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" [ʂʐ□] 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

Nicholas P. Chotiros, Cochair
Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029

Marcia J. Isakson, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713

David P. Knobles, Cochair
ARL, UT at Austin, 10000 Burnet Rd., Austin, TX 78758

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

4p THU. PM

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