant codas, i.e., they were perceived as longer. Although there was less vowel shortening in production, listeners showed perceptual vowel shortening and some tendencies toward compensation in contexts that favor shortening.

4pSCb10. Revisiting the consonantal voicing effect: Flapping in American English. Ylana Beller-Marino and Dianne Bradley (Linguist., CUNY Graduate Ctr., 360 1st Ave., Apt. 6D, New York, NY 10010, ybeller@gc.cuny.edu)

It is long-acknowledged that the consonantal voicing effect—whereby vowel duration is greater preceding voiced vs voiceless consonants (e.g., *rib*/*rip*)—is larger in English as compared with other languages, and that the effect's magnitude generally decreases in multisyllabic forms (e.g., *rabbit*/*rapid*). The current study examines consonantal voicing effects in the multisyllabic environment, crucially contrasting non-coronal with coronal cases (e.g., *rider*/*writer*). In American English, the latter are subject to a flapping process that surface-neutralizes the voicing distinction. Hence, while both phonological and phonetic sources for a vowel-duration difference are available in non-coronals, flapping eliminates the phonetic source in coronals. We present an analysis of critical vowel durations in elicited productions (target words uttered in a carrier phrase), and confirm the pattern expected if the post-vocalic consonant's place of articulation matters: the consonantal voicing effect was entirely reliable for non-coronals, but not for coronals. More detailed analyses set aside tokens where flapping failed to apply, and found that the consonantal voicing effect might be altogether absent with coronal place. We speculate that, here, the voicing distinction may have been neutralized in phonological representation, whether that distinction is a matter of orthography (*doodle*/*duty*) or is also supported by morphological alternation (*rider*/*writer*).

4pSCb11. Phonetics as a complement to phonology in the Canadian Shift. Matt H. Gardner (Linguist., Univ. of Toronto, Toronto, ON, Canada) and Rebecca Roeder (English, Univ. of North Carolina at Charlotte, 9201 University City Blvd., Charlotte, NC 28223, rroeder@uncc.edu)

Previous accounts of the Canadian Shift have interpreted this diachronic change in vowel pronunciation as a purely phonetic consequence of the low back LOT-THOUGHT vowel merger; however, such an analysis does not transparently explain the strong connection between the (phonological) low back merger and the subsequent (phonetic) retraction of the TRAP vowel in the acoustic vowel space. This paper addresses this issue by presenting an analysis of the shift that combines the approaches of Modified Contrastive Specification theory and the Contrastive Hierarchy—two phonological frameworks—with phonetic insights from Vowel Dispersion-Focalization theory. We propose that the catalyst of the Canadian Shift is a three-way vowel merger, in combination with a simultaneous change in the underlying feature specifications of the TRAP vowel. This results in a phonology that allows for the TRAP and DRESS vowels to succumb to the influence of the phonetic principles of dispersion and focalization. This hypothesis is illustrated by comparison of data from 59 speakers in Thunder Bay, Ontario, and Industrial Cape Breton, Nova Scotia. Our analysis predicts that a Canadian Shift-type phonetic change will occur in any North American dialect of English where the PALM-LOT-THOUGHT merger occurs, unless an intervening phonological change alters systemic contrasts.

4pSCb12. Perceptual and prosodic factors in cluster timing: Manner, order, and syllable position effects in Polish consonant clusters. Marianne Pouplier and Manfred Pastätter (Inst. of Phonet., LMU, Schellingstr. 3, Munich 80799, Germany, pouplier@phonetik.uni-muenchen.de)

We investigate timing in Polish tautosyllabic C1C2 clusters differing in manner, consonant order, and syllable position. Hoole *et al.* (2013) reported for German that perceptual constraints may condition timing differences in /kn/ and /kl/ clusters due to the nasal but not the lateral obscuring the preceding stop burst. Using articulography, we test this hypothesis for a variety of Polish onset clusters (C1={m, p, k}, C2={n, l, r}). Results from three speakers confirm a significant influence of both C1 and C2 on timing patterns. A C1 nasal shows more overlap than a stop. For C2, /l/ shows more overlap than /n/, consistent with the German results. However, the relative difference between C2=/n/ and C2=/l/ holds independently of whether C1 is a nasal or a stop, contra the perception hypothesis. Further, it is known from several languages that onset clusters overlap less than coda clusters, yet this observation has been confounded by the sonority conditioned change in consonant order in onset/coda. Polish has several clusters which do not change order as a function of syllable position, allowing us to tease these two factors apart. If consonant order is kept constant, there is no significant syllable position effect on C-C timing.

4pSCb13. Entrainment by vocal effort: Coordination in postural control and speech production. Robert Fuhrman (Dept. of Linguist, Univ. of Br. Columbia, Totem Field Studios 2613 West Mall, Vancouver, BC V6T1Z4, Canada, robert.a.fuhrman@gmail.com), Adriano Vilela Barbosa (Elec. Eng., Universidade Federal de Minas Gerais, Belo Horizonte, Minas Gerais, Brazil), and Eric Vatikiotis-Bateson (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

The biomechanical coupling between the systems implicated in speech production, postural control, and respiration suggests that some degree of coordination in the form of postural entrainment should take place given excessive task demands in the speech domain, as has been previously reported [Vatikiotis-Bateson, *et al.*, (2009) Proceedings of ESCOM 2009]. In this context, this work assesses the time-varying coordination and entrainment among multiple components of the postural control system that result from the modulation of vocal effort level in both read and spontaneous speech. Correlation map analysis is used to quantify the coordinated patterns of interaction between speech acoustics and a variety of related physical systems, including lower body postural configuration (center-of-pressure calculated from force plate measurements), head motion (measured with OPTOTRAK), and visual motion (optical flow from video). Cross-correlation analysis of the measured signals shows that modulation of vocal effort leads to both qualitative and quantitative shifts in the coordinative dynamics of the system, uniformly resulting in better spatiotemporal coordination and reduced rhythmic pattern complexity as vocal effort is increased.

4pSCb14. Acoustic correlates of consonant gesture timing in English. Elliot Selkirk and Karthik Durvasula (Linguist. and Lang., Michigan State Univ., B331 Wells Hall, East Lansing, MI 48824, selkirke@msu.edu)

There is extensive research on the organization of syllable-structure as indexed by the relative timing of the articulators (Browman and Goldstein, 1988; Byrd, 1995; Shaw *et al.*, 2011 inter alia). The research suggests consonants in complex onsets (in words such as scream, stream…) are aligned to a single position called the C-center, the mean of the midpoints of the onset consonants. However, such research typically uses very expensive articulatory equipment (X-ray Microbeam, Electromagnetic Articulography…). This restricts the research to a few laboratories across the world with access to such technology. Here, we explore the possibility of using acoustic measurements, which are cheaper and more accessible, for

such research. We conducted an experiment on 6 native speakers of English, who produced 12 repetitions of 24 English words (12 test, 12 filler) that varied in the number of onset consonants (C1, C1C2, C1C2C3) in three different vowel contexts. Paralleling previous studies, the results show that onset consonants align with the C-center even in acoustic measurements. The results suggest acoustic data has at least some meaningful information about gestural organization. Therefore, they prompt the (nuanced) use of acoustic techniques to study such effects.

4pSCb15. Coarticulation and contrast in static and dynamic models of second formant trajectories. Indranil Dutta (Dept. of Computational Linguist., The English and Foreign Lang. Univ., Tarnaka, Osmania University Campus, Hyderabad 500605, India, indranil.dutta.id@gmail.com) and Charlie Redmon (School of Lang. Sci., The English and Foreign Lang. Univ., Hyderabad, Andhra Pradesh, India)

Real (Stevens *et al.*, 1966) and virtual F2 locus (Sussman *et al.*, 1991) measures are presented for Malayalam lingual plosives. We show that in distinguishing voiceless coronals (dental, alveolar, and retroflex) in VC:V sequences, F2 onsets derived from first-order locus equations (LEs) show only partial delineation of the contrast. The dental-alveolar contrast is effectively maintained, but retroflex and alveolar stops show no significant difference in F2 onset. Following Lindblom and Sussman's (2012) examination of LEs as a measure of relative coarticulatory resistance, we report F2 slopes for the three coronal stops in VC and CV transitions to assess the implications of this metric in Malayalam. Our findings on the ordering of slope values from steepest to flattest did not follow predictions based on expectations of relative articulatory complexity; namely, alveolars generated a flatter slope than retroflexes, despite Dart and Nihalani's (1999) demonstration that the retroflex gesture is more complex within the coronals. These results, when compared with temporal measures from exponential models of formant trajectories at consonant implosion and release (i.e., transition velocity and projected F2 locus), suggest a necessary distinction between coarticulation-based place of articulation categorization and formant transition cues utilized in maintaining stop place contrasts.

4pSCb16. Estimation of vocal tract input impedance at the glottis from formant measurements. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

It is well known that the mapping from articulation to acoustics is many-to-one or many-to-many, so that so-called "articulatory-acoustic inversion" is a challenging problem. What has not been noted, however, is that the input impedance from the glottis can be determined from the inverted articulatory configuration regardless of whether this configuration is accurate. This can be useful as a step toward estimating the acoustic load on vocal fold vibration during phonation. The theory and procedure for thus obtaining estimates of the vocal tract input impedance is presented, its relation to the Mermelstein/Schroeder method is shown, and its limitations are discussed. Finally, experiments with synthetic and naturally produced vowels are presented and discussed. It is shown that the estimated input impedance is accurate up to the highest measured formant, with the largest deviations centering around the formant frequencies due to errors in formant measurements and the handling of acoustic losses.

4pSCb17. Modeling the listener? What resets acoustic durations of repeated English words. Prakaiwan Vajrabhaya and Vsevolod Kapatsinski (Linguist., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, pvajrabh@uoregon.edu)

Listener-based accounts of speech production claim that speakers modify their speech based on their evaluation of the listener's state of knowledge (Lindblom, 1990). In line with this, repeated words shorten when they have been previously said to the same listener (Fowler, 1988); however, repetition across an episode boundary in a narrative does not lead to decreased acoustic duration (Fowler *et al.*, 1997). We replicate Fowler *et al.*'s story boundary effect and extend the study by testing whether a switch in listener has an additional effect on word duration. Speakers were asked to tell and retell the same story in the sequence of (A) listener 1/(B) listener2/(C) listener1 again (Galati and Brennan, 2010). We expect word durations to reset when the speaker starts a new narrative, especially when there is a switch in listener. In other words, word durations should be comparable in conditions (A) and (B), but shorter in (C), since the listener in condition (C) has heard the story before. Acoustic data from 20 American English native speakers have been collected and transcribed; data analysis is ongoing. This study is intended to shed light on the interplay between production economy and the need to transmit information.

4pSCb18. Acoustic analysis of initial consonants in the California Syllable Test. E. W. Yund, Marc Ettlinger (Res. 151/MTZ, VA Medical Ctr., 150 Muir Rd., Martinez, CA 94553, yund@ebire.org), and David L. Woods (Neurology, VA Medical Ctr., Martinez, CA)

The goal of the present study is to conduct an acoustic analysis of onset consonants to identify the spectrotemporal features that distinguish them from each other and to identify acoustic consonant variations that produce the observed patterns of perceptual confusions seen in young- and older-normal-hearing listeners [J. Acoust. Soc. Am. **127**, 1609–1623 (2010); JRRD **49**, 1277–1292 (2012)]. We used the California Syllable Test (CaST) token set, which includes 40 exemplars of each initial consonant for each of three vowels and six talkers. The CaST measures recognition of 20 initial and 20 final consonants in speech-spectrum noise with each consonant presented at a 67%-correct signal-to-noise ratio (SNR). Time-normalized spectrograms are computed for each exemplar (from consonant onset to the end of the formant transition) by varying the time-spacing of the FFT spectral lines in proportion to the exemplar duration. Quantitative comparisons among normalized spectrograms of correctly recognized and confused exemplars at a range of SNRs suggest explanations for specific consonant confusions. The long-term goal is to apply this analysis to understand the effects of hearing loss and HAs on consonant perception and to predict consonant confusion patterns obtained with the CaST in normal-hearing and hearing-impaired listeners.

4pSCb19. The role of the posterior cricoarytenoid muscle in phonation. David Berry, Dinesh K. Chhetri, and Juergen Neubauer (Surgery, UCLA, 31-24 Rehab., Los Angeles, CA 90095-1794, daberry@ucla.edu)

The posterior cricoarytenoid muscle (PCA) is generally considered to be a respiratory muscle. Indeed, as the sole abductor of the glottis (i.e., the only laryngeal muscle with the capability of opening the true vocal folds), paralysis of the PCA may lead to asphyxiation. While the PCA muscle also appears to play a role in phonation, a consensus has not been reached among voice scientists regarding its precise role in the control of fundamental frequency, phonation threshold pressure, and other phonatory variables. Using a new developed method of graded stimulation to the laryngeal muscles, Chhetri, Neubauer, and Berry (2012) explored the role of the cricothyroid muscle (CT), thyroarytenoid muscle (TA), and the lateral cricoarytenoid and interarytenoid muscle complex (LCA + IA) on fundamental frequency, phonation threshold pressure and glottal posturing. The present study augments the previous study by also investigating the influence of the PCA muscle on these same phonatory variables. Similar to the adductor muscles, it is shown that the PCA muscle introduces new possibilities for achieving multiple phonation types at a given fundamental frequency.

4pSCb20. Anatomic development of the hyo-laryngeal complex in humans from birth to 95 Years: An imaging study. Houri K. Vorperian (Waisman Ctr., Univ. of Wisconsin, Waisman Ctr., 1500 Highland Ave., # 427, Madison, WI 53705, vorperian@waisman.wisc.edu), Yuan Wang (Dept. of Statistics, Biostatistics & Medical Informatics, Univ. of Wisconsin, Madison, WI), Reid B. Durtschi (Waisman Ctr., Univ. of Wisconsin, Madison, WI), Meghan M. Cotter (Dept. of Neurosci., Univ. of Wisconsin, Madison, WI), Ray D. Kent (Waisman Ctr., Univ. of Wisconsin, Madison, WI), Moo K. Chung (Dept. of Statistics, Biostatistics & Medical Informatics, Univ. of Wisconsin, Madison, WI), and Lindell R. Gentry (Dept. of Radiology, Univ. of Wisconsin, Madison, WI)

During postnatal development, the hyo-laryngeal complex descends in the pharyngeal cavity primarily during early childhood, followed by a secondary descent during puberty, particularly in males. The purpose of this

study is to quantify the descent of the human hyo-laryngeal complex, as well as its relational growth to other functionally related structures such as the epiglottis and the tongue from birth to 95 years. Anatomic data secured from 902 medical imaging studies (482 males; 420 females) were analyzed in two phases: (I) A detailed assessment of developmental changes of the hyo-laryngeal complex and functionally related structures from birth to 19 years using 771 imaging studies. I(I) Comparison of similar measurements between three adult groups (ages 20-to-45 years; 45–70 years, and 70–95 years) using 131 images. Findings indicate that: (a) growth/descent of the hyo-laryngeal complex is non-linear and protracted, displaying a predominantly somatic growth pattern; (b) small sex differences in growth are present during childhood, with increased differences emerging at about 10 years, and maximal differences present by 19 years; and (c) there appears to be a coincident relational growth of functionally related structures. These novel findings are of clinical significance, and enhance the understanding of vocal tract development. [NIH-Grants R01DC6282, P-30HD03352.]

4pSCb21. Signal detection of lipreading visemes using two dimensional and three dimensional images. Rita Quigley and Al Yonovitz (The Univ. of Montana, 32 Campus Dr., Missoula, MT 59812, rita.quigley@mso.umt.edu)

The actual process by which the lipreader translates the lip movements they identify into a message is very complex. The lip movements observed represent only fragments of the complete message. The main purpose of this study is to investigate (1) the ability of lipreaders to use visual information alone to identify phonemes in varying contexts including nearby coarticulation effects and vowel neighborhoods; (2) lipreading responses using the effect of improved video presentation through 3D video, providing better and more realistic video presentation; and (3) the use of a novel measurement technique, i.e., a signal detection two-alternative-forced choice method of subject response that should provide measures of discrimination between phonemes including "visemes." Video recordings were made in both 2D and 3D formats. This 3D image presented more realistically the movements such as lip-rounding and micro-movements of viewable articulators in three dimensions. Subjects with normal hearing were presented these video presentations. A Two-Alternative-Forced-Choice (2AFC) paradigm was used. The consonants were viewed with various vowel contexts. D-prime values were obtained for both the 2D and 3D videos. Particular consonant clusters were more discriminable in 3D.

4pSCb22. Speaking tongues are always braced. Bryan Gick, Blake Allen (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T1Z4, Canada, gick@mail.ubc.ca), Ian Stavness (Comput. Sci., Univ. of SK, Saskatoon, SK, Canada), and Ian Wilson (CLR Phonet. Lab., Univ. of Aizu, Aizuwakamatsu, Japan)

Bracing the tongue against rigid vocal tract surfaces (i.e., teeth or palate) has been suggested to be important in facilitating certain kinds of tongue movements [Stone, J. Acoust. Soc. Am. **81**, 2207–2218 (1990)]. However, previous studies have generally sought bracing in only a narrow range of phonetic contexts, resulting in a widespread view of bracing as an occasional state, peculiar to specific sounds or sound combinations. The present study uses electropalatography (EPG) as well as ultrasound imaging and electromagnetic articulometry (EMA) to describe tongue bracing in continuous speech passages, finding that the tongue is almost constantly braced against lateral surfaces during running speech. Analysis of archival data from the male and female speakers of American English in the KayPENTAX Palatometer Database (Model 4333) shows that they brace the tongue continuously, except during a small percentage of low vowels, and during a larger percentage of instances of /l/. Additional measures using all three devices, as well as biomechanical simulations using ArtiSynth (www.artisynth.org), provide further insight, indicating that the tongue also braces against the central palate and/or lower jaw, and that bracing points slide anteroposteriorly across speech sounds. These results suggest that bracing is a constant and necessary aspect of tongue motor control.

4pSCb23. Direct characterization of collagen recruitment in the human vocal fold lamina propria. Bahar Fata (Head & Neck Surgery, UCLA, 1000 Veteran Ave., Rm. 33-59, Los Angeles, CA 90024, bahar.fata@gmail.com), Julio L. Vergara (Physiol., UCLA, Los Angeles, CA), and Zhaoyan Zhang (Head & Neck Surgery, UCLA, Los Angeles, CA)

The goal of this study is to develop a structurally based constitutive model to characterize the structure-function relationship of the vocal folds. Compared to phenomenological models, structurally based constitutive models allow direct prediction of changes in the mechanical behavior of the vocal folds as a result of aging or pathological conditions. The first significant step in developing such a mathematical model is to characterize the structural arrangement and load-bearing behavior of the collagen and elastin fibers, the two most mechanically significant structural proteins in the vocal fold. A micro horizontal uniaxial tensile system has been designed and coupled with the non-invasive multi-photon microscopy method. The load-bearing or recruitment behavior of collagen was characterized by simultaneously measuring the waviness of the collagen fibers and stress of the cover layer at different strain conditions. The structural arrangement of the collagen and elastin fibers in the different layers of the lamina propria were also quantified. The results of this study will directly elucidate the specific contributions of the elastin and collagen fibers to the vocal fold mechanical behavior under uniaxial tension. [Work supported by NIH.]

4pSCb24. Electropalatography examination of groove width in Russian. Phil Howson (Linguist., The Univ. of Toronto, 100 St. George St., Toronto, ON M5S 3G3, Canada, phil.howson@mail.utoronto.ca)

Previous studies have indicated a difference between voiced and voiceless pairs of consonants with respect pre-constriction vocal tract volume. This article utilizes electropalatography (EPG) to examine the anterior and posterior groove width of palatalized and non-palatalized fricative pairs in Russians in order to observe different degrees of pre-constriction vocal tract volume. Measurements were taken at the point of maximum constriction using Articulate Assistant software. Higher degrees of contact with the palate were taken to indicate smaller pre-constriction vocal tract volume. The results (based on a single speaker), indicate a significant difference in the degree of contact with the palate between the voiced and voiceless pairs of non-palatalized fricatives. However, the palatalized consonants indicated no significant difference in the degree of contact with the palate. The findings suggest that the smaller vocal cavity created by the secondary articulatory gesture for palatalization is sufficient to facilitate voicing and frication; in the case of the voiced fricatives, the sub-glottal pressure is adjusted to permit vibration of the vocal cords. The findings further suggest that speakers adhere to the principle of minimal articulatory effort when producing speech.

4pSCb25. An analysis of tongue shape during parkinsonian speech. Katherine M. Dawson (Speech-Language-Hearing Sci., City Univ. New York Graduate Ctr., 365 5th Ave., New York, NY 10016, kdawson2@gc.cuny.edu), Khalil Iskarous (Linguist., Univ. of Southern California, Los Angeles, CA), and D. H. Whalen (Speech-Language-Hearing Sci., City Univ. New York Graduate Ctr., New Haven, Connecticut)

Parkinson's disease (PD) is a neurological disorder characterized by the degeneration of dopaminergic neurons. Speech impairments in PD are characterized by slowed muscle activation, muscle rigidity, variable rate, and imprecise consonant articulation. Complex muscular synergies are necessary to coordinate tongue motion for linguistic purposes. People with PD may show an altered rate of change in tongue shape during vowel to consonant transitions and may also ultimately attain less complex consonantal tongue shapes than controls during speech. In order to test this hypothesis, five PD participants, five older controls and five younger controls (all French-speaking) were imaged using ultrasound. They produced consonant–vowel–consonant word stimuli. Transitions analyzed were vowel-to-liquid (/l/) and vowel-to-velar stop. Tongue shapes were analyzed using a method designed to infer complexity by analogy with the bending energy of a thin shell [Young, Walker, and Bowie, Info. Control **25**(4), 357–370 (1974)]. This method works by integrating the squared curvature of a piece-wise polynomial function fitted to the extracted discrete tongue contour. Results will be discussed in terms of shape change during the transition and maximal consonantal shape attained between PD and control subjects.

4pSCb26. Characterizing post-glossectomy speech using real-time magnetic resonance imaging. Christina Hagedorn (Dept. of Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, chagedor@usc.edu), Adam Lammert (Viterbi School of Eng., Univ. of Southern California, Los Angeles, CA), Yihe Zu, Uttam Sinha (Dept. of Otolaryngol., Head and Neck Surgery, Keck School of Medicine, Univ. of Southern California, Los Angeles, CA), Louis Goldstein (Dept. of Linguist, Univ. of Southern California, Los Angeles, CA), and Shrikanth S. Narayanan (Viterbi School of Eng., Univ. of Southern California, Los Angeles, CA)

We investigate articulatory behavior in post-glossectomy speech using real-time magnetic resonance imaging. Our data reveal that listeners judge speech produced by partial-glossectomy patients as atypical when the surgical procedure affected the oral tongue. Speech produced by patients whose procedure affected the base of tongue, however, was judged as typical. We observe that preservation and compensation mechanisms are exhibited by the patients with atypical speech. They preserve appropriate modulation of F1 using tongue and/or jaw height despite inability to appropriately modulate F2 due to the reduced size and/or mobility of the tongue. Further, durational differences between tense and lax vowels are maintained. The preservation of these features serves as evidence in support of a framework within which individual gestural parameters are independently controlled; when achievement of a particular parameter specification (e.g., constriction location) is compromised, the remaining (e.g., constriction degree, activation duration) are unchanged. Compensatory behavior is exhibited when coronal tongue movement has been impeded and is exemplified by (i) production of labiodental stops in place of target coronal stops and laterals and (ii) forming a velar constriction to produce frication in place of the alveolar frication for /s/.

4pSCb27. Interspeaker variability in relative tongue size and vowel production. Adam Lammert (Signal Anal. and Interpretation Lab., Univ. of Southern California, 3740 McClintock Ave., Rm. 400, Los Angeles, CA 90089, lammert@usc.edu), Christina Hagedorn (Dept. of Linguist., Univ. of Southern California, Los Angeles, CA), Michael Proctor (Marcs Institute/School of Humanities and Lang., Univ. of Western Sydney, Sydney, NSW, Australia), Louis Goldstein (Dept. of Linguist., Univ. of Southern California, Los Angeles, CA), and Shrikanth Narayanan (Signal Anal. and Interpretation Lab., Univ. of Southern California, Los Angeles, CA)

The tongue varies across speakers in terms of the proportion of the overall speech production apparatus that it occupies. Differences in tongue size have the potential to result in speaker-specific articulatory strategies for shaping the vocal tract area function and, in turn, individual patterns of vowel acoustics. The present study examines the interplay between relative tongue size and vowel production using real-time magnetic resonance imaging with synchronous audio. Two populations of native American English subjects are considered, one containing healthy adult speakers with no relevant pathologies, and another containing speakers who had undergone glossectomy as treatment for tongue cancer. All subjects were imaged in the midsagittal plane while reading phonetically balanced English sentences. The size of the tongue and the speech production apparatus were quantified from an overall average posture, and their ratio was correlated with the shape of the vowel space in terms of acoustics (e.g., formant frequencies), constrictions (i.e., location and degree of minimum constriction), and parameterized vocal tract cross-distance functions. Results indicate that relative tongue size can be used to explain and predict observable interspeaker differences in vowel production.

4pSCb28. Control of voice intensity. Karin Sjögren, Emmma Ström (Dept. Logopedics, Phoniatrics and Audiol., Lund Univ., Lund, Sweden), and Anders Lofqvist (Dept. Logopedics, Phoniatrics and Audiol., Lund Univ., 300 George St., New Haven, Connecticut 06511, lofqvist@haskins.yale.edu)

This study examined the control of voice intensity using acoustic and aerodynamic recordings. A total of 34 subjects participated half of them with and half without song training, 21 females and 13 males. The subjects produced the syllable sequence /papapa/ while the acoustic signal, the oral air flow, and the oral air pressure were recorded using the Kay-Pentax Phonatory Aerodynamic System. The oral pressure provided an estimate of the subglottal pressure. A measure of glottal flow resistance was calculated as the ratio between subglottal pressure and oral air flow.Three different voice levels were used, normal, reduced, and increased; the change between the normal level and the two others was required to be 6–10 dB. Overall, an increase in voice intensity was associated with increased subglottal pressure and glottal flow resistance with only a small increase in air flow. A comparison between the subjects with and without song training showed those with training to produce higher intensities, to use higher subglottal pressure, but lower glottal flow resistance. Female voices had lower subglottal pressure and lower flow rates but higher glottal resistance than male voices.

4pSCb29. Menstrual cycle-dependent plasticity of auditory-vocal integration in vocal pitch production. Hanjun Liu, Xiaoxia Zhu, and Yang Niu (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., 58 Zhongshan 2nd Rd., Guangzhou, Guangdong 510080, China, lhanjun@mail.sysu.edu.cn)

Considerable evidence suggests that auditory function can be influenced by gonadal steroids (estradiol and progesterone), but whether there is a sex hormonal modulation of auditory-vocal integration in vocal production remains unknown. The present event-related potential (ERP) study sought to examine the behavioral and neurophysiological processing of auditory feedback during self-produced vocalization across the menstrual cycle. Eleven young Mandarin-native female speakers with regular menstrual cycle were tested during the menstrual, follicular, and luteal phases. Subjects heard their voice pitch-shifted 50 or 200 cents while producing a vowel sound /u/. Vocal compensations and ERPs in response to pitch perturbations as well as estradiol and progesterone concentrations were measured at three different phases. The behavioral findings showed significantly larger magnitude of vocal compensation at the menstrual phase in comparison to follicular or luteal phase. As to the neurophysiological findings, P2 amplitude in the luteal phase was significantly smaller compared to that in the menstrual and follicular phase. These results demonstrate the menstrual cycle-related effect on the behavioral and neurophysiological processing of auditory feedback in vocal pitch production, suggesting that the integration between auditory and vocal motor system can be modulated by the estradiol and progesterone levels across the menstrual cycle.

4pSCb30. A computer assisted pronunciation training system. Kwansun Cho and John G. Harris (Elec. and Comput. Eng., Univ. of Florida, University of Florida, Gainesville, FL 32611, kscho@cnel.ufl.edu)

A computer assisted pronunciation training (CAPT) system is implemented for native Korean speakers who are learning American English. The CAPT system is designed to help a Korean adult learner improve his/her production and perception of American English front vowels (/i, I, ε, æ/) since these vowels are the most difficult for Korean learners due to the different phonetic systems of the two languages. The CAPT system provides a learner a learning session mimicking a live interaction between teacher and student as well as a practice session triggering a learner's interest in continued practice. Pedagogically meaningful activities such as listen-and-repeat, minimal-pair-comparison, target-sound-isolation, and record-and-play are utilized in the learning session. During the learning session, the CAPT system analyzes a monosyllabic word including one of the target front vowels spoken by a learner and gives instantaneous personalized feedback. During the practice session, the CAPT system provides real-time games that are fun but also provide the necessary perception and articulation practice.

4pSCb31. Measurable acoustic variants as predictors of progress in speech therapy. Kathleen Siren (Speech-Lang. Pathology/Audiol., Loyola Univ. Maryland, 4501 North Charles St., Baltimore, MD 21210, ksiren@loyola.edu)

Despite the availability of free, user-friendly acoustic analysis programs, acoustic documentation of speech sound change during speech therapy is rarely mentioned in speech research literature. Thus, the utility of acoustic analysis to document speech change over time in children with speech errors

is unknown. A prior study documented children's /s/ productions as they progressed through speech therapy and compared spectrographic analysis of productions to clinicians' perceptual judgments of accuracy. Results indicated a greater number of /s/ productions were judged accurate based on visual (acoustic) analysis vs auditory (perceptual) judgment for all clients, particularly during a period of time when clients' /s/ productions were becoming more frequently accurate. The purpose of this current investigation is to identify the measurable acoustic features of /s/ production that indicate when an individual's /s/ production is improving even when the productions are still heard as incorrect. By comparing productions identified as correct visually but incorrect auditorily to productions identified the same both visually and auditorily, this study identifies acoustic variants that are indicative of subtle improvements in production not yet identifiable by adult listeners. These subtle, yet measurable, acoustic characteristics may identify potential acoustic markers for sound maturation in children's disordered speech production.

4pSCb32. Assessment of head reference placement methods for optical head-movement correction of ultrasound imaging in speech production. Kevin Roon, Eric Jackson (CUNY Graduate Ctr., 365 Fifth Ave., Ste. 7107, New York, NY 10013, kroon@gc.cuny.edu), Hosung Nam, Mark Tiede (Haskins Labs., New Haven, CT), and Doug H. Whalen (CUNY Graduate Ctr., New York, NY)

One method of quantification of tongue movement using ultrasound imaging during speech production requires determination of tongue position relative to the palate, corrected for probe and head motion so that successive frames can be meaningfully compared. This method involves placing infrared emitting diodes (IREDs) on a "tiara" attached to the participant's head (Whalen *et al.*, 2005). An alternative is to attach IREDs directly to the participant's skin. In either case, the IREDs can potentially move relative to the participant's skull. The present study examined movement with both methods for simple utterances, a read paragraph, and spontaneous speech. The amount of IRED movement observed using both methods allowed identification of regions where IREDs should be affixed on a participant's skin to minimize movement when the direct application method is used. Results of simulations showing the effects of this IRED movement on the calculated head-movement correction of the tongue images are presented. Given the results of these simulations, guidelines are proposed for establishing thresholds that can be used to determine whether a given experimental trial should be included based on the amount of reference IRED movement. Differences in movement due to linguistic content or style will also be discussed.

4pSCb33. Using an exponential sine sweep to measure the vocal tract resonances. Bertrand Delvaux and David Howard (Dept. of Electronics, Univ. of York, Heslington, York, York YO10 5DD, United Kingdom, bertrand.delvaux@gmail.com)

The vocal tract (VT) of a singer acts as a filter on the acoustic output from the vibrating vocal folds, enhancing several frequency bands whose peaks are called formants. The nature of these formants is characterized by the shape and dimensions of the VT and they are numbered with the first formant being the lowest in frequency. Perceptually, the first (F1) and second (F2) formants indicate the vowel being sung while the third (F3), fourth (F4) and fifth (F5) relate to the timbre or tone color of the output sound. It is therefore relevant to the understanding of the vocal organ to be able to measure the resonances of the tract with precision. Here we apply the exponential sine sweep method used in room acoustics to VT models and replicas. We use an exponential sine sweep as the source signal for the cavity and record its output. After convolving the output signal with the appropriate inverse filter, we can separate the linear impulse response of the tract from its harmonic distortions. This method is both applied on VT models of Chiba and Kajiyama and on MRI-based molded VTs.

4pSCb34. A comparison of kinematic and acoustic approaches to measuring speech stability between speakers who do and do not stutter. Eric Jackson (The Graduate Ctr. of the City Univ. of New York, 365 5th Ave., 7th Fl., Rm. 7304, New York, NY 10016, ejackson@gc.cuny.edu), Mark Tiede (Haskins Labs., New Haven, CT), and Douglas H. Whalen (The Graduate Ctr. of the City Univ. of New York, New York, NY)

People who stutter have been found to exhibit reduced speech stability during fluent speech production relative to people who do not stutter. One index for quantifying stability that has been applied to stuttering and nonstuttering speakers is the spatiotemporal index (STI; Smith *et al.*, 1995). STI measures the consistency of repeated speech movements aligned using linear normalization. Similar stability indices based on nonlinear methods for alignment have also been reported (e.g., Lucero *et al.*, 1997). Both linear and nonlinear methods have been applied to kinematic signals in previous experiments. The present study tests the possibility that measures of stability based on acoustic signals can also be useful indicators of speech stability in adults who do and do not stutter (cf. Howell *et al.*, 2009), as using audio recordings to calculate speech variability could provide an attractive alternative for speech-language pathologists and researchers who lack access to kinematic data. In addition, both kinematic and acoustic stability are assessed with respect to effects of linguistic complexity and social factors.

Session 4pUW

Underwater Acoustics and Acoustical Oceanography: Sediment Acoustics: Modeling, Measurements, and Inversions I

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Chair's Introduction—1:15

Invited Papers

1:20

4pUW1. A discussion of possible measurement techniques for muddy sediments and of the related modeling challenges. Allan D. Pierce (P. O. Box 339, P. O. Box 339, East Sandwich, MA 02537, adp@bu.edu), Joseph O. Fayton, and William L. Siegmann (Dept. of Mathematics, Rensselaer Polytechnic Inst., Troy, NY)

Present paper draws on recent work of the late William Carey at Dodge Pond, CT, and on related modeling efforts at RPI. Carey affirmed that muddly sediments can have a substantial air bubble content. The water, solid particles (clay), and bubbles lead to a low sound speed that for low frequencies is explained by a modification of the Mallock-Wood formula. Independent measurements of mass density and sound speed should enable estimates of the fractional composition. The attenuation of sound in muddy sediments without bubbles is small, much smaller than of sandy sediments, and this is explained in terms of the card house model because of the very small size of the clay particles. The larger bubbles are randomly dispersed and have flattened shapes (also explained by the card-house model), and lead to scattering and reflection phenomena. Speculations are made as to whether inversion techniques can be devised to determine bubble shapes and size distributions. The small shear modulus of muddy segments has been tentatively explained in terms of electrostatic effects inherent to the card-house model, and this can possibly be measured by interface waves. Paper also suggests that penetrometer measurements, guided by theoretical modeling, may lead to useful inferences.

1:40

4pUW2. The high frequency environmental acoustics sediment model in the light of recent advances. Nicholas Chotiros and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas, PO Box 8029, Austin, TX 78713-8029, chotiros@arlut.utexas.edu)

The high frequency environmental acoustics sediment model (HFEVA) published in the High-Frequency Ocean Environmental Acoustic Models Handbook (APL-UW 9407), which has been widely adopted by underwater acousticians and sonar modelers, is examined in the light of recent sediment acoustic models and measurements, particularly the multiple scattering and poro-elastic models. The former indicates that the sound speeds and attenuations for the larger grain sizes (phi < -1) need to be updated, and the latter that the sediment densities for the middle range of grain sizes (1 < phi < 5) are underestimated. On the last point, the authors of the original model were aware of the problem, and for practical reasons decided to accept the understatement in the interests of achieving the correct reflection loss. The discrepancies may be alleviated by adopting a poro-elastic model with multiple scattering corrections. For practical applications, an efficient parameterization of the poro-elastic model allows the number of adjustable parameters to be reduced to a level comparable with that of simpler fluid and elastic models, while retaining all its physical advantages. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

2:00

4pUW3. Measurements of compressional wave dispersion and gradients in muddy sediments. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Jan Dettmer, Stan Dosso, and Gavin Steininger (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

Cohesive or muddy sediments have received relatively sparse attention in the ocean acoustics community—this despite the fact that they form a non-negligible fraction of the sediments found in shallow water (roughly 25%) and by far the major sediment type in the deep ocean. Seabed reflection measurements have provided some understanding about the frequency dependence of the sound speed and attenuation in muddy sediments. The evidence is for weak dispersion from 300–200,000 Hz and an approximately linear dependence of

attenuation on frequency from 300–3000 Hz. In addition, the measurements have yielded information on gradients. Surprisingly large near-surface density gradients exist that vary across the shelf. Given the large density gradients, the gradients in sound speed are curiously small, suggesting that the bulk modulus is nearly proportional to the density, at least in depth. Dispersion and gradient results are discussed for muddy sediments in various mid to outer shelf regions.

Contributed Papers

2:20

4pUW4. Issues in reverberation modeling. Dajun Tang (Appl. Phys. Lab., Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

Reverberation usually consists of two-way propagation, or forward scatter, and a single backscatter. The scattering cross section is often employed to couple the two-way propagation to obtain approximate reverberation strength. Because this approach is inherently incoherent and heuristic in nature, certain limitations to its applicability need to be elucidated. In particular, unlike backscatter problems in half-space, reverberation in shallow water involves coherent incident fields at different wavenumbers. Starting with the fundamental definition of scattering T-matrix and through examples, this paper intends to address the following issues: (1) how to incorporate coherent component of reverberation into simulations, (2) how to rigorously relate time to range for given bandwidth, and (3) how to increase computation speed through proper smoothing. [Work supported by ONR.]

2:35

4pUW5. Seismic sources in seismo-acoustic propagation models. Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, 1500 Illinois St., Golden, CO 80401, jcollis@mines.edu), Scott D. Frank (Mathematics, Marist College, Poughkeepsie, NY), Adam M. Metzler (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Robert I. Odom (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

An important generating mechanism for received underwater acoustic and seismic signals are buried or earth-bound sources. Most underwater acoustic studies involve purely compressional sources in the water column. The more complicated case of a coupled shear and compressional seismic source in the sediment has recently been implemented in an elastic parabolic equation solution [Frank *et al*., J. Acoust. Soc. Am. **133**]. In this talk, generic seismic sources including those giving shear field contributions, are contrasted in normal mode and parabolic equation solutions. Scenarios considered are for an elastic-bottom Pekeris waveguide and a canonical Arctic propagation scenario with an elastic ice cover over the ocean and an elastic basement. For the Arctic case, the source is allowed in either the ice cover or in the elastic bottom. Solutions are benchmarked for purely compressional and shear seismic sources, and their relation to the seismic moment tensor is discussed. The ultimate goal of these solutions is to allow for seismic sources capable of representing generic geophysical events.

2:50

4pUW6. Nonlinear acoustic pulse propagation in dispersive sediments using fractional loss operators. Joseph T. Maestas (Appl. Mathematics and Statistics, Colorado School of Mines, 1500 Illinois St., Golden, CO 80401, jmaestas@mines.edu)

The nonlinear progressive wave equation (NPE) is a time-domain formulation of Euler's fluid equations designed to model low-angle wave propagation using a wave-following computational domain [McDonald *et al*., J. Acoust. Soc. Am. **81**]. The wave-following frame of reference permits the simulation of long-range propagation that is useful in modeling the effects of blast waves in the ocean waveguide. However, the current model does not take into account sediment attenuation, a feature necessary for accurately describing sound propagation into and out of the ocean sediment. These attenuating, dispersive sediments are naturally captured with linear, frequency-domain solutions through use of complex wavespeeds, but a comparable treatment is nontrivial in the time-domain. Recent developments in fractional loss operator methods allow for frequency-dependent loss mechanisms to be applied in the time-domain providing physically realistic results [Prieur *et al*., J. Acoust. Soc. Am. **130**]. Using these approaches, the governing equations used to describe the NPE are modified to use fractional derivatives in order to develop a fractional NPE. The updated model is then benchmarked against a Fourier-transformed parabolic equation solution for the linear case using various sediment attenuation factors.

3:05

4pUW7. A scaled mapping approach for range-dependent seismo-acoustic propagation using the parabolic approximation. Adam M. Metzler (Appl. Res. Labs., The Univ. of Texas at Austin- ARL-Environ. Sci. Group, PO Box 8029, Austin, TX 78713, ametzler@arlut.utexas.edu), Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Parabolic equation solutions are used to accurately and efficiently model range-dependent propagation effects in ocean environments. There has been much recent interest in improving accuracy, particularly for sloping interfaces between fluid and underlying sediment layers. A translational mapping approach [Collins *et al*., J. Acoust. Soc. Am. **107** (2000)] applies a coordinate transformation in which a sloping bottom interface becomes horizontal and range dependence is mapped to the upper free surface. While accurate for small slopes, this approach introduces errors for variably sloping bathymetries since the range dependence is transformed to the surface. In this work, a scaled mapping is constructed that both transforms the sloping bottom interface to horizontal and also preserves the range-independent form of the free surface by distorting the waveguide in depth. The parabolic approximation is applied in the fully range-independent transformed domain, and the result is inverse transformed to obtain the solution in the initial range-dependent environment. Applications of this approach are given and benchmarked for seismo-acoustic propagation scenarios. [Work supported by ARL:IR&D.]

3:20

4pUW8. Volume scattering and reverberation in shallow water: A simplified modeling approach. Anatoliy Ivakin (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th, Seattle, WA 98105, aniv@uw.edu)

A simplified physics-based approach is described that allows significantly faster yet reasonably accurate estimations of volume reverberation in complex shallow water environments. An integral expression is presented for scattering intensity with a factorized integrand comprised of two kernels, the double propagator and local volume scattering coefficient. The propagator describes the local intensity and can be calculated using available models, such as PE, normal modes, or ray approximations. The scattering kernel can be specified using available volume scattering models for continuous or discrete heterogeneity of sea-water column and seabed caused by spatial fluctuations of compressibility and density, or randomly distributed discrete targets, such as bubbles, fish, shells, and others. The approach is more general than and can be used for verification of existing reverberation models. For instance, calculation of bottom reverberation is not based on using the equivalent surface scattering strength (although considers it as a particular case). Numerical examples for shallow water reverberation time series, based on a PE propagation model, are presented to estimate potential contributions of different mechanisms of scattering. The estimations provide a comparison of relative contributions of scatterers with the same scattering strengths but located at different depths in water column or in the sediment. [Work supported by the US Office of Naval Research.]

3:35

4pUW9. Computation of the field of coupled modes using split-step algorithm. Nikolai Maltsev (R&D, Frontier Semiconductor, 2127 Ringwood Ave., San Jose, CA 95131, admin@asymptotus.com)

Euler equations in the form $\partial \mathbf{F}/\partial x = \mathbf{A}(x,z)\mathbf{F}$ where 2×2 matrix $\mathbf{A}$ has elements $A_{11}=0$, $A_{22}=0$, $A_{12}=i\omega\rho$, $A_{21}=1/(i\omega\rho)(-\triangle_{yz}+(\nabla_{yz}\ \ln\rho, \nabla_{yz}) - (\omega/c)^2)$ where $\mathbf{F}=(P(\mathbf{r}), u(\mathbf{r}))^T$ are sound pressure and horizontal velocity, $c(\mathbf{r})$, $\rho(\mathbf{r})$ -sound speed and density, $\omega = 2\ \pi f$—angular frequency and $\triangle_{yz},\nabla_{yz}$ are Laplace operator and gradient in the plane (y,z), has first order with respect to x and can be integrated by split step algorithm $\mathbf{F}(x+d)=\exp(0.5\mathbf{A}(x+d)d)\exp(0.5\mathbf{A}(x)d)\mathbf{F}(x)+O(d^3)$ using local modes for computation of exponential operators. Integration is performed in one direction but, due to the structure of normal modes of operator $\mathbf{A}$, allows estimate energy reflected back on every integration step. Different examples, including irregular waveguides with ideal boundaries and Pekeris style guide with variable depth are presented.

THURSDAY EVENING, 5 DECEMBER 2012 — 8:00 P.M. TO 10:00 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees on the Acoustical Society of America will hold open meetings on Tuesday and Thursday evenings beginning at 8:00 p.m. and on Wednesday evening beginning at 7:30 p.m.

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

Committees meeting on Thursday are as follows:

Musical Acoustics	Continental 1
Speech Communication	Plaza B
Underwater Acoustics	Golden Gate 2/3