Session 4aAA

Architectural Acoustics, Noise, and Committee on Standards: Networking in Soundscapes—Establishing a Worldwide Collaboration I

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Chair's Introduction—7:45

Invited Papers

7:50

4aAA1. Progress in soundscape research requires a common agenda. Osten Axelsson (Passvaegen 30, SE-147 53 Tumba, Sweden)

It is commonly believed that progress and success in any field requires competition. This is probably true, but this belief implies that all competitors have a common view on the objectives. There would not be much competition if all parties ran off in opposite directions, striving to achieve different goals. Nor would it lead to much progress. The present session calls for networking and international collaboration in soundscape research. For such collaboration to be successful, it is critical to agree on a common agenda; a mission; an objective. Recent development in soundscape research makes evident that the objective must be practical and applicable. Our minds must be set to implementing soundscape research in practice to avoid exhausting academic debates, which tend to be ends in themselves and do not contribute to progress. Two excellent, recent examples of international collaboration in soundscape research, contributing to progress, are ISO/TC 43/SC 1/WG 54 and the European COST Action TD0804 "Soundscape of European Cities and Landscapes." Both illustrate the need for international and interdisciplinary collaboration among acousticians, architects, and urban planners to accelerate progress in soundscape research. The present paper presents possible topics for a common agenda in soundscape research.

8:10

4aAA2. Languages and conceptualization of soundscapes: A cross-linguistic analysis. Caroline Cance (INCAS3 and CIRMMT, Dr. Nassaulaan 9, Assen 9401HJ, The Netherlands, ccance@gmail.com), Catherine Guastavino (McGill Univ. and CIRMMT Montreal, QC H3A1X1, Canada), and Danièle Dubois (CNRS Univ. Paris 6, Paris, France and INCAS3, Assen, 9401HJ, The Netherlands)

In the past decade, soundscape research has emerged accounting for acoustic phenomena as they are perceived and assessed by humans. In this view, concepts and methodologies from social sciences and humanities are needed to identify the diversity of conceptualizations across time, space, and languages. Specifically, our approach relies on linguistics and psychology in analyzing how people describe their sensory experience (what is being said and how it is being said), in order to identify different conceptualizations conveyed in their discourse. We first investigate the linguistic resources available in different languages with a cross-linguistic survey of free-format verbal descriptions of acoustic phenomena in European languages (e.g., French, English, Dutch, Spanish), extending the pilot investigation by Dubois and Guastavino (2008). Then, coupling this linguistic analysis with cognitive theories on categorization, we can infer a diversity of conceptualizations for the same acoustic phenomena. This approach further allows us to overcome some limitations of current survey design: the use of closed-ended questions confining responses to categories pre-defined by the experimenter, and basic translations not taking into consideration semantic languages specificities. Our results provide a theoretical grounding and methodological guidelines for designing questionnaires for cross-cultural evaluation of soundscapes.

8:30

4aAA3. Impacts on soundscape appreciation by focussing on sources. Andre Fiebig (HEAD acoustcis GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

Based on COST, the European Cooperation in Science and Technology as an instrument supporting cooperation among scientists and researchers across Europe, collaborations among soundscape researchers was and is still encouraged (COST action TD 0804). In this context, researchers from 18 COST countries and 7 partners outside Europe come together and work on terminology, collection and documentation of soundscapes, harmonization of methodologies and indicators, and design of soundscapes. One scientific issue deals with sound source constellations in soundscapes and evoked attention-attracting processes. Humans can easily focus on a certain source suppressing the noise of other sources. This process obviously influences the general appreciation of the whole soundscape. In different tests this phenomenon was investigated and its potential analyzed with respect to soundscape design. The process of international and interdisciplinary cooperation within the COST framework will be shortly described and benefits and limitations discussed. Moreover, different case studies will be presented to show the effect of source attention and its impact on soundscape evaluation.

8:50

4aAA4. Understanding soundscape as a specific environmental experience: Highlighting the importance of context relevance. Itziar Aspuru (Environ. Unit, Tecnalia Res. & Innovation, Parque Tecnológico Bizkaia, Derio, 48106, Spain, itziar.aspuru@tecnalia.com)

A proposed general conceptual model about environmental experience is presented to guide the soundscape studies. This proposal is the output of a review of the literature on soundscape and our experience regarding psychosocial studies in Tecnalia focuses on the relationship between environment and persons or communities. The general model has been structured around five main elements: person (community), place, activity, previous interaction between person and place, and environmental experience. The environmental experience is a holistic experience within that soundscape and is linked to other perceptions, such as landscape, odour, etc. The relevant factors and variables of each main element have been identified to help us to explain human and social holistic experience in relation to place, in general, and soundscape (perception) in particular. The main objective of this paper is to present this model and to discuss it with interested colleagues.

9:10

4aAA5. Defining the scope of soundscape studies and design methods. Gary W. Siebein (School of Architecture Univ. of Florida Gainesville, FL 32611)

One important goal of soundscape studies is to develop methods that architects, urban designers, planners, landscape architects, and others involved with the design of the indoor and outdoor physical environment can be implement as part of their work. As such the role of design is an inherent aspect of soundscape studies. Design of the physical environment, especially its sonic qualities, is similar to the composing and orchestrating of a piece of music. This paper explores the translation of soundscape theories into physical design concepts through a review of the seminal literature of Shafer and Truax and the systems ecology models of Odum and Brown. The use of physical design elements to create sonic niches for specific acoustic events to occur within that are defined in location and time is used as a technique to develop local sonic attributes within soundscapes. Advanced modeling and simulation techniques; long and short term acoustical measurements that reflect the way sounds are heard by people; interviews, focus group discussions, and questionnaires given to participants in the soundscape to explore the ways people understand sounds in their context; and transformations of the data into physical acoustical interventions are important parts of the process.

9:30

4aAA6. Architectures of sound: A new material in socio-spatial formation. Lorenzo Beretta (BIAD Res., Birmingham City Univ., Gosta Green, Corp. St, Birmingham, UK B4 7DX, Lorenzo.Beretta@bcu.ac.uk)

In architecture, sound is only haphazardly used as an accessory to pre-defined, pre-constituted architectural blocks. The paper examines those sonic properties that are able to alter cognition, behavior, and human interaction leading toward the definition of sound as a material. Over the centuries, buildings have changed their constituent materials ranging from ceramic bricks to concrete, from wood to glass, from light to vapor. Sound is not just a floating, ephemeral presence that exists over time but a solid material that defines and changes the space around us. As a result, spaces are not just lived through their solid elements but through their transposition in the mind of each user. It is for this that by acting on the properties of sound, it is possible to redefine the way people navigate and inhabit spaces. Sound should no longer be used as an accessory or content of architectural spaces but, instead, as a material able to give form, volume, and shape to what is generally referred to as architecture—an inhabitable entity of social relevance in the urban context.

9:50

4aAA7. The use of musical composition and spatial analysis to document and design soundscapes. Joshua C Fisher (6879 Pentland Way #63, Fort Myers, FL 33966, jfisher@jocofi.com)

A series of studies were conducted that used augmented environmental noise analysis techniques. Impulse response analysis and digital processing techniques were applied to both previously recorded sound samples and binaural recordings found on a target site. To provide a representation of the character of a given space, recordings of individual sounds on the site were used to construct digitally composed auralizations of the objects and spaces that were representative of the target space over time. By treating the spatial environment and the individual sounds as separate elements, this technique allowed existing soundscape studies to be used as base models for new soundscapes. This technique was implemented in a number of studies in various natural and manmade locations, all of which demonstrated that sound could be used as part of an integrated design process that allows the designer to treat sound as a material rather than as an unwanted substance (noise).

10:10–10:20 Break

10:20

4aAA8. Soundscape study methods of band rehearsal room. Lucky Tsaih (School of Architecture, Univ. of Florida, P.O. Box 115702, Gainesville, FL 32601, akustx@yahoo.com)

Soundscape study methods were applied to a band rehearsal room to understand the acoustical issues involved in this situation. A taxonomy of sound sources, direct observation to identify the specific sonic events and the multiple source and receiver paths involved in the complex listening and performing tasks that occur during band rehearsals were identified. It was found that students spend almost 1/2 of their rehearsal time involved with verbal instruction and discussion. Interviews with conductors and questionnaires administered to music students in three different band rooms were used to determine what musicians are listening for during rehearsals. Source and

receiver combinations for physical acoustical measurements were located to study the multiple listening tasks identified in the questionnaires. Musicians were constantly trying to "hear each other" for intonation, rhythm, dynamics, articulation, and tone quality during rehearsals. Statistical models linking the qualitative results of the questionnaires with the acoustical measurements and architectural features of the rooms show that the ceiling height, room volume, area of sound diffusing surfaces, low frequency sound level, early reflected sound energy, and reverberation were related to the ability of musicians to hear each other and the detailed attributes of music.

10:40

4aAA9. Application of soundscape method for a worship space. Sang Bum Park (6029 SW 85th St., Gainesville, FL 32608, sbpark04@gmail.com)

Soundscape methods have been used to investigate the sound quality of natural and urban environments and the effects of sonic events on people. Soundscape methods can also be used to study the dynamic listening, speaking, and musical acoustical environment of a worship space. Observations of services in three worship spaces led to identification of a taxonomy of sounds that occur in the rooms as well as the multiple participants involved in producing the sounds and listening to them, i.e., the acoustic community. Five groups including ministers, the choir, music director, the congregation, and sound engineers each with their own physical location in the room, communication paths, and values were identified. Sound sources and receivers for physical acoustical measurements and questionnaires were located for each of the groups. Sound systems were used in each space for various reasons. Analysis of physical acoustical measurements and survey results revealed that different groups had different experiences in the worship spaces when the sound system was used and when it was not used.

11:00

4aAA10. A proposed method of data collection and analysis for soundscape-based noise evaluation. Adam D. Bettcher (Univ. of Florida, Architecture Bldg., P.O. Box 115702, Gainesville, FL 32611)

As an information-rich means of describing and accessing a particular aural environment, soundscape analysis requires additional types of information beyond those used for energy-average and statistical methods of noise evaluation. Managing the flow of data from different types of measurements is crucial in providing a meaningful description of the sonic landscape of a site. The proposed method of data collection and processing produced a means of describing the aural environment with quantitative descriptors from analysis of simultaneously recorded sound level measurements and sound recordings. Sound recordings were analyzed to determine the makeup of a soundscape comprised of individual sounds. The sounds were organized into a taxonomy; and analysis of frequency of occurrence, sound levels, and spectra of each sound, duration of each sound, and the rate of change of each sound were conducted to quantitatively describe the elements of the sonic landscape. This method of data analysis was applied to a soundscape study and produced information that exceeded the abilities of sound-energy measurements alone in providing helpful quantitative descriptors for qualitative soundscape analysis.

11:20

4aAA11. Pursuing cooperative solutions to noise management issues arising in U. S. National Parks. Kurt M. Fristrup (Natural Sound and Night Sky Div. Natl. Park Service, 1201 Oakridge Dr. Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

An emerging paradigm in wildlife conservation holds that ecological knowledge is but one of several dimensions that must be addressed to realize successful outcomes. Human factors—history, culture, economics, and mechanisms for decision and implementation—must be taken into account to devise effective solutions. Addressing these factors demands systematic identification and pursuit of partnerships to synthesize a social tool for conservation. In soundscape management, there has been substantial discussion of metrics for measuring noise exposure and how these measures of noise exposure translate into functional consequences for humans. These challenges, which are substantial, may be the least formidable obstacles to constructive change. The recent history of noise management efforts in the National Park Service illustrate the necessity of cultivating partnerships in many pursuits: education and outreach, research and evaluation, informing policy decisions, and implementation of management plans.

11:40

4aAA12. Soundscape collaboration for science, management, and public outreach at a national historic site. Robert C. Maher (Elec. and Comput. Engr., Montana State Univ., 610 Cobleigh Hall, Bozeman, MT 59717-3780, rob.maher@montana.edu) and Christine Ford (Grant-Kohrs Ranch Natl. Historic Site, MT 59722)

Scientists and engineers involved in soundscape research at national parks and historic sites have the opportunity to collaborate with park management and public outreach professionals. The technical and signal processing considerations of the acoustician can complement the management and regulatory considerations of the park supervisors and the public awareness and outreach efforts of the professional staff and interpretive rangers. The triad of science, policy, and outreach involves all of the key stakeholders in soundscape assessment and evaluation. Examples of collaborative activity at a U.S. National Historic Site are presented.

Session 4aAB

Animal Bioacoustics: Long-Term Acoustic Monitoring of Animals I

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Simone Baumann-Pickering, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

4aAB1. Acoustic monitoring of fish movements across multiple spatial and temporal scales. Yannis P. Papastamatiou (Florida Museum of Natural History, Univ. of Florida, 280 Dickinson Hall, Gainesville, FL 32611, ypapastamatiou@gmail.com)

Understanding the movements of fishes is important for understanding ecological interactions, conservation, and fisheries management. One of the key techniques used to quantify fish movements is acoustic telemetry. Fish can either be actively tracked (where the fish is continuously followed) or be remotely monitored using an array of underwater listening stations (passive telemetry). Ultimately, an understanding of fish movements is required over multiple spatial and temporal scales, which requires the use of several tools. The specific tools used will also vary based on the species being studied, which can range from small reef fishes to pelagic sharks. The use of telemetry in fish movement studies will be reviewed and examples of what sort of questions may be answered using these tools will be given. Future advances in both the tools utilized and the analytical techniques used to interpret data will also be discussed.

8:25

4aAB2. Comparing data collection and processing options for terrestrial acoustical monitoring addressing long durations and large spatial scales. Kurt M. Fristrup (Natural Sound and Night Sky Div. Natl. Park Service, 1201 Oakridge Dr. Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

For generations, skilled naturalists have listened attentively to expand the scope of their field searches and surveys. The rapid proliferation of digital audio recorders with large storage capacity and low power consumption offers numerous options to pursue standardized acoustical surveys of several weeks duration across large areas. This presentation will assess a representative sample of available equipment in relation to the parameters determining their suitability for these applications: price, capacity, fidelity, and ancillary features. Enormous data collection capacity is not useful unless there are adequate tools for processing the data to render the necessary results. The features and performance of several software packages will be compared in the context of different classes of potential application. In particular, the merits of highly selective detectors will be discussed in relation to the alternative of more permissive screening for potentially interesting sounds followed by a measurement and clustering or classification process.

Contributed Papers

8:45

4aAB3. Long-term mapping of red grouper sound production on the West Florida Shelf. Carrie C. Wall (College of Marine Sci. Univ. of South Florida, 140 7th Ave. S., St., Petersburg, FL 33701, cwalll@mail.usf.edu), Michael Lindemuth (Ctr. for Ocean Tech., Univ. of South Florida, 830 1st St. S., St. Petersburg, FL, 33701), Peter Simard, and David A. Mann (College of Marine Sci. Univ. of South Florida, 140 7th Ave S., St. Petersburg, FL 33701)

While it is widely known that numerous fish species produce sound, discerning when and where is more challenging. Through the use of autonomous passive acoustic technology, the spatial and temporal patterns of fish sound production, namely red grouper Epinephelus morio, in the eastern Gulf of Mexico were documented. Two methods have been employed off west-central Florida: moored passive acoustic arrays deployed in 2008 and 2009 covering over 16 600 km^2 from the coast to 100 m deep, and autonomous gliders with integrated hydrophones deployed cross-shelf for up to 4 weeks. Over four million acoustic files generated from these methods were analyzed using DSGLab, an open-source database and data analysis system implemented using MATLAB and MYSQL. An automatic detection algorithm was created and implemented in DSGLab to determine the presence of red grouper calls. False detections were removed manually and the results were analyzed to determine diel and seasonal variability of red grouper sound production in addition to identifying the range of red grouper in the eastern Gulf of Mexico. Support was provided by the University of South Florida, Center for Ocean Technology glider staff, and the captains and crew of the R/Vs Weatherbird II, FishHawk, Eugenie Clark, and Allicat, and the M/V Narcosis. This research was funded by NOPP (OCE-0741705) awarded to DM and the USF/USGS Graduate Assistantship awarded to CW.

9:00

4aAB4. Passive acoustic fish location with a 3-D fixed array. Rodney Rountree, Yegor Sinelnikov, and Alexander Sutin (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

There is a strong need for improved sound source localization software for use by scientist interested in conducting passive acoustic surveys of marine and aquatic habitats where most biological sounds are currently unidentified. Fish sounds are typically low frequency (50–1200 Hz), narrow band, knock trains, short duration grunts, or tones, and can be repeating or irregular. These characteristics together with typically fuzzy signal onset, noisy environmental conditions, and shallow depths often make source localization challenging. We used small 3-D arrays of six hydrophones placed in a fixed-frame square diamond configuration to collect known and unidentified fish sounds for testing from a variety of shallow marine and freshwater habitats. The various methods of sound source localization based on measurements of Differences in Time of Arrival (DTOA) at various hydrophones were investigated, including cross-correlation, first pulse arrival, and phase measurement of tonal components in the fish signal. The most accurate DTOA measurements were obtained using cross-correlation with the phase transform (PHAT) methods. The various algorithms of source localization based on DTOA were considered and MATLAB program for these algorithms were developed. Our goal is to develop publically available software realizing the optimal method of DTOA measurements and source localization.

9:15

4aAB5. Ocean conditions and occurrence of cetacean species near Pt. Sur, California, in 2008–2010. Tetyana Margolina, Alison K. Stimpert, John E. Joseph, Curtis A. Collins, and Christopher W. Miller (Dept. of Oceanogr., Naval Postgrad. School, 1 University Cir., Monterey, CA 93943)

The central California region of the California Current system is characterized by rich marine life and highly variable physical and biological ocean conditions. Here, patterns of marine mammal occurrence near Point Sur are analyzed and linked to seasonal and larger scale variability of the California Current system. A summary of marine mammal vocalizations was created by scanning passive acoustic recordings in the 10–100 kHz frequency band acquired with High-frequency Acoustic Recording Packages (HARPs) deployed at depth of approximately 840 m from August 2008–November 2010. Calls of detected baleen and toothed whales were presented as occurrence time diagrams. To characterize the ocean conditions of the California Current system during this time period, various datasets were used including current meter data collected at Sur Ridge and satellite-derived information on chlorophyll-a concentration (MODIS), temperature (AVHRR), and sea surface height anomalies (AVISO SSH products). Data series of chlorophyll concentrations were used to examine possible changes in the ocean physical and biological state off central California in response to El Niño/Southern Oscillation processes. Correlations between patterns of cetacean occurrence and variability of oceanographic conditions are analyzed and discussed. [Research supported by US Navy CNO(N45).]

9:30

4aAB6. Marine mammal vocalizations and shipping patterns off the California coast near Sur Ridge. John E. Joseph, Alison K. Stimpert, Tetyana Margolina, and Christopher W. Miller (Dept. of Oceanogr., Naval Postgrad. School, 1 Univ. Circle, Monterey, CA 93943, jejoseph@nps.edu)

Studying the effects of anthropogenic noise on marine life is the focus of many ongoing research efforts, and regional studies can provide useful insight into the broader issues. Waters off the central California coast are well known for the rich occurrence of a variety of marine mammal species. The region also contains important ship routes used by vessels transiting between major US west coast ports. Here, we examine the relationship between marine mammal vocalizations collected near Sur Ridge and local shipping patterns determined from automatic identification system (AIS) reports broadcast by ships passing through the region. Passive acoustic recordings of vocalizations were acquired with a moored high-frequency acoustic recording package (HARP) over the frequency bandwidth 10 Hz–100 kHz. Number of vessels within several different radii from the HARP mooring is correlated with presence/absence data of several baleen whale, beaked whale, and dolphin species over 5-min intervals between January 2009 and November 2010. We also examine whether diel patterns of marine mammal distribution are influenced by diel patterns in ship traffic. These data will be useful for establishing mammal-vessel interaction rates in the Monterey Bay National Marine Sanctuary. [Research supported by US Navy CNO(N45)].

Invited Papers

9:45

4aAB7. Acoustic and thermal monitoring of temperate anuran populations. Rafael Marquez, Diego Llusia (Fonoteca Zoológica, Dep Biodiv y Biol Evol, Museo Nal Ciencias Naturales (CSIC). José Gutiérrez Abascal, 2, 28006, Madrid, Spain rmarquez@mncn.csic.es), and Juan Francisco Beltrán (Univ de Sevilla. Av Reina Mercedes, s/n, 41012, Sevilla, Spain)

The results of multi-year acoustic monitoring using automated recording systems (ARS) of the calling activity of two populations per species of five species of anurans from the Iberian peninsula are reported: two species of tree frogs (Hyla) and three species of midwife toads (Alytes), with populations of each species being located at the thermal extremes (coldest vs. hottest) of their distribution. We also report one instance where ARS systems contributed to the recording of a previously undescribed species call, the Moroccan midwife toad. We report methodological procedures such as calculation of the effective area of the recording station, temperature-specific phenological information, and a comparative analysis of environmental predictors of calling activity. Implications for amphibian monitoring and conservation are discussed.

10:05–10:25 Break

10:25

4aAB8. Long-term fish monitoring in the Southern California Bight. Ana Širović (Scripps Inst Oceanogr., UCSD, 9500 Gilman Dr., MC 0205, La Jolla, CA 92093-0205, asirovic@ucsd.edu), David A. Demer (Southwest Fisheries Sci. Ctr., NOAA Fisheries, La Jolla, CA 92037), Sean M. Wiggins, and John A. Hildebrand (UCSD, La Jolla, CA 92093-0205)

While over 100 fish families produce sounds during behaviors like spawning, aggression, and feeding, passive-acoustic sampling is not used commonly for long-term fish population monitoring. Fish sounds consist mostly of low-frequency pulses of variable duration, number, and repetition rate, but it is often difficult to identify their sources to species. For example, underwater sounds from marine life have been studied in the Southern California Bight (SCB) for over 60 years, but because the sound producing fishes are difficult to locate and identify visually, their sound production remains poorly understood. The spatial and temporal distributions of the likely fish sounds recorded in SCB were analyzed, but the species producing those sounds are generally unknown. Where the species are known, more information is needed on the seasonal and interannual variations of their sound production if the passive-acoustic records are to be used to estimate their abundances and distributions. We show that sound characteristics and diel sound production patterns for some species, like bocaccio (*Sebastes paucispinis*), have not changed for over four decades. More directed studies are needed on the behavioral context of fish sound production in SCB to facilitate the use of passive-acoustic monitoring for long-term studies of fish population dynamics.

Contributed Papers

10:45

4aAB9. Monitoring white seabass spawning sounds using a long-term acoustic recording system. Scott A. Aalbers and Chugey A. Sepulveda (Pfleger Inst. of Environ. Res. (PIER), 315 N. Clementine St., Oceanside, CA 92054, Scott@pier.org)

The white seabass (*Atractoscion nobilis*) is an economically important member of the family Sciaenidae that generates a series of distinct low-frequency sounds during spawning. Long-term acoustic recorders (LARS; Loggerhead Instruments) were moored to the seafloor at three sites along the southern California coastline to monitor white seabass sound production. LARS were programmed to record ambient underwater sounds at a sampling rate of 8820 Hz during periods of peak spawning activity (+/− 1 h sunset) from March through July of 2007–2011. White seabass spawning signals were detected at all three sites and verified through the concurrent collection of gravid individuals. Heightened white seabass sound production was documented during May and June, in conjunction with increasing water temperatures and photoperiod. Detection rates were highly variable between adjacent sites and over consecutive seasons, suggesting that spawning activity and site fidelity is influenced by oceanographic conditions. Although additional work is necessary to determine optimal spawning habitats and environmental conditions, this study confirms the utility of a bioacoustic approach to non-invasively identify white seabass spawning periods and locations.

11:00

4aAB10. Nighttime foraging by deep diving echolocating odtonocetes in the Hawaiian Islands. Whitlow W. L. Au, Giacomo Giorli, Michael Richlen, and Marc O. Lammers (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744)

Ecological acoustic recorders (EARs) were deployed in deep waters at five locations around the island of Kauai and one in waters off Ni'ihau in the main Hawaiian island chain. The EARs were moored to the bottom at depths between 400 and 800 m. The data acquisition sampling rate was 80 kHz and acoustic signals were recorded for 30 s every 5 min to conserve battery power and disk space. The acoustic data were analyzed using a suite of software including the M3R (marine mammal monitoring on navy ranges) algorithm, an energy-ratio-mapping algorithm developed at Oregon State University, TRITON software developed at Scripps Institute of Oceanography, and custom MATLAB programs. A variety of deep diving odontocetes, including pilot whales, Risso's dolphins, sperm whales, spinner and pan-tropical spotted dolphins, and beaked whales were detected at all sites. Foraging activity typically began to increase after dusk, peaked in the middle of the night, and began to decrease toward dawn. Between 75 and 87% of biosonar clicks were detected at night. At present, it is not clear why some of the known deep diving species, such as sperm whales and beaked whales, concentrate their foraging efforts at night.

11:15

4aAB11. Long term remote monitoring of cetacean calls using a solar powered autonomous detector. Douglas Gillespie (Sea Mammal Res. Unit, Scottish Oceans Inst., Univ. of St Andrews, KY16 8LB, Scotland), Andrew Maginnis, and Gordon Hastie (SMRU Ltd. New Technol. Ctr., North Haugh, St. Andrews, KY16 9SR, Scotland)

The use of autonomous underwater recording devices is now well established as a method for long term population monitoring. However, the life time of autonomous recorders is still restricted by battery lifetimes and, particularly when monitoring at high frequencies, by available storage space. We present a system for long term monitoring of cetacean populations using a solar powered system in which real time detection algorithms for multiple species can run concurrently on an embedded processing platform. Power consumption is typically below 3 W, with sample rates of up to 500 kHz; the system is therefore suitable for the detection of all known cetacean calls. Data volumes of detected calls are typically below 1 MB a day, meaning that data can be transmitted ashore in near real time using cell or satellite phone networks. Furthermore, communications are bi-directional allowing sampling and detection parameters to be updated remotely. The combination of solar power, real time processing, and data transmission denotes that deployment lifetimes are limited only by the mechanics of the mooring and the need to remove bio-fouling from hydrophones.

11:30

4aAB12. Automatic detection of vocalizations of the frog *Diasporus hylaeformis* in audio recordings. Arturo Camacho (Esc. de CC. de la Comp. e Inf., Univ. de Costa Rica, P.O. Box 2060, San José, Costa Rica, arturo.camacho@ecci.ucr.ac.cr), Adrián García-Rodríguez, and Federico Bolaños (Esc. de Biología, Univ. de Costa Rica)

A method for the automatic detection of calls of the frog *Diasporus hylaeformis* (Eleutherodactylidae) in audio recordings is proposed. The method uses the loudness, timber, and pitch of the vocalizations to identify the calls of the most prevalent individual in a recording. The first step consists in calculating the loudness of the signal to recognize the sections where the focal individual's vocalizations are. The second step consists in using the timber of the signal to recognize vocalizations. Finally, we use two principles we observed in the sounds produced by this species to discriminate between the calls of the most prevalent individual and other calls: individuals tend to vocalize using an almost constant pitch and different individuals use different pitches. Results show that the method is resistant to background noise (including calls of individuals of the same species), microphone-manipulation-induced noise, and human voice, and also that it adapts well to variations in the microphone level produced during the recording.

Session 4aBA

Biomedical Acoustics: Thrombolysis and Microbubble-Mediated Therapies

Azzdine Y. Ammi, Chair
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Chair's Introduction—7:45

Invited Papers

7:50

4aBA1. Low frequency therapeutic ultrasound causes vasodilation and enhanced tissue perfusion. Robert J. Siegel (Heart Inst., Cedars Sinai Medical Ctr., 8700 Beverly Blvd, Los Angeles, CA 90048)

It has been found that ultrasound has a unique effect on arterial and venous dilation as well as tissue perfusion. Our group found that low frequency ultrasound (20 kHz, 0.1 w/cm^2) results in coronary arterial and coronary venous dilation in dogs. The magnitude of vasodilation is similar to that seen with nitroglycerine (NTG) administration but unlike NTG, USD does not lower blood pressure. Our human studies show that USD induces brachial arterial dilation after 1 min with the vasodilatory effect lasting 20 min. In animals we ligated coronary arteries, stopping epicardial coronary flow, resulting in a drop in myocardial tissue perfusion to 70% of normal; myocardial tissue *p*H fell from 7.43 to 7.05. After 60 min of USD, tissue perfusion improved by 20% and *p*H normalized in spite of persistent coronary artery occlusion. The enhanced perfusion effect of ultrasound was eliminated if an inhibitor of nitric oxide synthase was given prior to ultrasound exposure. Conclusions: low frequency ultrasound causes vasodilation and improves tissue perfusion. These effects appear to be mediated at least in part by the USD enhancing tissue release of nitric oxide.

8:10

4aBA2. Bifrequency excitation for extracorporeal ultrasound thrombolysis. Bruno Gilles, Izella Saletes, Mamdouh Dhahbi, Maher Ben Chiekh, Jean-Christophe Béra (INSERM - U1032, Univ. Claude Bernard Lyon 1, 151 Cours A. Thomas, Lyon, 69003, France, bruno.gilles@inserm.fr), and Rares Salomir (Univ. Hospital of Geneva, Geneva, Switzerland)

A bifrequency excitation consisting of two neighboring frequency components can reduce intensities needed to achieve strong inertial cavitation activities. We present *in-vitro* experimental results aiming at testing such a bifrequency excitation for extracorporeal ultrasound thrombolysis. In a first set of experiments, human blood clots were inserted in small tubes filled with saline and placed at the focus of a piezoelectric transducer. The efficiencies of mono- (550 kHZ) and bifrequency (535 and 565 kHz) excitations were compared for (spta) intensities ranging from 50 to 160 W/cm^2, and a passive recording of the cavitation activity was performed during treatment. A modified setup enabled to measure the size distribution of the debris resulting from thrombolysis experiments realized under flow. A comparison of the spatial temperature distribution for each type of excitation was performed in another set of experiments using MR temperature imaging. Under the conditions of the experiments, 80% of thrombolysis was achieved with a monofrequency intensity of 150 W/cm^2, while 80 W/cm^2 were sufficient with a bifrequency excitation. Mean debris size was reduced by the use of a bifrequency excitation, and MR temperature imaging showed that, for a given intensity, the spatial temperature distributions are the same for both types of excitation.

8:30

4aBA3. Assessing thrombolytic efficacy *in vitro*: Clot mass loss versus fibrinogen protein fragment concentration. Stephen R. Perrin Jr. (Biomedical Eng. Program, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3940, Cincinnati, OH 45267, perrinsr@mail.uc.edu), Gail J. Pyne-Geithman (Univ. of Cincinnati, Cincinnati, OH 45267), Nikolas M. Ivancevich (Siemens Medical Solutions, Issaquah, WA 98029), Shauna L. Beiler, Kenneth R. Wagner (Univ. of Cincinnati, Cincinnati, OH 45267), and Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH 45267)

Ultrasound (US) acts synergistically with recombinant tissue plasminogen activator (rt-PA) to accelerate thrombolysis. A correlation between clot mass loss and production of the fibrin degradation product D-dimer was investigated using an *in vitro* clot thrombolysis model. Fully retracted clots formed from 1.5-ml human whole blood were suspended in plasma at 37°C and treated with rt-PA, or rt-PA, Definity®, and pulsed 120-kHz US for 30 min. Clots in plasma alone served as controls. Thrombolytic efficacy was assessed as percent clot mass loss. Samples from plasma surrounding the clot and from macerated clots were analyzed for D-dimer using an enzyme linked immunosorbent serologic assay. A statistically significant enhancement in clot mass loss was observed for clots exposed to rt-PA, Definity®, and US compared to rt-PA alone. Clots treated with rt-PA exhibited a higher concentration of D-dimer both in the clots and plasma. However, clots treated with rt-PA, Definity®, and US did not exhibit an enhanced level of D-dimer. In future studies, we plan to elucidate the role of erythrocyte liberation in US-enhanced clot mass loss. [This work was supported by NIH RO1 NS047603.]

8:50

4aBA4. Ultrasound-enhanced thrombolysis in porcine clots in a flow system. Azzdine Y. Ammi, Yan Zhao, Aris Xie, Jonathan Lindner (Div. of Cardiovascular Medicine, OHSU, 3181 SW Sam Jackson Park Rd., Portland, OR 97239), Thomas R. Porter (Univ. of Nebraska Medical Ctr., Omaha, NE 68198), and Sanjiv Kaul M.D (Div. of Cardiovascular Medicine, OHSU, Portland, OR 97239)

Ultrasound and ultrasound contrast agent microbubbles (UCAM) are able to mechanically induce clot reduction. The aim of the study was to demonstrate the efficacy of various ultrasound conditions to enhanced thrombolysis in porcine clots inside a flow system. Clots were formed by infusing 4.1 ml of blood in transfer pipets. The pipets contained Dacron grafts to anchor the clot and initiated its formation. A 14-gage needle was placed at the center of the pipet and removed after clot formation to allow flow inside the clot. The clots were treated for 20 min with ultrasound and homemade UCAM at a concentration of 107 microbubbles/ml (flow rate 0.9 ml/min). Thrombolysis was monitored using an ultrasound scanner in pulse inversion mode. Results show that the radiation force causes the microbubbles to be pushed against the inner clot wall and cavitation induced lysis. The portion of the clot closest to the transducer was not affected by the therapy as the microbubbles were pushed in the direction of propagation. Mechanical clot reduction was observed in real-time in a flow system at various acoustic settings.

Contributed Papers

9:10

4aBA5. Quantified lysis of cell-like lipid membranes due to nanoparticle-facilitated cavitation. Michael J. Benchimol, Stuart D. Ibsen (Jacobs School of Eng., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA, 92093, mbenchim@ucsd.edu), Dmitri Simberg (Moores UCSD Cancer Ctr., La Jolla, CA 92093), Zhe Wu, Robert F. Mattrey (Univ. of California, San Diego, La Jolla, CA 92093), and Sadik C. Esener (Univ. of California, San Diego, La Jolla, CA 92093)

Certain nanoparticles act as nucleation sites for acoustic cavitation. Their surface roughness and hydrophobic regions can decrease the pressure required to induce the cavitation event. This concept has been examined for potential to provide a sensitizer for high-intensity focused ultrasound (HIFU), where cavitation can be beneficial. While passive cavitation detection can accurately determine the pressure threshold, variations in acoustic impedance/backscatter from different nanoparticles can make it difficult to normalize the amount of cavitation and determine the precise magnitude of the physical consequences. Here we demonstrate a method to determine the extent of lysis of lipid membranes using liposomes loaded with a self-quenching fluorophore. Lysed liposomes released the fluorophore, causing an increase in the fluorescence signal. Liposomes were mixed with the nanoparticles and insonated with a HIFU transducer at physiological temperature. The pressure threshold for dye release was measured for a panel of nanoparticles. Nearly complete release of the dye was achievable in all cases, but it required higher pressures in the absence of nanoparticles. In addition, we measured the effect of a viscous medium, which is more representative of certain physiological states. Furthermore, the encapsulation of the nanoparticle within a liposome can create a platform delivery vehicle for anti-cancer therapeutics.

9:25

4aBA6. Enhanced viral activity in tumors using focused ultrasound and microbubbles—A long term study. James J. Choi (Inst. of Biomed. Eng., Old Rd. Campus Res. Bldg., Univ. of Oxford, Oxford OX3 7DQ United Kingdom. james.choi@eng.ox.ac.uk), Robert C. Carlisle (Univ. of Oxford, Oxford OX3 7DQ United Kingdom.), and Constantin C. Coussios (Univ. of Oxford, Oxford OX3 7DQ United Kingdom.)

Oncolytic viruses target and kill cancer cells and self-amplify through replication. However, viral therapy *in vivo* is limited by insufficient systemic delivery. Here, focused ultrasound (FUS) was used in conjunction with microbubbles to produce cavitation and enhance tumor viral delivery. Human breast cancer cells (ZR75.1) were injected subcutaneously in mice (n = 8), which grew to a tumor volume of at least 30 mm^3. The tumors were then exposed to FUS (frequency: 0.5 MHz, pulse duration: 50 000 cycles, pulse repetition frequency: 0.5 Hz) for 4 min following injection of 100 μl of SonoVue microbubbles (SVM) with polymer-coated adenovirus encoding luciferase (pc-Ad-Luc). Experiments were repeated for three controls: pc-Ad-luc with neither FUS nor SVMs, FUS and SVM without pc-Ad-Luc, and buffer alone. Acoustic emissions were recorded with a passive detector, which validated the presence of inertial cavitation and ensured good SVM reperfusion kinetics. Tumor viral expression was then imaged using IVIS following luciferin injection. At 1, 2, 3, and 7 days post-injection, 3, 20.3, 30.2, and 22.9-fold increases in photons/s/cm^2 were observed when pc-Ad-Luc was used with FUS and SVM compared to pc-Ad-Luc alone. In conclusion, FUS and SVM enhanced oncolytic virus delivery resulting in amplified viral expression over time.

9:40

4aBA7. Ultrasound-induced temperature elevation for in-vitro controlled release of temperature-sensitive liposomes. Christophoros Mannaris, Eleni Efthymiou (Dept. of Mech. and Manufacturing Eng., Univ. of Cyprus, 75 Kallipoleos St. 1678 Nicosia, Cyprus), Jean-Michel Escoffre, Ayache Bouakaz (UMRS INSERM U930, CNRS ERL3106, Universite Francois Rebelais, Tours, France), Marie-Edith Meyre, Matthieu Germain (NANOBIOTIX, 60, rue de Wattignies bat. B, 75012 Paris), and Michalakis A. Averkiou (Univ. of Cyprus, 1678 Nicosia, Cyprus)

Drug loaded temperature-sensitive liposomes (TSLs) release their payload with mild hyperthermia near their phase transition temperature ($T_m = 43–45\,^{\circ}C$). Such a release may improve therapeutic efficacy and reduce toxic side effects in cancer treatment. In the present work, two different approaches are considered where focused ultrasound is used to induce the required temperature elevation for the release of doxorubicin from TSLs: (a) primary heating due to thermo-viscous absorption of ultrasound in absorptive media (oil, glycerol) and (b) secondary heating in non-absorptive media (blood, cell medium) due to heat transfer from the surroundings. Fine-wire thermocouple readings where in close agreement with theoretical predictions of temperature elevation with the Bioheat equation. Pulsing schemes to elevate and maintain the temperature at the desired value were designed with the Bioheat equation and validated with experiments. Fluorescence spectroscopy was used to assess the release of free doxorubicin that exhibits higher fluorescence intensity than the liposomal formulation. Significant drug release was achieved with both approaches.

9:55

4aBA8. Dynamical-systems measures of ultrasound contrast agent proximity to target walls. Fatimah Dzaharudin (Dept. Mech. Eng., Univ. of Melbourne, Melbourne, VIC 3010, Australia, f.dzaharudin@student.unimelb.edu), Sergey A. Suslov (Swinburne Univ. of Technol., Hawthorn, VIC 3122, Melbourne, Australia), Andrew Ooi (Univ. of Melbourne, Melbourne, VIC 3010, Australia), and Richard Manasseh (Swinburne Univ. of Technol., Hawthorn, VIC 3122, Melbourne, Australia)

Targeted ultrasound contrast agents are microbubbles that strongly scatter ultrasound, providing contrast on a scan, and have also been coated in molecules that adhere to target pathologies. The ultimate aim is to identify diseased tissue in clinical ultrasound practice. One issue is to discriminate in real time between microbubbles that have adhered to their target pathologies on blood-vessel walls, from those that are freely flowing in the bloodstream. It is known that linear theory predicts a shift in resonant frequency owing to the presence of a wall. Weakly nonlinear theoretical results are presented on alterations to the dynamical-systems behavior of one or more microbubbles on and near to walls. In particular, the bifurcation diagram is altered as

microbubbles approach a wall. Near a wall, period-doubling and period-quadrupling bifurcations and transitions to broadband chaos occur at altered values of the incident pressure amplitude. Alterations in the bifurcation diagram increase as multiple bubbles are held fixed close to each other and to the wall. This suggests that filtering of the returning echoes around selected solution branches could provide a further real-time indicator of locations where targeted ultrasound contrast agents have adhered.

10:10–10:25 Break

10:25

4aBA9. Spatio-temporal mapping and characterization of acoustic cavitation seeded by microbubbles and solid microparticles during focused ultrasound exposure. James J. Choi and Constantin C. Coussios (Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Univ. of Oxford, OX3 7DQ, United Kingdom, james.choi@eng.ox.ac.uk)

Cavitation nuclei are often used to seed and promote acoustic cavitation in therapeutic applications. However, the effects of fluid velocity and ultrasound exposure parameters peak rarefactional pressure (Pr), pulse duration (PD), and pulse repetition frequency (PRF) on the spatial distribution, type, magnitude, and number of acoustic cavitation events for each nuclei remains poorly understood. In this study, a 1.6-mm diameter tunnel phantom (3% agar) was perfused (fluid velocity: 10–40 mm/s) with either microbubbles (SonoVue) or hydrophobic solid microparticles (TALC) and exposed to 74 different acoustic parameter combinations (frequency: 0.5 MHz, Pr: 150–1500 kPa, PD: 1–100,000 cycles, PRF: 1–50 Hz, number of pulses: 10–250). Spatial mapping of passively acquired acoustic cavitation emissions was performed with a 64-element array coaxial to the focused ultrasound transducer. At pressures above the cavitation threshold, cavitation activity generated from microbubbles was significantly reduced and spatially biased upstream after the first pulse at high PRFs relative to the fluid velocity. On the other hand, solid microparticles had no spatial bias and no significant reduction in the energy of acoustic emissions after the first pulse. Whereas microbubbles may be destroyed, and therefore, cease to act as cavitation nuclei, solid microparticles do not suffer from depletion of energy with high PRFs.

10:40

4aBA10. *In vitro* acoustic characterization of a novel poly-lactic acid polymer shelled contrast agents. Shirshendu Paul, Daniel Russakow, Tyler Rodger, Kausik Sarkar (Dept. Mech. Eng., Univ. of Delaware, Newark, DE 19701), and Margaret Wheatley (Drexel Univ., Philadelphia, PA 19104)

Micron sized encapsulated gas bubbles have been extensively studied as contrast enhancing agents for ultrasound imaging. This study will report on *in vitro* acoustic characterization of a novel poly-lactic acid (PLA) shelled contrast agent. PLA is a bio-degradable polymer approved by the FDA to be used in drug delivery applications. Thus, PLA shelled contrast bubbles have the potential of being developed as the next generation contrast agents. Both attenuation and scattering measurements will be reported. Attenuation measurements are obtained using three different transducers (central frequencies 2.25, 3.5, and 5 MHz). Pressure dependent scattered response is obtained for two different excitation frequencies of 2.25 and 3.5 MHz. Results indicate excellent scattering properties of the PLA shelled bubbles. The strongly non-linear nature of scattered response makes PLA bubbles a potential choice for harmonic and sub-harmonic contrast imaging applications.

10:55

4aBA11. Experimental characterization of dye-loaded echogenic liposomes. Shirshendu Paul, Daniel Russakow, Tyler Rodger, Kausik Sarkar (Mech. Eng., Univ. of Delaware, Newark, DE 19716), Rahul Nahire, and Sanku Mallik (Pharmaceutical Sci., North Dakota State Univ., Fargo, ND 58108)

Liposomes are submicron sized vesicles with a lipid bilayer encapsulating an aqueous phase inside. Due to their favorable properties like longer circulation time, lesser toxicity, and greater uptake, they are a prime candidate for drug delivery. Recently, they are being specially prepared so as to encapsulate air, making them good scatterers of ultrasound wave. These echogenic liposomes, therefore, can be used both for ultrasound contrast imaging and drug delivery. We will report *in vitro* attenuation and scattering measurement from echogenic liposomes loaded with carboxyfluorescein (used as a surrogate for small molecular weight drugs). The results will be compared with non-dye-loaded ones. Effects of dye loading and presence of bovine serum albumin (BSA) on size distribution, echogenicity, and release characteristics will also be discussed.

11:10

4aBA12. Localized activation and cellular effects of ultrasound triggered drug delivery vehicles with encapsulated microbubbles. Stuart Ibsen (Dept. of Bioengineering, Moores Cancer Ctr., Univ. of California San Diego, 3855 Health Sci. Dr. # 0815, La Jolla, CA 92093-0815), Michael Benchimol, Dmitri Simberg (Univ. of California San Diego, La Jolla, CA 92093), Carolyn Schutt (Univ. of California San Diego, La Jolla, CA 92093-0815), Jason Steiner (Univ. of California San Diego, La Jolla, CA 92093), and Sadik Esener (Univ. of California at San Diego, La Jolla, CA 92093)

The harmful side effects of chemotherapy originate from indiscriminate exposure of healthy tissue to the drugs. The goal of targeted drug delivery is to reduce these side effects by encapsulating concentrated drug in a vehicle which releases it only in the tumor region. Low intensity focused ultrasound can be used as a trigger to specifically activate these vehicles by highlighting only tumor tissue, creating a stark differentiation with healthy tissue. A new injectable drug delivery vehicle has been developed with a stabilized nested lipid shell geometry that encapsulates a high capacity chemotherapy payload, and a stabilized microbubble into one structure. Ultrasound affects the microbubble only in the small focal volume, creating a localized shockwave which ruptures the vehicle's outer membrane triggering pinpoint release in tissue phantoms. These shockwaves, and their interactions with the delivery vehicle membranes and live cells, have been documented for the first time using a custom system which combines high-speed videography and fluorescent microscopy with focused ultrasound. Vehicles which do not pass through the tumor will be excreted through normal processes. This externally-activated scheme could lead to truly tumor-specific drug delivery. [NCI Grant No. 5U54CA119335-05, and UCSD Cancer Center Specialized Support Grant No. P30 CA23100 supported this work.]

11:25

4aBA13. Role of effective surface tension on the frequency dependent subharmonic threshold for contrast microbubbles. Amit Katiyar and Kausik Sarkar (Mech. Eng., Univ. of Delaware, Newark, DE 19716)

We numerically investigate the predictions from several contrast microbubble models to determine the excitation threshold for subharmonic generation. In contrast to the classical perturbative result, the minimum threshold for subharmonic generation is not always obtained near twice the resonance frequency; instead it can occur over a range of frequency from resonance to twice the resonance frequency. The quantitative variation of the threshold with frequency depends on the model, bubble radius, and encapsulation properties. All models are transformed into a common interfacial rheological form, where encapsulation is represented by two radius dependent surface properties—effective surface tension and surface dilatational viscosity. Variation of the effective surface tension with radius, specifically having an upper limit (resulting from strain softening or rupture of the encapsulation during expansion), plays a critical role. It destroys a sharp minimum at twice the resonance frequency. Without the upper limit on effective interfacial tension, the threshold is extremely large especially near the resonance frequency.

11:40

4aBA14. Three-dimensional dynamical equations of interacting bubbles. Eru Kurihara (Dept. of Eng., Oita Univ., Dannoharu, Oita City 870-1192, Japan, kurihara@oita-u.ac.jp)

It is known that bubble cavitation plays important role in kidney stone fragmentation in the shock wave lithotripsy and other medical applications of shock wave. The behavior of such bubbles, however, considerably complicated because of its nonlinearity and mutual interactions among the bubbles. For weakly nonlinear oscillations, dynamics of interacting bubble can be approximately expressed by the method of multi-pole expansion with

spherical harmonics. In the previous study, the author derived a set of dynamical equations of two interacting aspherical bubbles with Lagrangian mechanics. The axisymmetrical system of two interacting bubbles can be described in two-dimensional coordinate system, and then shape oscillation of the bubbles is expressed with Legendre polynomials. The bubble behavior described by the derived equations qualitatively agreed with experimental results by high-speed photographs. In the dynamics of three or more bubbles, however, the behavior of bubbles is essentially three dimensional, and thus the system of these bubbles should be represented by three-dimensional spherical harmonics (associated Legendre functions). In this study dynamical equations for three interacting aspherical bubbles are derived by multi-pole expansion in the framework of Lagrangian formalism. [Work supported by Grants-in-Aid for Scientific Research 23760142 and a research grant from The Mazda Foundation.]

THURSDAY MORNING, 3 NOVEMBER 2011 PACIFIC SALON 6/7, 8:30 TO 9:45 A.M.

Session 4aEAa

Engineering Acoustics: Energy Harvesting

Stephen C. Thompson, Chair
Applied Research Lab., Pennsylvania State Univ., P.O. Box 30, State College, PA 16804-0030

Contributed Papers

8:30

4aEAa1. Energy harvesting of tonal sound excited by heat addition and vortex shedding. Sungmin Jung, Rafael Hernandez, and Konstantin I. Matveev (MME School, Washington State Univ., Pullman, WA 99164, matveev@wsu.edu)

Tonal sound may appear inside resonating duct systems due to heat addition or vortex shedding in the presence of mean flow. Amplitudes of this sound can reach significant levels. The sound power can be captured and converted to electricity using electroacoustic transducers. Two pipe setups were constructed to demonstrate energy harvesting of tonal sound using piezoelements. In the first system, the sound was excited by vortex shedding and impingement on baffles in the presence of mean flow. The second system represented closed and open types of a standing-wave thermoacoustic engine. Electric power in excess of 0.5 mW was captured and released on passive electric loads from the tonal sound. This power level is sufficient for some low-power sensors. Further system optimization can significantly increase the amount of harvested energy. [Work supported by the NSF Grant 0853171.]

8:45

4aEAa2. Electromechanical transduction system design for optimal energy harvesting from ocean waves. Amadou G. Thiam and Allan D. Pierce (Dept. Mech. Eng., Boston Univ., Boston, MA 02215, thiam@bu.edu)

While details of the currently most highly publicized devices for conversion of ocean wave energy to electrical energy are generally not disclosed in the open literature, the authors believe that, for devices not on the coastline, the common transduction mechanism involves electromagnetic induction with conducting wires moving relative to permanent magnets. A general discussion is given of how such a mechanism can be used in this application. The overall analysis of the mechanical system with lumped or distributed masses and elastic elements driven by buoyancy forces associated with incident ocean waves is facilitated, if the transduction system is modeled as linear mechanical dashpots, and the procedures for deriving effective dashpot constants are described. The mechanical analysis suggests that, for waves in a general frequency range, there is an optimal choice for the parameters of the mechanical system, so that the maximum electrical power can be harvested. The optimal energy extracted per wave cycle is invariably much less than the total mechanical energy of the oscillating components of the system. A distinction is made between freely floating systems and systems anchored to the ocean bottom and between systems which are driven near a resonant frequency and those driven substantially below resonance.

9:00

4aEAa3. Energy conversion through thermoacoustics and piezoelectricity. Robert M. Keolian (Appl. Res. Lab, Penn State Univ., P.O. Box 30, State College, PA 16804-0030, keolian@psu.edu) and Scott Backhaus (Condensed Matter and Magnet Sci. Group, Los Alamos Natl. Lab., Los Alamos, NM 87545)

Waste or prime heat can be converted into electricity with thermoacoustic-Stirling engines coupled to piezoelectric alternators. An inline arrangement of engines and alternators allows a vibration balanced, multiphase power generator that is compact, light weight and low cost. The engines convert heat into high amplitude ≈400 Hz oscillations in pressurized helium gas. These pressure oscillations cause a thin steel diaphragm to flex like a drumhead. The diaphragm is supported at its perimeter by a ring of piezoelectric elements. As the diaphragm flexes in either direction, it pulls inward on the piezoelectric elements causing a large amplified ≈800 Hz fluctuating compressive stress in the elements which then convert the stress into electricity with high efficiency. The flexible-diaphragm piezoelectric alternator overcomes the large acoustic impedance mismatch between the helium and piezoelectric elements without exceeding the limited fatigue strength of available materials. So far, a prototype generator has produced 37 W, and is being modified to produce 600 W. Also, a project is underway to recover 7 kW peak electrical power from the exhaust of an over-the-road heavy-duty diesel truck. The generator appears scalable up to megawatt power levels. [Work supported by DOE, ONR, Clean Power Resources, Innovation Works, and Applied Research Laboratory.]

9:15

4aEAa4. Modifying a balanced armature speaker for energy harvesting applications. Nikolas T. Vitt and Stephen C. Thompson (Appl. Research Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804)

Balanced armature transducers are produced in large quantity for use as miniature speakers in hearing aids and in-ear headsets. These devices are reciprocal and might be used as the generator in vibration energy harvesting. However, previous work has shown that, as manufactured, the speakers do not have sufficient vibration sensitivity to be directly used in this way. This

paper explores a set of design modifications to increase the sensitivity to a useful level. Preference is given to modifications that require minimum investment in new manufacturing tooling.

9:30

4aEAa5. Stepped-plate transducer as an energy transmitter. Yonghwan Hwang, Yub Je (Dept. of Mech. Eng., POSTECH, Pohang, South Korea, serenius@postech.ac.kr), SungQ Lee, Gunn Hwang (ETRI, Daejeon, South Korea, Hermann@etri.re.kr), and Wonkyu Moon (POSTECH, Pohang, South Korea)

Power transmission through acoustic energy may be useful in such cases as supplying power to wireless small sensors. In the case, the radiation and reception power efficiency is important. Even in ultrasonic frequency bands, the power efficiency of most acoustic radiators in air is not high enough. The stepped-plate ultrasonic transducers, introduced by Gallego-Juarez *et al.* [Ultrasonics **16**, 267–271(1978)], may be a good candidate for the radiator for this purpose because it can effectively generate highly directive, large-amplitude, ultrasonic sounds in air. The transducer consists of langevin transducer that causes wave generation, mechanical amplifier, and stepped radiation plate. Although it is reported to achieve 80% of power efficiency, it is not reported how to achieve maximum power efficiency. For design of large-amplitude, high-efficiency stepped-plate transducer, the design of not only individual parts but also system integration of entire transducer is important. In this research, we developed an analytical model for the whole transducer by combining continuum models of each parts and found proper design parameters for the radiation power and the power transmitting efficiencies. Then, we seek the optimal design for maximizing power efficiency through parametric analyses, and the results are confirmed through finite element method analysis. [Work supported by ETRI (South Korea).]

THURSDAY MORNING, 3 NOVEMBER 2011 PACIFIC SALON 6/7, 10:00 A.M. TO 12:00 NOON

Session 4aEAb

Engineering Acoustics and Underwater Acoustics: Vector Sensors, Projectors, and Receivers I: Projectors and Reversible Transducers

Stephen C. Butler, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02840

Roger T. Richards, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02840

Invited Papers

10:00

4aEAb1. The octoid modal vector projector. Alexander L. Butler and John L. Butler (Image Acoust., Inc., 97 Elm St., Cohasst, MA 02025, abutler@imageacoustics.com)

The eight piston octoid transducer is a descendent of the three piston trioid and four piston astroid transducers. These transducers were developed for low frequency underwater sound applications where wide bandwidth and small volume is desirable. Magnified piston motion is achieved through attached leveraged shell action driven by radial piezoelectric stacks. This condition yields magnified piston displacement and lowered Qm through magnified loading on the piezoelectric stacks. The new octoid design concept was implemented on an eight stack power wheel modal transducer. And this new design has been used to significantly reduce the outer diameter for the same response or, alternatively, yield lower frequency response for the same diameter. Finite element model and measured results are presented for the power wheel with and without the octoid leveraging and the octoid transducer is also compared to the previous trioid and astroid transducers.

10:20

4aEAb2. A compact Terfenol-D vector projector. Julie C. Slaughter (Etrema Products, Inc., 2500 North Loop Dr. Ames, IA 50010, julie.slaughter@etrema.com)

In active sonar applications, it is desirable to have a directional, steerable sound source to accelerate the target localization process. A narrow cardioid beam pattern with approximately 78° beam width and rear side lobes 25 dB down can be generated by combining monopole, dipole, and quadrupole beam patterns with appropriate amplitudes and phases. A cylindrical array composed of eight Terfenol-D Tonpilz transducers arranged in two rings with each transducer pointing radially outward from a common center mass has been developed to generate narrow cardioid beam patterns. Focus during the design process was on minimizing size, maximizing the bandwidth, and maximizing the duty cycle. The diameter of the source is 0.19 wavelengths at the lowest operating frequency and 0.76 wavelengths at the highest operating frequency. Bandwidth of the source is greater than two octaves. Lumped parameter modeling and finite element modeling are used to demonstrate the monopole, dipole, quadrupole, and narrow cardioid beam patterns. The effects of extraneous vibration modes on the beam patterns and sound output are discussed. Maximum continuous wave sound output and duty cycle at maximum output are estimated from thermal models and test data from a single Tonpilz element. [Work supported by the Office of Naval Research.]

10:40

4aEAb3. High power, broad bandwidth, compact, single crystal vector projector. P. David Baird, James M. Powers, and Ivan A. Rodriquez (Progeny Systems, 2401 South 1070 West, St. 16A, West Valley City, UT 84119)

An approach to producing a high power, broad bandwidth, compact transducer with steerable super-directionality will be described. The design, utilizing high coupling coefficient PMN-PT single crystal material, will be compared to conventional transducers utilizing PZT ceramic. Performance predictions for transmit response, impedance, efficiency, power, volt-amperes, source level, and bandwidth will be provided.

Contributed Papers

11:00

4aEAb4. Unidirectional multimode piezoelectric spherical transducers. David A. Brown, Boris Aronov, and Corey Bachand (BTech Acoust. LLC, Adv. Tech. & Manuf. Ctr., Univ. of Massachusetts, 151 Martine St., Falls River, MA 02723)

Analytical and experimental results are presented for unidirectional broadband multimode piezoelectric acoustic transducers utilizing axisymmetric vibrations of spherical transducers. The analysis covers the acoustic radiation and reception by including the acoustic impedance's and diffraction coefficients for transducers with conformal baffles. The energy method is used to obtain equivalent parameters for a multi-contour electromechanical circuit representation of the transducer and to calculate performance. Experimental data obtained are in good agreement with analytical results.

11:15

4aEAb5. Compact cylindrical transducer arrays for directional communications and navigation. David Brown and Boris Aronov (BTech Acoust. LLC, Adv. Tech. & Manuf. Ctr., Univ. of Massachusetts, 151 Martine St., Falls River, MA 02723)

Designs and experimental results are presented for compact self-baffled cylindrical arrays comprising piezoelectric transducers suitable for telemetry stations and small vehicle directional communications or navigation. Designs with cylindrical transducers, rod/bar transducers, and tonpilz elements are compared with single transducers using turnable baffles to achieve unidirectional beams that may be steered in azimuthal plane.

11:30

4aEAb6. Improved spiral-wavefront transducer for underwater acoustic navigation. David Brown, Boris Aronov, and Corey Bachand (BTech Acoust. LLC, Adv. Tech. & Manuf. Ctr., Univ. of Massachusetts, 151 Martine St., Falls River, MA 02723)

Spiral-wavefront transducers have received recent attention as enabling elements in phase-based underwater navigation systems, whereby the signaling transducer launches a diverging wave that is omnidirectional by magnitude but linearly phase-biased with azimuthal angle. The detected signal is compared with a control signal having constant phase to allow the unique determination of bearing angle. Range can be determined by time-of-flight. While such systems are used in air traffic control navigation, the transition to underwater acoustics had only emerged with the advent of increased unmanned underwater vehicle traffic (Dzikowicz and Hefner). Analytical and experimental test results, including beam patterns, TVR, and power factor, are presented for resonance operations at 25 kHz (c-band). [Work supported by BTech Acoustics LLC.]

11:45

4aEAb7. Design of a wideband multimode tonpilz transducer with a nonuniform piezoelectric layer stack. Yongrae Roh and Saosometh Chhith (School of Mech. Eng., Kyungpook Natl. Univ., 1370 Sankyukdong, Bukgu, Daegu 702-701, Korea, yryong@knu.ac.kr)

It has been well-known that a multimode transducer could provide a wider frequency bandwidth than a single mode one, and conventionally a multimode transducer can be achieved by designing the geometry of its head mass, increasing the head mass radius and reducing the head mass thickness. However, a very large head mass can cause some drawbacks to transducer performance when used in an array, i.e., low source level and big crosstalk with neighboring transducer elements. In this work, a new and very simple design method has been developed to widen the bandwidth of a Tonpilz transducer, which is replacing the uniform PZT layer stack by a nonuniform one. A piezoelectric stack composed of nonuniform PZT layers can generate higher mechanical energy than that composed of uniform layers for the same input electrical energy, which means a higher coupling coefficient thus a wider bandwidth. The effects of the nonuniformity of PZT layer thicknesses on a multimode Tonpilz transducer performance were investigated through finite element analyses. Then, the functional forms of the performance were derived in relation to the nonuniform PZT thicknesses and were inserted into a genetic algorithm to achieve the widest possible bandwidth of the Tonpilz transducer.

Session 4aMUa

Musical Acoustics: Physical Models For Sound Synthesis II

Edgar J. Berdahl, Chair
Dept. of Music, Stanford Univ., Stanford, CA 94305

Contributed Papers

8:00

4aMUa1. Estimating the clarinet mouthpiece reflection from measurement and the instrument's produced sound. Tamara Smyth (School of Computing Sci. Simon Fraser Univ., 50-13450 102nd Ave. Surrey, B.C., Canada V3T 0A3, tamaras@cs.sfu.ca) and Jonathan Abel (CCRMA, Dept. of Music, Stanford Univ., Stanford, CA, abel@ccrma.stanford.edu)

In this work, a method is presented for estimating the reflection off the clarinet mouthpiece, using a priori measurement of the bell, and post processing of the instrument's produced sound. A previously introduced measurement technique is used to obtain measurement of clarinet bell and transmission filters. In addition to these elements, however, the round-trip propagation loss in the clarinet bore and bell also includes wall loss and mouthpiece reflection. Though the former is accurately modeled theoretically, assuming the clarinet bell is close to cylindrical, the mouthpiece is more difficult to measure, both because of a supposed oscillating reed, and because the required placement of a measurement device would obstruct the mouthpiece's characteristic reflection. The lumped round-trip loss filter in the bore is estimated from the clarinet signal by first considering the signal's periodic structure. After taking the signal's autocorrelation, which preserves its periodicity and naturally provides the beginning of the period, the round-trip filter is iteratively estimated by constructing an optimization function from the first and second phases of the autocorrelation sequence. Once the round-trip loss is estimated, the mouthpiece reflection may be extracted by removing the effect of the other known comprising filter elements.

8:15

4aMUa2. Inverse problem in sound synthesis and musical creation using mass-interaction networks. Jérôme Villeneuve and Claude Cadoz (Laboratoire ICA, 46 Ave. Felix Viallet, 38000, GRENOBLE, FRANCE, jerome.villeneuve@imag.fr)

Sound synthesis with mass-interaction physical modeling networks is known as a general paradigm capable of being the central part of complete software environments for both sound synthesis and musical creation. GENESIS 3, resting on the CORDIS-ANIMA formalism and developed by ACROE/ICA Laboratory, is the first environment of this kind. Using it, the artist may be facing an inherent problematic of every creation process: how to use a given tool in order to obtain an expected result. In our context, the question would be: "Considering a sound, which physical model could produce it?" Our work aims at formalizing this inverse problem and therefore at helping the user through his creative process. Therefore, we will consider how he could describe a sound (entry of the inverse problem), how to define a generator model based on mass-interaction physical networks and each one of its subcomponents (formal solution of the inverse problem), and, obviously, how to compute this solution considering an entry (resolution of the inverse problem). In this paper, we will develop each one of those three points and present the first algorithmic resolutions already implemented and used within GENESIS 3.

8:30

4aMUa3. Real-time finite-difference string-bow interaction floating point gate array (FPGA) model coupled to a violin body. Pfeifle Florian and Bader Rolf (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, 20354 Hamburg, Germany, Florian.Pfeifle@haw-hamburg.de)

A string/bow interaction model as proposed by Bader, R.: Whole geometry Finite-Difference modelling of the violin. In: Proceedings of the Forum Acusticum 2005, 629-634, 2005 and a 3D violin geometry with top plate, back plate, and enclosed air was implemented in real-time using an FPGA hardware implementation. The bow/string interaction uses conditions for gluing and tear-off the string to/from the bow, where gluing happens with relative string/bow velocity below a threshold and tear-off with either the string curvature or the string tension at the bow point too high modelling the cases appearing with normal Helmholtz motion or the occurrence of subharmonics, respectively. The model can be played changing bow pressure and velocity, bow point on the string, and string length. The resulting sounds are highly realistic showing gradual timbre changes from normal Helmholtz motion, double slit motion, weakening of the fundamental, or noise with very low pressures. It appears that these interaction conditions can realistically be used to model the bow/string interaction.

8:45

4aMUa4. Measurement and physical modelling of sound hole radiations of lutes. Florian Pfeifle (Musicological Inst., Univ. of Hamburg, Neue Rabenstrasse 13, 20354 Hamburg, Germany, Florian.Pfeifle@haw-hamburg.de)

A structural feature that can be found in many string instruments is a hollow body with one or several sound holes. The sound radiated from these holes interacts with the sound radiation from the rest of the body and perceivably influences the timbre and the loudness of the instrument. In this work three non-European lutes with sound holes are measured: the Mauretanian ginbri, the West-African gunubri, and the Chinese ruan. All of these instruments have distinct cavity air modes and a measureable Helmholtz frequency. Each instrument is measured with a 11×11 microphone array and analyzed with a focus on the radiated spectrum and sound intensity of the hole(s). In a further step, the findings of the measurements are compared to a Finite Element model and incorporated into a real-time finite differences physical model of the ruan.

9:00

4aMUa5. Synthesizing classic recording microphones characteristics using a spherical microphone array. Nils Peters and Andrew W. Schmeder (Ctr. for New Music and Audio Technologies, UC Berkeley, 1750 Arch St., Berkeley, CA 94720, nils@icsi.berkeley.edu)

Using spherical microphone arrays to form directed beams is becoming an important technology in sound field analysis, teleconferencing, and surveillance systems. Moreover, in scenarios for capturing musical content, the

recording and post-production process could be simplified through flexible beamforming technology. Often, audio engineers favor the use of conventional recording microphones over spherical microphone arrays which might be due to the engineer's preference for distinct spatial and timbral characteristics of different microphone types and brands. We present an approach to create beamforming pattern using a 144 channel spherical microphone array, which aims to match the distinct spatial and timbral characteristics of classic microphones. For this, we first measured the spatial and timbral characteristics of several classic microphones types as well as the characteristics of our spherical microphone array in an anechoic chamber. Using a regularized least-square approach, these data were then used for computing the filters for the spherical microphone array that forms the desired beams. We show the results of several microphone-beam simulations and compare them with the impulse responses of the original classic microphones. Advantages and limitations of our approach will be discussed.

9:15

4aMUa6. Numerical simulation of the sound of organ pipes. Michael Steppat (Beuth Univ. of Appl. Sci., Luxemburger Str. 10, 13353 Berlin, Germany, steppat@beuth-hochschule.de)

Numerical simulations of musical instrument sounds are useful in studies, when questions about the influence of physical parameters stand in first place. Differential equations can be used to solve the given problem. The results given as a discrete sequence can be stored in audiofiles and made audible. In the current project, the influence of the attack of organ pipes with respect to the reverb time of the acoustical environment in the church building is studied by different numerical calculation methods such as finite element method and computational fluid dynamics. The resulting sound pressure and the velocity in a vortex street of the jet emerging from a pipe can be calculated at different listening positions. The computation allows also visualization of the jet in a three dimensional view. In the tests, the same parameters used by organ builders to voice organs are used for the simulation. Main point is the strength of the resulting starting transient and its influence to the acoustical environment. The calculation process which includes the synthesizing and adding a reverb and the results with different voicing parameters will be presented.

9:30

4aMUa7. FireFader: A single degree-of-freedom force-feedback device for multimodal interaction with physical models. Edgar J. Berdahl (Audiokommunikation, Tech. Univ. of Berlin, Sekr. EN-8, Einsteinufer 17c, 10587 Berlin, Germany, eberdahl@mail.tu-berlin.de)

The design of a relatively inexpensive force-feedback device known as the FireFader is presented. It is controlled using physical models to provide multimodal force, auditory, and visual feedback in real time, and it is based upon a linear potentiometer fader coupled to a DC motor, also known as a "motorized fader". Lamps are connected electrically in parallel with the motor in order to visually communicate the strength of the force. The device is linked by a serial USB interface to a general-purpose computer, which employs a physical model to calculate the motor force as a function of the fader position. The USB interface causes delay of the control signal, but it facilitates easier programming and less expensive control. Furthermore, additional sensed parameters can help provide the illusion of more than a single degree-of-freedom (DOF) feedback, via modulation of the physical model parameters. For estimation of the downward force applied by the performer on the fader, a pair of force sensors can be sandwiched in between the motorized fader and the housing. In conclusion, we hypothesize that by providing multimodal feedback in real time, the FireFader may help promote the expressivity of new media interactions.

THURSDAY MORNING, 3 NOVEMBER 2011 — TOWNE, 10:00 TO 11:30 A.M.

Session 4aMUb

Musical Acoustics: Analysis of Instrument Sounds

Thomas R. Moore, Chair
Dept. of Physics, Rollins College, Winter Park, FL 32789

Contributed Papers

10:00

4aMUb1. Tuning parameters of the Nigerian slit drum. Ashley Cannaday and Thomas Moore (Dept. of Phys., Rollins College, Winter Park, FL 32789)

The slit log drum, sometimes referred to as a "talking drum," is an idiophone indigenous to many African and South Pacific cultures. It is made from a log that has been hollowed out through two square openings, which are separated by a solid piece of the wood. The solid piece is cut down the middle to produce two tabs that when struck produce pitches that are usually separated by a musical fifth. We report on an investigation of the tuning parameters of slit log drums from Nigeria using both numerical and analytical models. We conclude that the most efficient and effective method of tuning the drum is to carve the interior walls near the tabs so that they have different thicknesses, which is indeed is how the Nigerian artisans produce the two distinct pitches.

10:15

4aMUb2. Calculation of Helmholtz frequency of a Renaissance vihuela string instrument with five tone holes. Rolf Bader RB (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, 20354 Hamburg, Germany, R_Bader@t-online.de)

A replica of the historical *Guadalupe* vihuela, a Renaissance string instrument was investigated. As it has five sound holes a formula was developed to calculate its Helmholtz frequency radiated from these five holes. Therefore, using a 128-microphone array, the radiation pattern at the Helmholtz frequency of 138 Hz was measured showing strong radiation at the holes near the center of the body and lower radiation at the holes near the upper top plate boundary. Calculating a mean radius from the single radii and weighting them with their radiation strength, the measured Helmholtz frequency is calculated correctly. This is not the case when including the ornamentation of the sound holes into the calculation which indeed cover

65% of their area. Additionally, the overall radiation from the different top plate parts of this vihuela was compared to that of another much smaller vihuela and those of a classical guitar showing the *Guadalupe* replica to have a very large frequency range of strong sound hole radiation up to 500 Hz, where the classical guitar is stronger in the bass but its sound hole radiation part is restricted to lower frequencies. This makes the vihuela a mixture between a guitar and a lute.

10:30

4aMUb3. Video analysis and modeling of the kalimba tine. Daniel O. Ludwigsen (Phys. Dept., Kettering Univ., 1700 University Ave., Flint, MI 48504, dludwigs@kettering.edu)

The kalimba, like the more traditional mbira, uses plucked metal tines mounted to a wood resonator. The mounting is similar to a three point flexural test and provides an initial strain to part of the tine. The plucked end of the tine is free once released, but modeling the other end is less obvious. The motion of the longest tine of the treble kalimba (B3, 247 Hz) was captured via high speed video (1200 f.p.s.). Analysis of tine displacement informed the boundary conditions of an Euler–Bernoulli model for this thin beam vibration, which in turn predicted mode shapes and frequencies. The two lowest mode frequencies can be compared to prominent features in the spectrogram of recorded tones. Rapidly decaying harmonic content observed in the spectrogram is not suggested by this simple model; a more sophisticated approach is required to fully understand the behavior of the kalimba tine.

10:45

4aMUb4. Validation of a descriptor, the "sum function", related to "quality" and derived from the input impedance of wind instruments. R. E. Causse, P. Eveno, B. Kieffer (IRCAM (UMR CNRS 9912), 1 place Igor Stravinsky, 75004 Paris, France), J. Gilbert, J. P. Dalmont (LAUM (UMR CNRS 6613), Le Mans, 72085 France), and J. F. Petiot (IRCCYN (UMR CNRS 6597) Ecole Centrale de Nantes, 44321 Nantes, France)

The quality of a musical instrument embraces many aspects such as tuning, ease of play, tone, etc. This study aims to validate the use of the sum function (SF) proposed by Wogram from the measurement of input impedance as a descriptor of quality. This work is part of a wider project, PAFI (Aid Platform for Instrumental Making), supported by the French National Agency of Research. To validate the choice of the SF, we created a family of trumpets made from a basic instrument for which the leadpipe will be slightly modified for each model. The SF was calculated for a range of selected frequencies from the measurement of the input impedance of these different trumpets. The next step was to ask musicians experts to play these instruments, to measure the playing frequencies, and to note their feedback about quality. These tests were supplemented by comparative tests carried out this time with the help of a robotic artificial mouth. The final step involved was to try to identify correlations between the SF and the results of various tests and propose correction factors to be made to the formula of the SF, related to the nuance or range for example.

11:00

4aMUb5. Modal response and sound radiation from a hammered dulcimer. Benjamin Y. Christensen, Kent L. Gee, Brian E. Anderson, and Alan T. Wall (Dept. of Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, ukeben@gmail.com)

The sound radiation of the hammered dulcimer has been investigated. The dulcimer studied is a 16/15 fixed-top instrument of typical size with Baltic birch (laminated plywood) soundboard and back. To determine the instrument body resonances, the dulcimer was driven at the treble and bass bridges with a shaker. Accelerometers were used to obtain the resonance frequencies of the soundboard and back, and a microphone was placed inside the instrument to obtain the cavity resonances. The individual resonance peaks found were further investigated using scanning laser Doppler vibrometry and near-field acoustical holography. Preliminary results show that there is little modal response of the instrument at the fundamental frequencies of the lowest notes of the dulcimer. In addition, the vibration coupling to the back plate through the internal bracing causes it to serve as a second soundboard. Lastly, the holography results indicate significant radiation from the sound holes at some frequencies, which may contradict the commonly held notion that the dulcimer sound holes are largely decorative.

11:15

4aMUb6. Equivalent circuit modeling and vibrometry analysis of the Udu Utar Nigerian drum. C. Beau Hilton (Dept. of Humanities, Classics, and Comp. Lit., Brigham Young Univ., 4110 JFSB, Provo, UT 84602, wearoscarwilde@myway.com), Brian E. Anderson, and Hillary Jones (Brigham Young Univ., N283 ESC, Provo, UT 84602)

The udu drum is both an aerophone and an idiophone played with both of the musician's hands. It originates from Nigeria where it began as a functional water pot made out of fired clay. At some point, a side hole was cut into the pot and it became a musical instrument, traditionally played by women, which had an important role in religious ceremonies. The udu is capable of producing deep tones that result from acoustic resonances similar to those of Helmholtz resonators, though with a second hole. It also may produce higher pitch sounds that result from the musician tapping the surface of the udu. This paper will discuss one-dimensional equivalent circuit modeling of the acoustic resonances of the udu. A comparison of the resonance frequencies in the equivalent circuit modeling to measured resonance frequencies will be given. Additionally, an analysis of the structural modes of the udu as measured by a scanning laser vibrometer will be given, along with some insights into the sound produced by striking the drum at different locations. This information may be used by udu designers to better tune these instruments.

Session 4aNS

Noise and Physical Acoustics: Launch Vehicle Noise I

Kent L. Gee, Cochair
Dept. of Physics and Astronomy, Brigham Young Univ., N243 Eyring Science Center, Provo, UT 84602

R. Jeremy Kenny, Cochair
Marshall Space Flight Center, Bldg. 4203, Huntsville, AL 35812

Chair's Introduction—8:00

Invited Papers

8:05

4aNS1. Further development of a launch pad noise prediction model. Kenneth J. Plotkin (Wyle Labs., 241 18th St. South, Ste. 701, Arlington, VA 22202) and Bruce T. Vu (NASA Kennedy Space Ctr., Mail Code NE-M1, Kennedy Space Ctr., FL 32899)

A model, PAD, has been developed for prediction of noise in the vicinity of launch vehicles, with specific application to the mobile launcher and tower for the Ares I launch vehicle. It follows the basic principles of a traditional NASA model (NASA SP-8072, 1971), with updated source components, including impingement and water suppression. The recent 5% scale Ares scale model acoustic test (ASMAT) exhibited sources not properly represented in PAD. These sources are noise increase associated with flat plate deflection of a supersonic plume, and generation of noise from impingement of the plume on an edge such as a launch mount or the edge of the exhaust hole in the launcher deck. New sources, based on ASMAT measurements, have been added to PAD to account for these effects. Treatment of the launch deck has also been generalized to permit full three dimensional launcher configurations, rather than a simple two dimensional arrangement with the tower over the flame trench. The prediction domain has also been expanded to include exhaust plume noise levels on the underside of the deck. (Work supported by the National Aeronautics and Space Administration.)

8:25

4aNS2. Effects of diffraction for acoustic loads on building structures. Louis C. Sutherland (5701 Crestridge Rd., Apt. 243, Rancho Palos Verdes, CA 90275)

Acoustic loads on building structures from external noise are influenced by acoustic diffraction of the incident wave front. An experimental study by Wyle Labs of diffraction consisted of measurements on a solid ground-mounted cubical obstacle insonified by approximately plane waves. The diffraction effect is equal to the sound level at various positions on the cube relative to the free field sound level. The former was measured with a microphone mounted flush with the surface of the cube. As expected, the diffraction effect varied systematically with the ratio of the cube side to wavelength and is portrayed by contours of equal diffraction correction. The maximum correction was equal to or more than the expected 6 dB near the center of the side facing the source but also exceeded 6 dB near the center of the back (shadow) side of the cube when the ratio of cube side to wavelength is close to 1. This positive diffraction correction may not normally be considered in assessment of noise on the shadow side of buildings. The experimental data are shown to be consistent with diffraction theory for a single cube. Theoretical predictions of diffraction effects for spherical and cylindrical obstacles are also shown.

8:45

4aNS3. Effects of ground impedance on large rocket motor noise measurements. Debbie Pilkey (ATK Aerosp. Systems, UT40-LF3, P.O. Box 707, Brigham City, UT 84302, deborah.pilkey@atk.com), R. Jeremy Kenny (Acoust. and Stability Team, ER42 , NASA Marshall Space Flight Ctr., Huntsville, AL 35812), and Jared Haynes (Qualis Corp. / ESTS Group, NASA Marshall Space Flight Ctr., Huntsville, AL 35812)

Free field acoustic measurements have been collected on large solid rocket motors at the ATK test facility in Promontory, Utah. Ground effects have been measured to understand the impact on the overall data collection effort. Ground effects were measured at two test stands with vastly different terrain that hold the reusable solid rocket motor (RSRM) and RSRMV (five-segment RSRM) static test motors. Techniques for measuring and understanding ground effects are investigated, and examples presented from two different methods.

9:05

4aNS4. A review of large solid rocket motor free field acoustics. Debbie Pilkey (ATK Aerosp. Systems, UT40-LF3, P.O. Box 707, Brigham City, UT 84302-0707, deborah.pilkey@atk.com) and R. Jeremy Kenny (Acoust. and Stability Team, ER42, NASA Marshall Spaceflight Ctr., Huntsville, AL 35812)

Approximately twice a year, ATK Aerospace Systems has static fired an assembled reusable solid rocket motor (RSRM) in a horizontal configuration at its test facility in Utah. The firings took place on elevated terrain with the nozzle exit plume mostly undeflected and the landscape allowing placement of microphones within direct line of sight to the exhaust plume. NASA and ATK underwent a significant effort over several years to collect acoustic data in the free field and on the motor itself during RSRM static test firings. These data were used to characterize the acoustic field and to update the prediction methodologies in the monograph NASA SP-8072 “Acoustic Loads Generated by the Propulsion System.” This work represents a review of the free field acoustics generated during large solid rocket motor firings and may be the only repeatable free field acoustic experiment on motors like these.

9:25

4aNS5. Mechanism of acoustic radiation from Supersonic Jets Impinging to Inclined Flat Plates. Seiji Tsutsumi, Taku Nonomura, Kozo Fujii (JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa, 252-5210, Japan), Yuta Nakanishi, Koji Okamoto (Univ. of Tokyo, Kashiwa, Chiba, 277-8561, Japan), and Susumu Teramoto (Univ. of Tokyo, Bunkyo-ku, Tokyo, 113-8656, Japan)

Acoustic wave radiated from supersonic cold jets impinging to inclined flat-plates is investigated numerically with the help of the experimental work. This type of acoustic generation is important for estimating and minimizing the acoustic loading of launch vehicle at lift-off. Through the study on the 45-degree-inclined flat plate located 5 D downstream from the nozzle exit, two noise sources are found to be generated due to the jet impingement: (1) interaction between the vortex of the jet shear-layer and the shock waves appearing at the jet impingement region and at the downstream region, (2) the Mach wave radiated from the large-scale vortex structure of the flow downstream of the plate. The former is similar to the broadband shock-associated noise. Those features are clearly confirmed by applying the proper orthogonal decomposition analysis to the numerical result. Prediction accuracy of 5 dB in far-field OASPL is obtained in the current numerical technique.

9:45

4aNS6. Large-Eddy-Simulations of over-expanded supersonic jet noise for launcher applications. Jean-Baptiste Dargaud, Julien Troyes, and François Vuillot (Onera - The French Aerosp. Lab, F-92322 Châtillon, France)

During the lift-off phase of a space launcher, powerful rocket motors generate harsh acoustic environment on the launch pad. Following the blast waves created at ignition, jet noise is a major contributor to the acoustic loads received by the launcher and its payload. This paper describes recent simulations performed at ONERA to compute the noise emitted by solid rocket motors at lift-off conditions. Far-field noise prediction is achieved by associating a LES solution of the jet flow with an acoustics surface integral method. The computations are carried out with in-house codes CEDRE for the LES solution and KIM for Ffowcs Williams and Hawkings porous surface integration method. This work has been conducted in the framework of the cooperation on launcher acoustics between CNES (French National Space Agency) and JAXA (Japan Aerospace Exploration Agency) involving the French AEID research group. The test case is that of a reduced scale solid rocket motor, fired vertically and has been provided by JAXA. Computations were run for varied numerical conditions, and the final paper will detail results and discuss comparisons with experimental acoustic measurements.

10:05–10:20 Break

10:20

4aNS7. On the use of tailored functional bases for space launcher noise sources localization and reconstruction. Damiano Casalino, Samuele Santini (CIRA, Italian Aerosp. Res. Ctr., Capua, I-81043, Italy, d.casalino@cira.it), Mariano Genito (ELV S.p.A., Colleferro, I-00034, Italy), and Valerio Ferrara (AVIO S.p.A., Colleferro, I-00034, Italy)

A source localization and inverse reconstruction methodology is applied to analyze wall pressure signals measured on the surface of a space launcher mock-up. The methodology is based on the use of elementary acoustic solutions tailored to the space launcher geometry as functional basis. Two classes of functional bases are considered: plane waves impinging on the surface of an infinite cylinder computed analytically through a literature formula, and plane waves impinging on the real space launcher computed numerically through a FEM computational acoustic technique. In both cases, the 3D acoustic field resulting from the impingement of an arbitrarily oriented plane wave is obtained by a truncated series summation of elementary axial-symmetric solutions. These two classes of functional bases are proven to provide consistent results in the addressed frequency range. However, although the analytical basis enables a faster localization of the noise sources that take place during the rocket firing, the numerical basis is expected to enable, at an acceptable computational cost, a more reliable reconstruction of the acoustic loads on the space launcher surface in a higher frequency range. The proposed FEM-based beam-forming and source reconstruction technique is therefore a useful tool for the vibro-acoustic design of future launch vehicles.

10:40

4aNS8. Energy-based acoustic measurement system for rocket noise. Michael M. James (Blue Ridge Res. and Consulting, LLC, 15 W. Walnut St., Ste. C, Asheville, NC 28801, Michael.James@BlueRidgeRes.com) and Kent L. Gee (Brigham Young Univ., Provo, UT 84602)

Accurate estimates of the vibroacoustic loading placed on space vehicles and payloads during launch require knowledge of rocket noise source properties and near-field acoustic energy flow characteristics. Without these data, structures may not be designed to handle the correct vibroacoustic loads, which can result in either an over-built, excessively massive structure or an under-designed vibration mitigation system that could result in damage to payloads. These measurements are difficult to perform because of the extreme nature of the acoustic and temperature environments near the rocket plume as well as the large physical size of the rocket noise source. With these design constraints in mind, a field-deployable data acquisition system and energy-based measurement probe have been developed to measure the magnitude, directivity, and spectral content of the rocket source. Initial measurements with various prototypes were conducted during a static test fire at ATK Space Systems Test Services in Promontory, Utah with limited results presented here. [Work sponsored by NASA John C. Stennis Space Center.]

Contributed Papers

11:00

4aNS9. Low-frequency calibration of a multidimensional acoustic intensity probe for application to rocket noise. Jarom H. Giraud, Kent L. Gee, Scott D. Sommerfeldt, R. Troy Taylor (Dept. of Phys. and Astronomy, Brigham Young Univ., Eyring Sci. Ctr., Provo, UT 84602, kentgee@byu.edu), and Jonathan D. Blotter (Brigham Young Univ., Provo, UT, 84602)

Measurements of acoustic vector quantities in the near field of a solid-fuel rocket motor are useful for enhancing prediction models related to rocket noise. However, the probes used to measure these quantities traditionally have low-frequency bandwidth limitations (e.g., below 45 Hz) thereby excluding the lowest, and in some cases, the loudest frequencies generated by large rocket motors. At these low frequencies, the phase and magnitude mismatch between microphones become greater and the acoustic phase separation between any two microphones becomes smaller, resulting in more error in estimating the pressure gradient between microphones. To investigate the low-frequency response of an acoustic intensity probe, a turntable is used to rotate a four-microphone probe with variable microphone spacing in a low-frequency noise field and an experimental assessment of the bandwidth is given for both magnitude and directional response. Also discussed is the effectiveness of a microphone interchange calibration technique to remove amplitude and phase mismatch and increase the usable bandwidth of the probe.

11:15

4aNS10. On the use of prepolarized microphones in rocket noise measurements. R. Troy Taylor, Kent L. Gee, Jarom H. Giraud, Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., Eyring Sci. Ctr. 84602, kentgee@physics.byu.edu), Jonathan D. Blotter, and Curtis P. Wiederhold (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

The acoustic field near large-scale solid rocket motors represents a harsh, high-amplitude noise environment rich with high-bandwidth acoustic shocks. Type-1 prepolarized microphones may be used in these environments with the benefit of reduced cost and measurement because they require only a constant-current supply available in many data acquisition systems. However, there are two potential issues related to microphone response that should be considered. The first is a well-known RC-lowpass filter effect that is associated with using insufficient current to drive long cables with relatively high capacitance. The second has to do with temporary failure of the constant-current supply due to an insufficiently fast response time in representing rapid voltage changes at shocks, which results in spurious, capacitive-like effects in the waveform data that are also manifest as a low-frequency roll-up in the spectrum noise floor. An experiment was conducted to identify under what circumstances these waveform effects arise. Data were measured from a solid rocket motor using several combinations of transducer, cable type, cable length and constant current supply. Results and mitigation methods found from the experiment are discussed. These include increasing the supply current, using low-impedance cables, and selecting microphones with low sensitivities.

11:30

4aNS11. Statistical analysis of noise from solid rocket motors. Stuart A. Harper, Kent L. Gee, Jarom H. Giraud, and Michael B. Muhlestein (BYU Dept. of Phys. and Astronomy, N283 ESC, Provo, UT 84602)

In an effort to better understand the properties of rocket noise, the statistical properties of noise data from various-sized solid propellant rocket motors. Time waveform data sampled at 204.8 kHz using 6.35 and 3.18 mm microphones were collected near motors with nozzle exit diameters ranging from 0.13 to 1.22 m. Non-Gaussian features of the data have been explored by calculating estimates of the probability density functions of the data and various statistical moments, including skewness and kurtosis. This has been carried out for both the pressure waveform and its first order time difference to better understand the formation of acoustic shocks within the noise. The analysis shows greater similarity between the statistics for the pressure than for the time derivative estimates.

11:45

4aNS12. On the computation of farfield cross-spectra and coherences from reduced parameter models of high speed jet noise. Havard Vold, Parthiv Shah, and Mike Yang (ATA Eng., Inc., 688 Deepwood Dr. Charleston, SC 29412)

Reduced parameter models of jet noise mechanisms serve as computational vehicles to estimate sound pressure and directivies at arbitrary locations in the farfield. The customary formulations have been successful at predicting autospectra and directivities, but the calculation of crossspectra and coherences has not been attempted. The authors will present a procedure for calculating crossspectra and coherences from the simple source model, i.e., an equivalent monopole density over a volume enclosing the jet noise sources. It will be shown that realistic coherences and crossspectra are only well defined when several mutually incoherent noise sources are being considered, and both convergence and spatial aliasing phenomena will be defined and investigated.

Session 4aPA

Physical Acoustics: Measurements, Applications, and More

Martin D. Verweij, Chair
Lab. of Electromagnetics Research, Delft Univ. of Tech., Mekelweg 4, Delft, 2628 CD, The Netherlands

Contributed Papers

7:30

4aPA1. Photoacoustic detection of thin layers of explosives. Logan Marcus, Richard Raspet, and Slava Aranchuk (Univ. of Mississippi, NCPA, 1 Coliseum Dr. University, MS 38677, LSMarcus@olemiss.edu)

Explosives may be identified by their infrared absorption spectra. Photoacoustic measurements, which measure the acoustic output generated by the explosive sample absorbing radiation from a modulated laser beam, are efficient at detecting and identifying explosives. We are investigating the standoff photoacoustic detection of surface traces of explosives on a variety of substrates using a laser Doppler vibrometer. This talk describes the detailed modeling of the measurement. A modulated infrared Gaussian profile laser source is absorbed in the layer of explosive and in the substrate. The heat absorbed raises the temperature of the explosive and the substrate leading to expansion and surface displacement. The heat absorbed by the gas adjacent to the solid generates an outward propagating acoustic wave. A laser Doppler vibrometer measures the phase shift due to the surface displacement and the compression and refraction of the acoustic wave. A cylindrically symmetric calculation of each physical step will be described. Calculations demonstrating how substrate parameters such as coefficient of thermal expansion, IR absorptivity, and Poisson's ratio effect the detection threshold of the measurements will be presented.

7:45

4aPA2. Sliding dynamic studies by use of elastography. Soumaya Latour, Stefan Catheline, Michel Campillo, Francois Renard, Christophe Voisin, Eric Larose, and Thomas Gallot (ISTERRE, Grenoble Univ., 38000 Grenoble, soumaya.latour@obs.ujf-grenoble.fr)

To get an insight into the processes underlying dynamic friction that plays an important role in seismic sources, we developed a sliding dynamic experiment coupled to elastography imaging. This experimental setup permits to observe simultaneously the frictional interface and the waves emitted in the bulk during slipping. We use soft solids made of hydro-organic gel of PVA, in contact with either glass or sandpaper. The huge interest of such soft solids is that elastography allows to observe in real time the rupture nucleation and propagation, as well as shear waves themselves inside the medium. We investigate the friction in two different cases. In the case of friction on sand paper, links are formed between the gel and the sand paper by local pinning. The breaking of these links emits a characteristic wave pattern, and their occurrence is related to the local sliding velocity. In a very different way, when the gel slide on a glass surface, with an interlayer of sand grains, the slip occurs as successive rupture events, with a rupture front crossing the whole surface. We can study then the rupture velocity, and in the cases of ruptures faster than the shear wave velocity, we observe a Mach cone of shear waves.

8:00

4aPA3. Two techniques for measurement of flow resistance. Eric C. Mitchell, Anand Swaminathan (Grad. Program in Acoust., Pennsylvania State, P.O. Box 30, State College, PA 16804), Steven L. Garrett, and Matthew E. Poese (Pennsylvania State Univ., State College, PA 16804)

The accurate measurement of flow resistance has many applications in acoustics. In our laboratory, it is particularly important for the characterization of materials used as regenerators in thermoacoustic refrigerators and for the quantification of leakage paths in complex assemblies. This presentation will describe two techniques for flow resistance measurements made at sufficiently low Reynolds numbers that the resistances measured for unidirectional flow are relevant to acoustic flows. One technique uses a "constant current generator" configuration to characterize stacked stainless steel screens and the other uses an Airpot® graphite piston in a glass cylinder to produce a constant pressure difference and accurate flow rate. Both techniques use air at atmospheric pressure as the test fluid. The constant current technique produces results that are consistent to $\pm 3\%$ for stacks of stainless steel screens that vary in thickness from 10 to 50 screens. The Airpot® technique can produce similar accuracy for flow resistances as large as 10^{10} Pa-sec/m^3. [Work supported by the Applied Research Laboratory and the U.S. Department of Energy.]

8:15

4aPA4. Statistical analysis of a characteristic shock formation distance for high-amplitude noise. Michael B. Muhlestein and Kent L. Gee (BYU Dept. of Phys. and Astronomy, N283 ESC, Provo, UT 84602, mimuhle@gmail.com)

Previous research involved investigating a characteristic shock formation distance for Gaussian, finite-amplitude noise propagating in a cylindrical plane wave tube [Muhlestein and Gee, POMA 12, (in press)]. In particular, the evolution of the probability density function of the pressure and the first-order time derivative of pressure along with the skewness of the pressure derivative were experimentally studied. It was concluded that a constant-factor modification to the nonlinear distortion length defined by Gurbatov and Rudenko may yield a suitable characteristic shock formation distance in a statistical sense. Additional Gaussian noise data with a broader frequency range have now been taken, and the effects of boundary layer dispersion considered. Furthermore, noise with other statistical distributions and pressure statistics mimicking high velocity jet noise have been examined. These data are analyzed statistically as before using probability density function estimates and the skewness of the pressure derivative. An additional figure of merit, the characteristic number of shocks per zero crossing, is also examined.

8:30

4aPA5. An investigation into the interaction of a grazing angle broadband spherical audio signal with the dynamically rough air–water interface of shallow flows in rivers and channels. Andrew Nichols, Kirill V. Horoshenkov, Simon J. Tait (School of Eng., Univ. of Bradford, Bradford, West Yorkshire, BD71DP, United Kingdom), and Keith Attenborough (The Open Univ., Milton Keynes, MK7 6AA, United Kingdom)

Laboratory measurements were made of a broadband audio signal transmitted and received at grazing angles over a range of shallow water flow regimes. Synchronous measurements of local surface fluctuation were taken using a thin-wire wave gauge positioned at the point of specular reflection. The first and second statistical moments of the acoustic intensity at the receiver are shown to be directly related to the second statistical moment of the water surface fluctuations. It was hypothesized that this relationship was due to the path-length of the dominant signal fluctuating in direct relation to the interface fluctuations at the specular reflection point. This hypothesis is

corroborated by analysis of the second statistical moment of the "time-of-flight" of the surface-reflected signal, and by direct analysis of the instantaneous water surface position.

8:45

4aPA6. Waveguide sound propagation in a turbulent atmosphere above a statistically rough surface of the ground. Vladimir E. Ostashev (Coop. Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 325 Broadway, Boulder, CO 80305, vladimir.ostashev@colorado.edu), D. Keith Wilson, and Sergey N. Vecherin (U.S. Army Engineer Res. and Development Ctr., Hanover, NH 03755)

Waveguide sound propagation in a refractive, turbulent atmosphere above a statistically rough surface of the ground is considered. The waveguide is formed between the surface of the ground and turning points of sound waves in the downwind direction or in a nocturnal boundary layer of the atmosphere when the temperature increases with height. Using a modal decomposition of the sound field and the Chernov method, closed-form equations for the coherence function of the sound field are derived. The solution is expressed in terms of the effective turbulence spectrum, which is a linear combination of the 1-D spectra of temperature and wind velocity fluctuations, and a spectrum of the surface roughness. The coherence function can be calculated with equations obtained or by using the effective spectrum in the already existing code for the coherence function of the sound field propagating in a refractive, turbulent atmosphere above a flat ground [D. K. Wilson, V. E. Ostashev, and M. S. Lewis, Waves Rand. Comp. Media, V. **19**, 369–391 (2009)]. The derived effective spectrum is also used to study the relative contributions of atmospheric turbulence and surface roughness to the coherence loss of propagating sound.

9:00

4aPA7. Performance of thermoacoustic device and its thermal contact to a source. Ivan Rodriguez, Orest G. Symko, and Myra Flitcroft (Dept. of Phys. Astronomy, Univ. of Utah, 201 James Fletcher Bldg., 115 South 1400 East, Salt Lake City, UT 84112-0830)

An important application of a thermoacoustic prime mover is in the conversion of heat to electricity when coupled to a sound to electricity converter. In order to achieve maximum sound output, the thermal coupling of the heat source is critical. This was studied here by using (i) a constant heat flow heat source and (ii) a constant temperature difference heat source. As in an electric system where the higher efficiency of power delivery is for a constant voltage source, maximum heat is delivered to the acoustic device from a constant temperature difference heat source. This was investigated on a 1.94 kHz thermoacoustic engine coupled to an acoustic cavity. A shutter located inside the cavity made it possible to have sound on and sound off. In the constant heat flow approach, the shutter technique gave a direct value and measure of heat which was converted to sound. The constant temperature difference approach provided the most heat input for maximum sound output. The temperature profile of cold heat exchanger, hot heat exchanger, and stack was determined by thermocouples.

9:15

4aPA8. Sound propagation in a pipe with dynamically rough boundary. A. Romanova, K. V. Horoshenkov, and S. J. Tait (Dept. of Eng., Design and Technol., Univ. of Bradford, Bradford, BD7 1DP, United Kingdom, a.romashk@gmail.com)

The surface pattern of the water flow in partly filled circular pipe contains information on some key characteristics which are important for a better understanding of the hydraulic energy losses and turbulent processes. Surface pattern variation is a dynamic and nonstationary process which is difficult to measure directly. In this sense, airborne sound waves provide an attractive statistical mean which describes the apparent boundary roughness, its spatial correlation function, and frequency spectrum. These parameters can then be linked to key hydraulic characteristics of the flow. This work presents new experimental setup, which is used to study these characteristics under controlled laboratory conditions and allows for simultaneous measurements of the acoustic field in the pipe and water surface roughness. The acoustic technique makes use of Gaussian pulses which are transmitted in air above the turbulent flow of water over a carefully instrumented section and recorded on an intensity probe. The results obtained for a range of flow regimes illustrate that it is possible to relate unambiguously the variation in the recorded acoustic field to a short-term variance in the water surface roughness and its spectrum. A suitable theoretical foundation based on small perturbation theory is proposed to interpret the obtained data.

9:30

4aPA9. Acoustic radiation torque of arbitrary shaped waves. Glauber Silva (Inst. Fisica, Univ. Fed. Alagoas, Maceio, AL 57072-970, Brasil, glauber@pq.cnpq.br) and Farid Mitri (Los Alamos Natl. Lab. Acoust. & Sensors Technol. Team, MS D429 Los Alamos, NM 87545)

In this work, a general expression of the acoustic radiation torque produced over an object by an arbitrary shaped beam is presented. The object is immersed in an inviscid fluid. To obtain the expression, the stress tensor of the angular momentum is integrated over a farfield virtual sphere, which encloses the object. The incident and scattered acoustic fields are represented through the partial wave expansion in the spherical coordinates. After performing the integration, the radiation torque is given in terms of the beam-shape and the scattering coefficients, which come from the incident and scattered partial wave series, respectively. The method is applied to compute the torque upon a fluid sphere by a vortex (first-order) Bessel beam in both on- and off-axis configurations. It is shown that the torque only arises on absorbing spheres. In this case, the angular acceleration and velocity are obtained from the torque. It is found that the angular acceleration may reverse its direction depending on the wave frequency. In conclusion, the presented theory might be useful for describing the particle dynamics of a sphere subjected to acoustic vortex beams.

9:45

4aPA10. Non-adiabatic geometric phase of elastic waves. Jeremie Boulanger, Nicolas Lebihan (Gipsa-Lab, Grenoble Univ., FRANCE, Jeremie.Boulanger@gipsa-lab.grenoble-inp.fr), Stefan Catheline (ISTERRE, Grenoble Univ., 38000 Grenoble), and Vincent Rossetto (LPMMC, Grenoble Univ., 38000 Grenoble)

We study the transport of elastic waves in a waveguide with helical shape. Polarization exhibits a geometric phase (or Berry phase): The polarization plane rotates along the helix following a geometric rule called parallel transport. Whereas this experiment is similar to the first experimental evidence of a Berry phase, by Tomita and Chiao [Phys. Rev. Lett. **57** (1986)], there is a major difference: The evolution of polarization is not adiabatic. This experiment therefore addresses the universality of the geometric phase beyond the adiabatic regime. We show that properties of the observed geometric phase coincide with the ones predicted by the adiabatic theory. The measured value of the phase is consistent (up to experimental uncertainty) with the theoretical value and no dependency with frequency is observable either.

10:00–10:15 Break

10:15

4aPA11. Utilization of an acoustic tomography array as a large sonic anemometer/thermometer. Sergey N. Vecherin (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH, Sergey.N.Vecherin@usace.army), Vladimir E. Ostashev (Coop. Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 325 Broadway, Boulder, CO 80305), and D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH 03755)

The temperature, wind velocity, and vertical and horizontal kinematic heat fluxes are important characteristics of the atmospheric surface layer. Point measurements of these meteorological parameters are often not representative due to their horizontal variations. For remote sensing of the area-averaged values of these parameters, we suggest using the acoustic tomography array at the Boulder Atmospheric Observatory (BAO). In this approach, the tomography array (with the horizontal size of 80 m by 80 m) is used, in essence, as a large sonic anemometer/thermometer for measurements of the area-averaged, instantaneous values of temperature and wind velocity. The area-averaged horizontal heat flux is then calculated from a time series of the area-averaged temperature and wind velocity. Feasibility of this approach is studied in numerical simulations of the BAO tomography array

with the use of LES fields of temperature and wind velocity. The results obtained show that the area-averaged values of temperature, wind velocity, and horizontal heat flux are reliably reconstructed. Numerical analysis of the LES fields indicates that the area-averaged vertical heat flux might be inferred from the horizontal flux. Preliminary experimental results obtained with the BAO acoustic tomography array show that this remote sensing technique is feasible.

10:30

4aPA12. Triggering of self-excited thermo-acoustic oscillations in a Rijke tube with a premixed laminar flame. Dan Zhao (Aerosp. Eng. Div. Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, zhaodan@ntu.edu.sg)

For a given Rijke tube, self-excited combustion oscillations could be caused by the transient growth of flow disturbances. A premixed laminar flame, anchored to a metal gauze, is considered to investigate the role of non-normality and the resulting transient growth in triggering such oscillations. The unsteady heat release is assumed to be caused by the flame surface variations, which results from the fluctuations of the oncoming flow. The flame is acoustically compact and its presence causes a jump in mean temperature. Coupling the flame model with a Galerkin series expansion of the acoustic waves enables the time evolution of the flow disturbances to be calculated. It was found that the nonlinear model can predict the mode shape and the frequencies of the excited oscillations very well. Moreover, the fundamental mode with the lowest frequency is the easiest one to be excited among all the acoustic modes. Linearizing the model and recasting it into the classical time-lag formulation provide insights on the mode selection and triggering. Finally, to gain insight about the stability behaviors of such non-normal Rijke tube, pseudo-spectra analysis is performed to obtain upper and lower bounds on the transient growth factor.

10:45

4aPA13. Synchronization of ultrasonic thermoacoustic devices. Myra Flitcroft and Orest G. Symko (Dept. of Phys. Astronomy, Univ. of Utah, 201 James Fletcher Bldg., 115 South 1400 East, Salt Lake City, UT 84112-0830)

The development of ultrasonic thermoacoustic devices opens up a new field of applications, in particular where heat can be converted acoustically to electricity in small systems. The power level for applications can be raised by incorporating such devices into array configuration. Since the acoustic devices are self-sustained oscillators, their phase at onset for oscillation is unpredictable; they are triggered on by a random fluctuation. Hence, maximum power output will be achieved by synchronizing the elements of an array. This can be accomplished by suitable coupling between them. Results are presented on the in-phase synchronization of ultrasonic thermoacoustic prime movers at $\sim$23 kHz, acoustically coupled by means of an acoustic cavity. Each engine has an inner volume of $\sim$2.7 mm^3. They were activated by means of separate wire heaters; the working fluid is air at one atmosphere. The demonstration of the synchronization of acoustic engines can be extended to many applications.

11:00

4aPA14. Photoacoustic spectrometer with a calculable cell constant for accurate absorption measurements of atmospheric aerosols and greenhouse gases. Keith A. Gillis (Temperature, Pressure, and Flow Metrology Div. NIST, 100 Bureau Dr., Gaithersburg, MD 20899-8360, keith.gillis@nist.gov) and Joseph T. Hodges (Chemical and Biochemical Reference Data Div. NIST, 100 Bureau Dr., Gaithersburg, MD 20899-8320)

A photoacoustic (PA) spectrometer for absolute optical absorption measurements of aerosols and greenhouse gases in ambient air has been developed. A theoretical analysis of the system in terms of the gas properties, continuous wave laser intensity modulation, and energy transfer relaxation rates will be discussed. The measured and predicted values for the PA system response differ by about 1%. The accuracy of the spectrometer is demonstrated by a probe of the absorption transitions of the *A*-band of O_2 in atmospheric humid air using reference line-shape parameters and accounting for reduced conversion efficiency due to relaxation effects. These transitions also provide a convenient method to monitor the system stability in the field. Observed detection limits are 3.1×10^{-9} W·cm^{-1}·Hz$^{-1/2}$ for absorption by gases and 1.5×10^{-8} W·cm^{-1}·Hz$^{-1/2}$ for absorption by soot particles (limited by fluctuations in the aerosol concentration). The sensitivity of the instrument is demonstrated with measurements of the amplified absorption resulting from ultrathin (>5 nm), nonabsorbing coatings on nanoscale soot particles. Applications to real-time monitoring of CO_2 concentration in ambient air and to measurements of the albedo of soot aerosols will be discussed. [Work is supported by NISTs Greenhouse Gas Measurements Program.]

11:15

4aPA15. Computational model for the dynamic stabilization of the Rayleigh-Bénard instability in rectangular enclosures. Randy M. Carbo (Graduate Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804), Robert W. M. Smith, and Matthew E. Poese (Penn State Univ., PA 16804)

When fluid within a container is heated from the bottom, onset of convection occurs when Rayleigh number, $\frac{g\beta\Delta T\ell^3}{\nu\chi}$, exceeds some critical value. If an acoustic field is imposed on the fluid in the container, the critical Rayleigh number is a strong function of the frequency and amplitude of that acoustic field as noted by Swift and Backhaus [J. Acoust. Soc. Am. **126**(5), 2009]. Results will be reported for a linear model constructed to predict the modified critical Rayleigh number, based on a full field solution of the hydrodynamic equations using the approach of Gelfgat [*J. Comp. Phys.* **156**, 1999]. The spatial portion of the differential equations was solved using the Galerkin method, and the dynamic stability was determined using Floquet analysis. One of the benefits of the approach compared to the averaging methods used by Gershuni and Lyubimov, [*Thermal Vibration Convection* (Wiley, New York, 1998)] is that the parametric stability boundary can also be recovered. This study includes a variety of container aspect ratios, boundary conditions, and Rayleigh numbers ranging from 10^3 to 10^8. [Work supported by the Office of Naval Research and ARL Exploratory and Foundational Research Program.]

11:30

4aPA16. One channel spatio-temporal inversion of acoustic wave in reverberant cavities. Ruppin Matthieu, Catheline Stefan, and Roux Philippe (ISTERRE, Grenoble Univ., 38000 Grenoble, FRANCE, matthieu.ruppin@obs.ujf-grenoble.fr)

It has been recently shown that it was possible to optimally recover the Green functions from a complex wave field despite of a non-isotropic distribution of the noise sources. The method used is based on a particular use of the inverse filter (IF) formalism which is called the passive IF. Based on this formalism, we have investigated the possibility to control the spatio-temporal degrees of freedom in a reverberant cavity for the focusing of waves (active processes). The understanding of this phenomenon can be very useful in a lot of different applications like in acoustical imaging, seismology, or telecommunications. In the present work, the spatio-temporal focalization of ultrasounds in reverberant cavities is studied using medical arrays and water tanks. Through experiments, a complete spatio-temporal inversion is realized to synthesize optimized emitting signals. The result generalizes the focalization control over a spatial vector and during an arbitrary time window.

11:45

4aPA17. Acoustics cavitation in microfluidics for sonoluminescence and sonochemistry. S. W. Ohl, T. Tandiono (Inst. of High Performance Computing, Singapore), D. Ow (Bioprocessing Technol. Inst., Singapore), E. Klaseboer (Inst. of High Performance Computing, Singapore), V. Wong (Bioprocessing Technol. Inst., Singapore), and C. D. Ohl (Nanyang Technol. Univ., Singapore)

Strong ultrasound is applied to a microfluidic channel to generate nonlinear surface waves which entrap bubbles at the gas–liquid interface to form oscillating bubbles. The ultrasound is generated by the piezoelectric transducer on the side of polydimethylsiloxane microchannel. The microchannel is attached to a glass slide through plasma bonding, while the transducer is glued by epoxy for strong coupling. The high speed photography shows that continuous cavitation clusters are formed within the

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microchannel as gas is injected. As they collapse rapidly, they are able to produce very intense concentration of energy that is able to emit light. This phenomenon is known as sonoluminescence. Previously, sonoluminescence is achieved via a single bubble or multiple bubbles in a bulk liquid. The authors report a realization of sonoluminescence in a microfluidic device. The same oscillating bubbles can also be used as micro-labs. They can trigger chemical reactions that require high temperature and pressure. We achieve the formation of OH radicals in a lab-on-a-chip device. In conclusion, nonequilibrium microbubbles can be induced in a microfluidic system. They oscillate and collapse, and in the process provide a source of energy concentration for the emission of light and the activation of chemical reactions.

THURSDAY MORNING, 3 NOVEMBER 2011 CALIFORNIA, 9:00 A.M. TO 12:00 NOON

Session 4aPP

Psychological and Physiological Acoustics: Perception, Physiology, and Models (Poster Session)

Andrew J. Lotto, Chair
Dept. of Speech, Language, and Hearing Science, Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721-0071

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

4aPP1. Psychophysical weights estimated for interaural cues in envelope slopes of amplitude-modulated waveforms. I-Hui Hsieh (Inst. of Cognit. Neurosci., Natl. Central Univ., 300 Jhongda Rd., Taoyuan, Taiwan, ihuihsieh@gmail.com) and Kourosh Saberi (Univ. of California-Irvine, Irvine, CA, saberi@uci.edu)

Current binaural theory contends that the auditory system encodes interaural delays in highpass-filtered complex sounds by phase locking to their slowly modulating envelopes. Spectrotemporal analysis of interaurally time-delayed highpass waveforms reveals the presence of a concomitant interaural level cue which could contribute to lateralization judgments. The current study systematically investigated the relative contribution of time and concomitant level cues carried by positive and negative envelope slopes of modified sinusoidally amplitude-modulated (SAM) high-frequency carriers. Psychophysical thresholds and observer decision weights were measured independently for the positive and negative modulation slopes of the acoustic signal. Decision weights were also measured to determine whether or not interaural delays are uniformly weighted at different temporal cycles of a SAM waveform. We found that lateralization of interaurally delayed SAM waveforms is influenced equally by ITDs in the rise and decay envelope slopes, and not by concomitant ILD cues, and that ITD cues are more heavily weighted in the initial few cycles of the SAM envelope.

4aPP2. Sensitivity to changing characteristics of Gaussian-shaped stimulus distributions in auditory categorization. Sarah C. Sullivan, Johnna A. Tanji, Andrew J. Lotto (Dept. of Speech, Lang., & Hearing Sci. , Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721-0071), and Randy L. Diehl (Univ. of Texas, Austin, TX 78712-0187)

This experiment examined the ability of human listeners to categorize sounds as a function of changing training distribution characteristics. Participants were presented non-speech sounds randomly sampled from two overlapping, Gaussian-like distributions. The sounds consisted of narrow-band noise bursts varying in center frequency. Participants mapped the distributions of sounds onto creatures in a video game receiving visual and auditory feedback about accuracy. The distributions were constant for half of the participants. For the other half, an unsignaled switch to distributions with a new optimal boundary occurred in the middle of the session. By examining obtained category boundaries one can distinguish between 3 possible responses to the switch: (1) adaptive switching resulting in the new optimal boundary being learned; (2) persistence of first learned boundary; and (3) averaging input across the entire session, resulting in an obtained boundary midway between the first and second optimal boundaries. The results demonstrate that most listeners could adaptively switch between distribution conditions with no external signal informing them of such a switch. The results indicate that most listeners are updating possible underlying distributions of input in a remarkably adaptive manner. [Work supported by NIH.]

4aPP3. Word production analysis of native speakers and second language learners by phonemic and categorical verbal fluency test. Keiko Asano (School of Medicine, Juntendo Univ., Hiragagauendai 1-1, Inzai-city, Chiba Pref ZIP270-1695, keiko_asano@sakura.juntendo.ac.jp)

This study investigated how different Japanese, Arabic, and Thai second language learners produce words orally from their native languages aspects of the Verbal Fluency Test. This test, which is in wide spread use in clinical and neuropsychological assessments, consists of two different tasks: phonemic and categorical sections. As a phonemic procedure, the participants were to produce orally as many different words as possible beginning with a certain letter within one minute. The names of specified items within categorical one were to be produced. These clinical field scores are only adopted as quantitative information. However, in this study, the relationship between the phonemic and categorical sections is also analyzed to evaluate the ability of second language learners' word production. It is commonly known that normal and healthy native speakers can produce more words in categorical section than that of the phonemic one while the speakers who are clinically disordered have results that are the reverse. The results show that native speakers produce more categorical words, which were the same results found in normal and healthy people. However among all three second language learners, there is different tendency. Categorical words were produced less frequency with no relevance to the proficiency level. The implication for the function of different brain area activation will also be discussed.

4aPP4. Monotic versus diotic thresholds in an amplitude modulation and quasi-frequency modulation discrimination task. Ewa Borucki, Allison I. Shim, and Bruce G. Berg (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697-5100)

This study investigated monaural and binaural differences in a task traditionally used to investigate the bandwidths of phase sensitivity. Subjects discriminated amplitude modulated (AM) tones and quasi-frequency modulated (QFM) tones presented diotically. An adaptive threshold procedure was used to estimate modulation depth needed to make the discrimination as a function modulation frequency for a 2000-Hz carrier. Threshold functions were often nonmonotonic, with nonmonotonicities observed at higher-modulation frequencies (above 600 Hz). This is likely due to the effects of cubic difference tones (CDTs) creating spectral cues, yielding lower thresholds. Subjects then completed the task at higher modulation frequencies monotically, demonstrating differences between monotic and diotic thresholds. Within subject thresholds also differed between left ear and right ear presentations. Some subjects yield nonmonotonic thresholds in the monotic condition, even when diotic thresholds were monotonic. For these subjects, it is likely that the CDT interaction is not consistent between the two ears, rendering them an unusable cue when stimuli are presented diotically. Distortion product otoacoustic emissions (DPOAEs) were measured and support the hypothesis that nonmonotonicities found are the effect of a CDT interacting with the low tone of the stimulus.

4aPP5. Converging evidence from behavior and electroencephalography for differences in the storage of streams versus individual sound objects. Lenny A. Varghese (Dept. of Biomedical Eng., 677 Beacon St., Boston, MA 02215), Virginia Best (Dept. of Speech, Lang., and Hearing Sci., Boston, MA 02215), and Barbara G Shinn-Cunningham (Dept. of Biomedical Eng., Boston, MA 02215)

Previous psychophysical experiments from our lab have suggested a difference in the way perceptual sound streams and discrete sound events are stored in short-term memory. We explored differences in brain activity during memory retention of these two kinds of sequences using EEG. Listeners heard a target sequence composed either entirely of short pitched tones (forming a stream) or natural sounds (e.g., a horn or a scream, which are perceptually disconnected), while scalp activity was recorded using EEG. After a 2 s retention period, listeners were required to detect a change in the ordering of a probe sequence (same/different). For a given sequence length, performance for pitched tone sequences was consistently higher than for natural sound sequences, suggesting that memory retention of an auditory stream requires less cognitive effort than retention of a sequence of perceptually disconnected sounds. Retention period alpha (8–12 Hz) oscillatory activity generally increased from pre-target levels in parietal and occipital electrodes for both types of sound sequences, with a trend toward more widespread increases for natural sounds. These results may indicate that alpha activity is related to the amount of cognitive effort required to maintain sound sequences in short-term memory. [Work supported by NSSEFF grant to BGS-C.]

4aPP6. Auditory training effects on auditory steady state responses. Vidal I. Hinojosa and Su-Hyun Jin (Su-Hyun Jin Dept. of Commun. Sci. and Disord., 1 University Station A1100, Austin, TX 78712, vihinojosa@gmail.com)

Human auditory steady-state responses (ASSRs) have been linked to recognition scores in normal hearing and hearing impaired adults (Dimitrijevic *et al.* 2004). By taking this into account ASSRs should improve over time when speech perception improves. Computerized aural rehabilitation programs such as the LACE program by Neurotone have claims of being able to improve speech in noise perception, such as a 2.2 dB improvement using the QuickSin Test as an outcome measure. The main purpose of this study is to answer the question "Can improvements in speech perception be tracked by electrophysiological methods such as ASSR?". For this study, ASSRs and HINT scores will be tracked before and after auditory training in two groups of subjects. The first group will be hearing impaired people who have had hearing aid experience before. The second group consists of hearing impaired people who have just been fit with hearing aids for the first time. ASSR stimulus will be modeled after speech as in the Dimitrijevic 2004 study. By tracking these improvements in speech perception objectively using the ASSR and subjectively using HINT scores, the study will hopefully add validation to objectively testing speech perception using ASSR techniques.

4aPP7. Physiological arousal and laughter acoustics. R. Toby Amoss, Noel B. Martin, and Michael J. Owren (Dept. of Psych., Georgia State Univ., P.O. Box 5010, Atlanta, GA, 30302-5010, tobyamoss@gmail.com)

Laughter is a ubiquitous human phenomenon that has been little investigated scientifically. To examine the relationship between laugher arousal level and acoustic output, bouts of laughter were recorded from undergraduates viewing humorous video clips. These participants provided continuous subjective ratings of funniness while watching the clips, and heart rate (HR) was collected concurrently to provide an objective measure of physiological arousal. As a first approach, comparisons focused on voiced laughter from nine males and eight females. For each individual, one high-amplitude and one low-amplitude laugh bout was identified for which HR could be extracted. Beats per minute increased significantly more with high-amplitude than low-amplitude bouts, an effect that was not likely due to physical exertion or movement artifact. No sex difference was found in the magnitude of HR change for either amplitude condition. Both subjective ratings of funniness and fundamental-frequency measures were significantly higher for higher-amplitude bouts, while harmonic-to-noise ratios trended lower for these sounds. Overall, results are consistent with the intuition that higher physiological arousal in vocalizers is reflected in higher vocal amplitude and faster, potentially less stable vocal-fold vibration in voiced laughter.

4aPP8. Phonetic and acoustic differences in child and adult laughter. Caroline Menezes and Samantha Diaz (Dept. of Health and Rehabilitation Sci., Univ. of Toledo, 2801 W. Bancroft St., Toledo, OH 43606)

This is a preliminary study comparing the acoustic differences in recordings of child and adult spontaneous laughter. Altogether, 100 laughter calls where analyzed from one male and female adult and one male and female child. Results indicate that bout and call durations of children and adult laughter are similar in duration; however, segmental durations show developmental differences. Children show variation between vowel and consonant durations unlike adults. Moreover, child vowels are longer in duration when compared to the adult vowels. Surprisingly, between the adult and child vowels, no difference was observed in mean pitch and mean intensity values. The most prominent difference between child and adult laughter is observed in vowel quality where children's F1 values of laughter vowels are relatively higher than adult's. The consonantal resonance in children is similar to their vowels. However, in adults, the consonant resonant frequencies are much higher than the vowel resonances. Therefore, while children may employ more extreme placements of jaw or tongue they have relatively limited articulatory movement from consonant to vowel when compared to adult laughter. This suggests interesting insights into development of children's speech utterances which need to be explored further.

4aPP9. A modified model for predicting breathiness judgments using partial and noise loudness measures. Mark D. Skowronski, Rahul Shrivastav (Dept. Speech, Lang., and Hearing Sci. , Univ. of Florida, Gainesville, FL 32611, markskow@ufl.edu), and David A. Eddins (Dept. Comm. Sci. and Disord., Univ. S. Florida, Tampa, FL 32620)

Breathiness is one of three classifications, along with roughness and strain, used to characterize disordered voice quality. Breathiness has been measured using a magnitude estimation task and modeled with a power function of perceptual measures [Shrivastav *et al.*, J. Acoust. Soc. Am. **129**, 1605–1615 (2011)]. The experiment was repeated using a matching task which required listeners to match the perceived breathiness of a vowel to a sawtooth-in-noise comparison by adjusting the signal-to-noise ratio of the comparison. The matching task provides a more accurate measure of perception and can potentially accommodate all three dimensions of dysphonic voice quality (breathiness, roughness, and strain) in a single experimental paradigm. Breathiness was modeled with a power function of noise-to-partial loudness ratio. Breathiness judgments (in decibels) varied over about 4 orders of magnitude and exhibited a linear relationship with the log ratio of noise-to-partial loudness. This data was modeled with a logistic function which accounted for breathiness saturation at low levels of noise loudness. Accuracy of the models is discussed as well as sources of variation in the experiment (talker, listener) and strategies for mitigating their effects on modeling accuracy.

4aPP10. A computational model for evaluating the interaural disparities using coincidence detection. Zbynek Bures (College of Polytechnics, Tolsteho 16, 58601 Jihlava, Czech Republic, buresz@vspj.cz)

In mammals, spatial hearing is supported by evaluation of two types of binaural disparities, interaural time difference (ITD) and interaural level difference (ILD), which is performed by auditory neurons in the superior olivary complex. Using a complex auditory model, which includes a binaural evaluation stage, the output of binaural neurons from medial and lateral superior olive is simulated and quantitatively compared with available experimental data. Both binaural disparities are evaluated using detection of coincidence of spikes arriving from the two channels: in case of ITD, the coincidence of two excitatory spikes is detected; in case of ILD, the coincidence of one excitatory and one inhibitory spike is detected. It is shown that

using a physiologically realistic set of parameters, evaluation of both ITD and ILD based on coincidence detection is capable of reproducing the observed neuronal responses. Furthermore, the model is shown to qualitatively reproduce the just-noticeable differences of binaural parameters depending on frequency and intensity. [Work supported by the project "Podpora a individualni rozvoj perspektivnich akademickych pracovniku na VSPJ" at the College Of Polytechnics Jihlava.]

4aPP11. Comparison of a 3 dimension model versus a 2 dimension-axisymmetric finite element model of an occluded ear canal. Guilhem Viallet (Dept. of Mech. Eng., ETS, 1100 Notre-Dame West, Montreal, QC H3C 1K3, Canada, guilhem.viallet.1@ens.etsmtl.ca), Franck Sgard Pr (IRRST, Montreal, QC, H3A 3C2, Canada), and Frédéric Laville (ETS, Montreal, Montreal, QC, HC 1K3, Canada)

Due to low cost and simplicity, earplugs are a widespread solution to prevent the problem of hearing loss in the workplace environment. In practice, earplugs often perceived are being uncomfortable and/or do not always perform as desired. The attenuation, based on a laboratory measurement, is often overestimated compared to *in situ* measurements. The use of a model of an occluded ear canal can help to build an individual measurement system of the attenuation and to develop an earplug with optimized attenuation. It is established that the unoccluded auditory canal can be approximated by a cylinder to predict the interior pressure field up to 6 kHz. A remaining question is whether this approximation holds true for an occluded ear. First, a simplified 2-D-axisymmetric finite element model of an ear canal coupled to a cylindrical earplug is developed. Special emphasis is on the coupling between the earplug and the lateral walls of the auditory canal. Second, a 3-D model of a real ear canal coupled to a cylindrical earplug is developed to examine the limits of the simplified model. Several assumptions about the deformation of the ear canal/earplug system are tested when comparing the sound attenuation provided by both models.

THURSDAY MORNING, 3 NOVEMBER 2011 — PACIFIC SALON 4/5, 9:00 TO 11:30 A.M.

Session 4aSCa

Speech Communication: Forensic Acoustics—On the Leading Edge of the Tidal Wave of Change About to Hit Forensic Science in the US? I

Geoffrey Stewart Morrison, Chair
School of Electrical Engineering, Univ. of New South Wales, Sydney, NSW, 2052, Australia

Chair's Introduction—9:00

Invited Papers

9:05

4aSCa1. Legal standards for the admissibility of expert testimony: Implications of the 2009 National Research Council report on forensic sciences. William C. Thompson (Dept. of Criminology, Law and Society, School of Social Ecology, Univ. of California, Irvine, 5300 Social and Behavioral Sci. Gateway, Irvine, CA 92697-7050)

This talk will review formal legal standards for the admissibility of expert testimony in the United States, focusing on the Daubert standard that is applied in Federal courts and the Frye standard that is applied in several major states (including California and New York). It will also discuss the National Research Council's 2009 critique of judicial "gatekeeping," particularly the NRC's stunning claim that courts have violated their own purported standards by allowing forensic scientists to present scientifically dubious testimony based on inadequately validated methods. It will conclude by providing suggestions for researchers in emerging areas of forensic inquiry, like forensic acoustical science, who contemplate testifying in court.

9:25

4aSCa2. The response to *R v T*. Can forensic acoustics play a leading rôle in a new wave of adoption of the likelihood-ratio framework? Geoffrey Stewart Morrison (Forensic Voice Comparison Lab., School of Elec. Eng. & Telecommun., Univ. of New South Wales, UNSW Sydney, NSW 2052, Australia)

In 2010, the England and Wales Court of Appeals ruled in *R v T* that the likelihood-ratio framework should not be used for the evaluation of evidence except "where there is a firm statistical base." It was not favorable to the use of likelihood ratios even if a firm statistical base does exist. It was, however, willing to accept evidence without a firm statistical base, but on the condition that likelihood ratios are not used. In response, 31 leading forensic scientists published a statement asserting that the likelihood-ratio framework is the logically correct way to evaluate evidence (as a logical framework it is not itself dependent on data or statistical models). This statement has been endorsed by the board of the European Network of Forensic Science Institutes, representing 58 laboratories in 33 countries. Forensic scientists have argued that the ruling is based on misunderstandings of both the likelihood-ratio framework and statistics. The presenter proposes that the magnitude of the response to *R v T* potentially heralds a new wave of adoption of the likelihood-ratio framework by forensic scientists and by the courts (the first wave was DNA in the 1990s) and that forensic acoustics can potentially play a leading rôle.

9:45–10:00 panel-discussion

10:00–10:15 Break

Contributed Papers

10:15

4aSCa3. An introduction to forensic gunshot acoustics. Steven D. Beck, Hirotaka Nakasone, and Kenneth W. Marr (BAE Systems, 6500 Tracor Ln., MS 27-16, Austin, TX 78725, steve.beck@baesystems.com)

Due to the proliferation of audio recording devices in the military, law enforcement, and the civilian community, there has been an increase in the number of recorded gunshot sounds submitted for forensic analysis. A gunshot sound is composed of one or more discrete acoustic transient events. The two primary acoustic events are the muzzle blast (bang) and the ballistic shockwave (crack). The acoustic event characteristics depend on their source generating mechanisms and vary according to the firearm make, model, barrel length, and the specific ammunition characteristics. Forensic gunshot analysis deals with a single recorded shot lasting for a fraction of a second. These acoustic events are usually high intensity, often up to 160 dB SPL, are highly directional, and are often recorded in high distortion environments. Forensic gunshot analysis must take into account variations in the source generation characteristics and the sources of distortion for these recorded acoustic events in order to answer these fundamental forensic questions: Is this event a gunshot? Are two events from the same firearm? Who fired first? To illustrate the complex nature of the analysis, we present the gunshot data collected in a pristine controlled environment and the data collected in a forensic environment.

10:30

4aSCa4. Comparison of human-supervised and fully automatic formant-trajectory measurement for forensic voice comparison. Cuiling Zhang, Felipe Ochoa, Ewald Enzinger, and Geoffrey Stewart Morrison (Forensic Voice Comparison Lab., School of Elec. Eng. (Telecom., Univ. of New South Wales, Sydney, New South Wales, Australia; Dept. of Forensic Sci.) Technol., China Criminal Police Univ., Shenyang, China)

Acoustic–phonetic approaches to forensic voice comparison often include analysis of vowel formants. Such methods depend on human-supervised formant extraction, which is often assumed to be reliable and relatively robust to transmission-channel effects, but requires substantial investment of human labor. Fully automatic formant trackers require less human labor but are usually not considered reliable. This study assesses the variability within and between four human experts and compares the results of human-supervised formant measurement with several fully automatic procedures, both on studio-quality recordings and transmission-channel degraded recordings. Measurements are made of the formant trajectories of /iau/ tokens in a database of recordings of 60 female speakers of Chinese. As well as directly comparing the formant-measurements results, the formant measurements are also used as input to likelihood-ratio forensic-voice-comparison systems, and the validity and reliability of each system is empirically assessed.

10:45

4aSCa5. Nasal spectra for forensic voice comparison. Ewald Enzinger (Forensic Voice Comparison Lab., School of Elec. Eng., Telecom., Univ. of New South Wales, Sydney, New South Wales, Australia) and Cuiling Zhang (Univ. of New South Wales, Sydney, New South Wales, Australia)

For features to be effective in forensic voice comparison, they must have relatively low within-speaker variability and relatively high between-speaker variability. An understudied source of features, which potentially meets these criteria is the acoustic spectrum of nasals. Nasals spectra contain poles and zeros dependent upon nasal cavities. The latter are complex static structures which vary from person to person. Theoretically, nasal spectra may therefore have low within-speaker and high between-speaker variabilities. This study evaluates different methods for extracting spectral features (e.g., pole-zero models, all-pole models, and cepstra) and using them as part of a likelihood-ratio forensic-voice-comparison system. The validity and reliability of each system is empirically evaluated using /m/ and /n/ token extracted from a database of voice recordings of 60 female speakers of Chinese.

11:00

4aSCa6. Intra- and inter-speaker variability in duration and spectral properties of English /s/. Colleen Kavanagh (Dept. of Lang. and Linguistic Sci. Univ. of York, Heslington, York YO10 5DD, United Kingdom, cd519@york.ac.uk)

This study investigates the speaker-specificity of acoustic characteristics of the English fricative /s/ and contributes background population statistics for use in forensic speaker comparison work. The intra- and inter-speaker variability in duration and spectral properties of /s/ was investigated in data from 30 young adult male speakers of Cambridge and Leeds English. Read speech was used in the present study to allow direct comparison across speakers. Segment duration was normalized for speaking rate. Spectra were filtered at 4 kHz in order to explore speaker discrimination performance at settings mimicking the bandpass filter effect of telephone transmission. Additional filters were applied at 8, 16, and 22.05 kHz to investigate discrimination with data from various frequency ranges. Spectral measures were calculated from a 40-ms window centered on the midpoint of each token. Although mean values display relatively little inter-speaker variation, the individuals at the extreme high and low ends of the distributions may be the best discriminated, particularly those at the extremes on more than one parameter. Discriminant analyses were conducted to determine the most speaker-specific predictors; relative performance was compared across the four filter conditions. The discriminatory ability of these parameters will also be presented using a likelihood ratio framework.

11:15

4aSCa7. Collecting population statistics: The discriminant power of clicks. Erica Gold (Dept. of Lang. and Linguistic Sci., Univ. of York, Heslington, York Y010 5DD, United Kingdom)

This research gathers population statistics on clicks for use in likelihood ratios (LRs). As reported in Gold and French (2011), clicks have been analyzed by 57% of experts in forensic speaker comparison cases and 18% of experts find them to be useful speaker discriminants. Eight minutes of speech from 100 male speakers of Southern Standard British English were analyzed from the DyVis Database, using categorical annotations of clicks (Wright, 2007). The distribution of click use in subjects is highly skewed with a large majority not clicking. However, the distribution of clicks is highly variable with *non-clickers* ranging from 25–44% of the population depending on the length of the speech sample. The same 100 speakers were also analyzed for click use when speaking with two additional interlocutors. Again the results are highly variable, which suggests the intra- and inter-speaker instability of clicks, the lack of overall robustness, and the accommodation of clicks in speech. This study serves as a beginning point in incorporating previously unreported population statistics into LRs, and specifically examining the potential of including higher order and paralinguistic features in a Bayesian framework. [Research funded by the European Community's Seventh Framework Program (FP7/2007-2013) under grant agreement #238803].

Session 4aSCb

Speech Communication: Acoustics of Speech Production and Acoustic Signal Processing (Poster Session)

Sun-Ah Jun, Chair
Dept. of Linguistics, Univ. of California, 405 Hilgard Ave., Los Angeles, CA 90095-1543

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

Contributed Papers

4aSCb1. Short and long diphthongs in Hainan Cham. Ela Thurgood (Dept. of English, California State Univ., Taylor Hall 209, Chico, CA 95929-0830, ethurgood@csuchico.edu)

This paper focuses on the acoustic features in the contrast between two pairs of phonemically distinctive diphthongs in Hainan Cham, /ai/ versus /a:i/ and /au/ versus /a:u/. The data from six native speakers of Hainan Cham show that the overall duration of long diphthongs does not statistically differ from the overall duration of short diphthongs. The differences, instead, lie in the durational differences between onsets and offsets. In Hainan Cham, the long diphthong onsets are longer than the short diphthong onsets, while their offsets are shorter. The transition duration does not differentiate short and long diphthongs. In Hainan Cham, it occupies only 25–34% of the duration of the whole diphthong, short or long. Another acoustic feature examined in diphthongs is the range and rate of F2 change. In Hainan Cham, /ai/ is well differentiated from /a:i/: The short diphthong /ai/ has a greater F2 range of change and a faster F2 rate of change than the long diphthong /a:i/. However, /au/ and /a:u/ have very similar F2 rate of change.

4aSCb2. The Canadian shift in Ontario: Transmission and diffusion. Rebecca V. Roeder (Dept. of English, Univ. of North Carolina at Charlotte, 9201 Univ. City Blvd., Charlotte, NC 28223, rroeder@uncc.edu)

The Canadian Shift, a sound change in progress that is affecting the front lax vowels of Canadian English, was initially characterized as a chain shift. However, recent studies have observed a different pattern of movement over apparent time, requiring an alternative theoretical model to explain the change. Relying on instrumental analysis of data from nearly 100 speakers across five Ontario cities and towns, this paper provides additional observations of parallel shifting across apparent time in these vowels and adopts vowel dispersion theory as a theoretical framework for positing phonetic movement toward a system that is balanced both phonetically and phonologically. This phonetically-based model has primarily been used to explain the relationship between phonological inventories and acoustic space, but the generalizations and principles that are emerging through such research are also useful in the interpretation of observations regarding sound change in progress. Findings also indicate that the geographical progression of the shift corresponds with a cascade or gravity model of diffusion.

4aSCb3. Stimulus and rate effects on central vowels in young adult and aged adult speakers. Phoebe Natzke and Marios Fourakis (Dept. of Com Dis, Univ. of Wisconsin, 1975 Willow Dr., Madison, WI 53706)

The central vowels caret and schwa occur frequently and contribute significantly to the rhythmic (prosodic) characteristics of English (Umeda, J. Acoustic. Soc. Am., 1975). Appropriate prosody contributes to the intelligibility of speech (Klopfenstein, Int. J. Speech Lang. Path., 2009), and there is evidence that it is disturbed in the speech produced by aged adults (Benjamin, Lang. Comm., 1986). The present study explores the effects of speech rate and speech material on these central vowels in the aged population, as compared to a group of young adult speakers. In addition, it contributes to the relatively scarce published research on the acoustics of speech produced by geriatric adults (Zraick *et al*., J. Med. Speech Lang. Path., 2005). The study reports temporal and spectral data of the vowels caret and schwa occurring in a variety of words produced at three different speech rates. Speakers included 10 healthy young speakers and 10 healthy geriatric speakers, with five men and five women in each group. [Work supported by NIH T32 DC 005359.]

4aSCb4. Dispersion and variability of vowels of different inventory sizes. Wai-Sum Lee and Eric Zee (Dept. of CTL, City Univ. of Hong Kong, 83 Tat Chee Ave., Kowloon Tong, Hong Kong, w.s.lee@cityu.edu.hk)

This study investigates dispersion and variability in the vowels of systems with different phonemic inventory sizes in three Chinese dialects. Yongding Hakka, Hong Kong Cantonese, and Wenling Wu have three, seven, and eleven vowel phonemes, respectively. Measurements of formant frequencies were obtained through a spectral analysis of speech data from twenty male and twenty female speakers of each dialect. Results show that the size of acoustical vowel space in the F1F2 plane increases with an increase in vowel inventory size. The finding supports the vowel dispersion theory (Lindblom, 1968; Liljencrants and Lindblom, 1972) which claims that the larger the inventory is, the more expanded the vowel space will be. However, the prediction by the theory that variability in vowel formants is inversely related to the vowel inventory size is not supported by the vowel formant data from the three Chinese dialects. In Fant (1966, 1975), female vowels are said to exhibit greater between-category dispersion in the F1F2 plane than male vowels. The observation is supported by the vowel formant data from the three Chinese dialects. However, this is true before vowel normalization, but not after. Lastly, no gender-related patterning of vowel dispersion is observed in the three Chinese dialects.

4aSCb5. Voice onset time production in Singapore English. Priscilla Z. Liu (2981 N Cardell Cir., Tucson, AZ 85712)

This current study investigates the effects of linguistic accommodation through the production of voice onset time (VOT). Six Singaporeans were recorded in three separate conversations that differed by interlocutors: 1) another Singaporean, the researcher, and a non-Singaporean. The two Singapore English dialects, Standard Singapore English (SSE), and Singlish, are investigated. SSE is influenced by British English while Singlish draws from a myriad of languages, “mother tongues” that are spoken in Singapore: Mandarin Chinese, Hokkien, Tamil, and Malay. Thus, Singaporeans were expected to shift production of VOT as motivated by linguistic accommodation to their interlocutors. Previous research indicates that VOT is a valuable acoustic demarcation of phonetic and phonological boundaries. Thus, speakers were found to significantly shift their production of VOT for /p t k/ towards phonetic categories of British English or Malay and Tamil dependent on their audience. (Though the trends for /b d g/ were not significant, the numerical data also shift in the same direction.) Lastly, the paper discusses

the implications of these results, that although SSE and Singlish are not entirely discrete and both exist as heavily used dialects of English in Singapore, speakers manipulate VOT in order to accommodate their listeners.

4aSCb6. Effects of consonant context on vowel formant contours in spontaneous and read speech. Michael Kiefte (Sch. Human Commun. Disord., Dalhousie Univ., 5599 Fenwick St., Halifax, N.S. B3H 1R2 Canada, mkiefte@dal.ca) and Terrance M. Nearey (Univ. of AB, Edmonton AB TG6 2E7, Canada)

A database of recordings from 163 speakers from Nova Scotia, Canada was collected with the aim of comparing formant contours between spontaneous and read speech. In the reading task, participants were asked to produce a number of real and nonsense words spanning the inventory of vowels in this dialect in a variety of consonant contexts. In the second part of the reading task, speakers were asked to read 20 sentences from the TIMIT database. These latter recordings were used to assist in the training of the force-alignment system which used to segment the recordings into phonemes from a text transcript. In addition to the reading task, speakers also provided a monologue on a topic of their choice. These recordings were screened for disfluencies, noise, and disruptions, manually segmented into breathgroups, and then transcribed. Stressed vowels in /CVC/ contexts, where C corresponds to plosives, were sampled from both the read and spontaneous speech. Formants were tracked and measured automatically, and an analysis similar to that of Broad and Clermont [J. Acoust. Soc. Am., **81**, 155–165 1987] was performed in which consonant effects on formant transitions are treated as additive effects. [Work supported by SSHRC.]

4aSCb7. Effects of speaking rate, sentential position, and coda voicing on formant frequency. Keelan Evanini (Educational Testing Service, Rosedale Rd. MS-R11, Princeton, NJ 08541, kevanini@ets.org) and Eon-Suk Ko (Univ. at Buffalo, Buffalo, NY 14260)

This study examines the effects of speaking rate, sentential position, and coda voicing on formant frequency values in English. Some previous studies have found gestural undershoot for formant target values in words with shorter durations, e.g., Lindblom (1963), although other studies have shown little to no effect of duration, e.g., Gay (1978), and the effects of the other factors are less-studied. This study examines formant frequencies of three different vowels (/i/, /e/, and /ae/) in CVC words containing both voiced and voiceless codas produced in three different sentence positions (initial, medial, and final) and three different speaking rates (slow, habitual, and fast). In total, seven speakers (five female and two male) of the Northern dialect of American English produced 1295 tokens, and vowel formant measurements were extracted at 1/3 of the duration of each vowel token. Separate linear regression analyses of the three vowels for the male and female speakers show that F1 and F2 target values do not vary systematically with vowel duration. In many cases, however, sentence position and coda voicing do have significant effects: in general, F1 and F2 values are more peripheral before voiced codas and in sentence-initial position.

4aSCb8. Acoustic contrastivity in conversational and loud speech. Yunjung Kim (Dept. of Commun. Sci. and Disord., Louisiana State Univ., Baton Rouge, LA 70803)

Despite frequent clinical observations of improved speech intelligibility following high vocal intensity training in speakers with dysarthria, the mechanism by which loud speech results in increased speech intelligibility is little understood. Prior research has reported conflicting acoustic results of articulatory modifications in loud speech conditions such as changes in vowel durations, acoustic vowel space, and F2 transition duration/extent. More interestingly, the impact of an overall increase in amplitude of speech signals on perceptual judgment of speech intelligibility is unclear, especially when the entire speech signal is amplified as compared to the amplification of selected phonetic events (see Kim and Kuo, in press). This presentation focuses on the change of *relative* contrastivity within utterances that were produced at both conversational and loud levels to better understand the underlying mechanism of enhanced speech intelligibility secondary to greater vocal intensity. In this presentation, data on the ratio of vowel durations (long versus short), formant structures of vowels (tense versus lax), as well as the ratio of syllable intensity (stressed vs unstressed) will be compared between conversational and loud speech conditions produced by young adult speakers.

4aSCb9. Timing differences in read speech and spontaneous conversation: English, Japanese, Korean, and Mandarin. Dan Brenner (Univ. of Arizona, Dept. of Linguist., Douglass Bldg. 200E, Tucson, AZ 85721, dbrenner@email.arizona.edu)

Timing and rhythm in language reflect broad auditory properties of languages to which listeners acclimate very early on in acquisition (Nassi *et al.*, 1998, 2000), and are heavily implicated in the differentiation of speech segmentation strategies cross-linguistically (Cutler and Butterfield, 1992; Otake *et al.*, 1993; Saffran *et al.*, 1996; Kim *et al.*, 2008). The rhythmic properties of everyday conversational speech (as compared to read speech or motherese), however, are not well understood. The present work employs several measures developed to summarize rhythmic differences such as %V vs. δC (Ramus *et al.*, 1999) and C-rPVI versus V-nPVI (Grabe and Low, 2002) in order to study the timing variation found in English, Japanese, Korean, and Mandarin (unrelated languages varying in purported "rhythm class") comparing rhythmic measures during careful read speech and in spontaneous casual conversation. This reveals the effect of highly variable conversational speech on the timing behavior of rhythmically diverse languages.

4aSCb10. Phonetic imitation in contexts of stimulus-directed and non-stimulus-directed attention. Jennifer Abel, Molly Babel, and Alexis Black (Dept. of Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada)

Research suggests that phonetic imitation is an automatic and subconscious process, but it is clearly a behavior that is variable across participants and conditions. This experiment explores how a participant's amount and type of attention to the speech signal moderates their amount and type of imitation. Six paired conditions varied the activities of participants (all native speakers of English) while listening to the model talker (a female speaker of North American English): blocked exposure (no instructions for listening block) vs immediate shadowing; math task vs picture-drawing task; and word-memorization task vs talker-description task. In all conditions, participants' baseline productions of the stimuli list were recorded prior to exposure to the model talker. Early analyses of whole word duration suggest participants are more likely to imitate the model when their attention is not directed toward the stimuli (i.e., no particular redirection or redirection through a picture-drawing task) than when their attention is stimulus-directed (i.e., being instructed to memorize the words or describe the talker). Further analyses of other acoustic parameters which may reveal imitative behavior are currently underway and include measures of vowel duration, vowel spectra, and f0.

4aSCb11. Visual speech influences on interactive speech alignment. James W. Dias, Therasa Cook, and Lawrence D. Rosenblum (Dept. of Psych., Univ. of California, 900 University Ave., Riverside, CA 92521)

Speech alignment describes the unconscious tendency of humans to produce speech that shares the acoustical characteristics of a perceived speech signal. Participants have been found to align more when interacting in full view of their partner than when conversing only through sound [J. W. Dias and L. D. Rosenblum, J. Acoust. Soc. Am. **128**, 2458 (2010)]. However, this previous study did not address what specific visual information enhances speech alignment. The present study evaluates whether being able to see visual *speech* gestures enhances alignment in a conversational setting. Pairs of participants performed an interactive search task, for which they were required to utter nine key words multiple times. Participants performed the task, while interacting face-to-face, or with visibility of the mouth and throat occluded using a small sound-permeable screen. Participants' utterances of the key words were recorded before, during, and after the interaction. Alignment was evaluated by naïve raters in an AXB matching task. In a second experiment, participants performed the task in a background of talker-babble noise. Preliminary results indicate that alignment increases with visibility of the mouth. The findings are consistent with evidence that alignment occurs to amodal, gestural properties of perceived speech.

4aSCb12. Examining the voice bar. Sean A. Fulop (Dept. of Linguist. PB92, Calif. State Univ., 5245 N Backer Ave., Fresno, CA 93740, sfulop@csufresno.edu) and Sandra Disner (Dept. of Linguist., Univ. of Southern Calif., Los Angeles, CA 90089)

In a spectrogram of a human vowel sound, it is possible to observe the formant resonances which define the vowel auditorily. It is usually also possible to observe an emphasized frequency below F1, which has often been called the *voice bar*. Although recognition of the voice bar dates back to 19th century phonetics, it has never been the subject of a specific investigation. As a result, the nature and origin of the voice bar remain mysterious. Recent work on voice source synthesis [C. d'Alessandro *et al.*, *eNTERFACE 2005 Proceedings*, pp. 52–61] has explained the appearance of an emphasized frequency in the neighborhood of 200 Hz–simply, it results from the frequency peak of the radiated source spectrum. Yet many speech scientists continue to ignore the voice bar, even to the point of denying its reality. Measurements of the voice bar in a number of different speakers and languages will be clearly shown in this paper, using reassigned spectrograms and linear prediction spectral estimates. The voice bar has confused phoneticians for over a century, and has often confounded efforts to measure F1. The proper recognition of the voice bar can begin with this preliminary study.

4aSCb13. Sub-phonemic correlates of gender and regional identity in California. Grant L. McGuire (Dept. of Linguist., UC Santa Cruz, 1156 High St. Santa Cruz, CA 95064, gmcguir1@ucsc.edu), Angela R. Aiello (Dept. of Communicative Disord., San Jose St. U., 1 Washington St., San Jose, CA 95192), Jackie L. De Leon, Tariq El-Gabalawy, Lauren Negrete, and Kasondra Vanpykeren-Gerth (Dept. of Linguist., UC Santa Cruz, 1156 High St. Santa Cruz, CA 95064)

Given that the Northern and Southern California have large metropolitan areas geographically and culturally separated from each other, it is to be expected that each is developing a unique linguistic identity. Despite a handful of ethnographic studies showing otherwise (e.g., Hall-Lew 2009), the West has generally been lumped into a single dialect region (Labov *et al.* 2006). This paper presents data showing sub-phonemic differences between the regions that break along gender lines. Vowel productions from 14 (female = 8) Northern Californians (NCs) and 15 (female = 8) Southern Californians (SCs) were analyzed for regional differences in normalized vowel quality, voice quality (spectral tilt), pitch, and duration. No major differences in vowel quality were found. However, interactions were found between region and gender for duration and voice quality. Specifically, NC females had significantly longer word durations than NC males, with no difference between genders for SC. For voice quality, H1-H2 and H1-A3 measures both demonstrated significant differences between males and females for SC, with female voices being breathier, but with no differences for NC. Currently, a perception experiment is underway to determine if listeners can use these differences to categorize voices by region.

4aSCb14. Liquids as syllable peaks: Preconsonantal laterals in closed syllables of American English. Onna A. Nelson (Dept. of Linguist., UC Santa Barbara, South Hall, Santa Barbara, CA 93106, oanleson@umail.ucsb.edu)

Liquids occur in all syllable positions in English and may behave as syllable peaks or nuclei (Proctor, 2009). Previous work has examined syllabic liquids in open syllables like *little* and *doctor*, in the onset of closed syllables like *prayed* and *played* (Price, 1980) and has established that rhotics are syllabic in certain closed syllables like *bird*, *church*, and *learn*. However, little work has investigated whether laterals can serve as syllable peaks in preconsonantal position. This study examines potential syllabic liquids in closed and open syllables in the Santa Barbara Corpus of Spoken American English (Du Bois *et al.*, 2000, 2003, 2004, 2005), focusing on CVlC syllables, such as *bulk*, *filled*, and *help*, which are structurally similar to the aforementioned contexts containing syllabic rhotics. Vowel and lateral duration and intensity are measured to determine whether these laterals display properties associated with syllabicity (Price, 1980). Additionally, the first and second formants of the vowel and lateral are measured at 10 ms intervals to examine the vowel-like behavior of the liquids (Gick, 2002). Further influencing factors are considered, including morphology, surrounding vowel quality, place and manner of surrounding consonants, intonation, and other prosodic elements to determine the environments in which lateral syllabicity occurs.

4aSCb15. The production of Spanish–English code-switching. Page E. Piccinini (Dept. of Linguist., Univ. of California San Diego, 9500 Gilman D., La Jolla, CA 92093-0108)

It is generally assumed that in code-switching (CS) switches between two languages are categorical, however, recent research suggests that the phonologies involved in CS are merged and bilinguals must actively suppress one language when encoding in the other. Thus, it was hypothesized that CS does not take place abruptly but that cues before the point of language change are also present. This hypothesis is tested with a corpus of Spanish-English CS examining word-initial voiceless stop VOT and the vowel in the discourse marker "like." English VOTs at CS boundaries were shorter, or more "Spanish-like," than in monolingual utterances. Preliminary results suggest Spanish VOTs at CS boundaries were shorter than in monolingual utterances, thus even more Spanish-like than monolingual Spanish utterances. The vowel of "like" in English utterances was more monophthongal and had a lower final F2 as compared to "like" in Spanish utterances. At CS boundaries, "like" began similarly to the language preceding the token and ended similarly to the language following it. For example, in a "English-like-Spanishnn" utterance, initial F2 measurements were more English-like but final measurements more Spanish-like. These results suggest code-switching boundaries are not categorical, but an area where phonologies of both languages affect productions.

4aSCb16. Dynamic differences in the production of diphthongs by French–English bilingual children. Vincent Chanethom (Dept. of Linguist., New York Univ., 10 Washington Pl., New York, NY 10003)

This study examines the cross-linguistic phonetic interactions in the production of diphthongs by French–English bilingual children. Tautosyllabic vowel-glide combinations in English and French have different phonological statuses. This combination corresponds to a single segment (i.e., a diphthong) in English, but two separate segments (i.e., vowel+glide) in French. Using a picture-naming experiment, the study aims to investigate (1) whether English diphthongs (e.g., /aɪ/ as in bye) and French tautosyllabic vowel-glide combinations (e.g. /aj/ as in baille 'yawn') have different phonetic implementations and, if so, (2) whether bilingual children maintain two separate categories. Diphthongs were recorded for six monolingual speakers of French and English, and four 6-7 year-old bilingual French-English speakers living in the US. To best capture the dynamic properties of diphthongs, the curves corresponding to F1 and F2 trajectories were submitted to statistical comparisons using the Smoothing Spline ANOVA. The cross-linguistic comparisons from the adult monolingual data indicate distinct phonetic properties for the two categories. The child bilingual data, on the other hand, show variation from one child to another as a function of amount of input. Children who attend English-only schools show greater degrees of language interference than those who attend French–English bilingual schools.

4aSCb17. Vowel undershoot in production of English tense and lax vowels by Mandarin and American speakers. Chung-Lin Yang (Dept. of Linguist., Indiana Univ., Bloomington, IN 47405, cy1@indiana.edu)

Vowel undershoot (Lindblom, 1963), an effect where the articulatory gesture fails to reach the target due to the following contrary gesture (de Jong, 2004), is found to be particularly prominent (lowered F1 and shortened duration) in polysyllables (Lindblom, 1968; Moon and Lindblom, 1989). The current study investigates Mandarin and American speakers' production of English tense-lax vowels /i/-/ɪ/ and /eɪ/-/ɛ/, and examines undershoot in monosyllabic and disyllabic words. Three Mandarin and three American speakers were recorded. Fifteen target vowels were embedded in a voiced stop-V-voiceless stop context. All disyllabic words were first-syllable stressed. Carrier sentences were of variable length to create a natural-speech-like context. F1, F2, vowel duration and utterance duration were measured. The results show that Americans did show a significant distinction between /i/-/ɪ/ but tended to merge the formants of /eɪ/-/ɛ/ in disyllabic words, which was partly due to the coarticulation with the following syllable. They also demonstrated undershoot effect in /eɪ/-/ɛ/ but not much in /i/-/ɪ/. Mandarin speakers, however, could not make a significant tense-lax distinction, and showed formant undershoot in disyllabic words, especially tense vowels. One possible account for this effect is the influence from L1. The issue of Mandarin vowel inventory is discussed.

4aSCb18. Clear speech production by nonnative English speakers. Jenna Silver Luque and Ann R. Bradlow (Dept. of Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, SLPJenna@gmail.com)

Clear speech is an intelligibility-enhancing mode of communication often used when speakers have trouble being understood. Previous work has

established that both native and non-native listeners can receive a clear speech perception benefit, though possibly to differing degrees (Bradlow and Bent, 2002). Few studies have looked at whether nonnative talkers can induce this clear speech benefit (e.g., Smiljanic and Bradlow, 2007; Rogers *et al.*, 2010). The current study examined English clear and conversational speech by nonnative speakers from three language backgrounds (Japanese, Portuguese, and Turkish) and two proficiency levels to determine their effect on the inducement of a clear speech benefit. Native English listeners repeated back semantically anomalous sentences. The signal to noise ratio was adjusted to the level at which they could correctly repeat 50% of the words using an adaptive test similar to the Hearing in Noise Test (Nilsson *et al.*, 1993). The results suggest that the speaker's native language may play a role in the size of the induced clear speech benefit independently of proficiency level. Additionally, accent ratings indicated a dissociation of intelligibility and accentedness. These results are consistent with the notion that variability in intelligibility is subject to language-specific knowledge by both the talker and the listener.

4aSCb19. Voice quality and pitch contrast in non-native Korean. Seung-Eun Chang (Dept. of East Asian Lang. and Cultures, Univ. of California, Berkeley, 3413 Dwinelle, Berkeley, CA 94720, sechang71@berkeley.edu)

In this study, I compare the effects of linguistic experience on voice quality (H1-H2) and fundamental frequency (f0) in Korean stops among native and non-native Korean speakers. Native speakers of Chinese, English, Korean, and Spanish produced Korean words in a /CVC/ context, and H1-H2 and f0 of the initial stops in each set of materials were measured. Korean and Chinese speakers showed creakiness (smaller H1-H2) for Korean tense stops and breathiness (larger H1-H2) for lenis and aspirated stops, whereas English and Spanish speakers showed relatively larger H1-H2 for all stops. For f0 values, Korean and Chinese speakers displayed a lower f0 for lenis, an intermediate f0 for tense, and a higher f0 for aspirated stops. For English speakers, however, lenis and tense stops were merged in the lower f0 region, and aspirated stops showed a higher f0. In Spanish speakers, tense and aspirated stops merged in the higher f0 region, and lenis stops showed a lower f0. These results demonstrate a strong effect of linguistic experience on voice quality and f0: speakers of Chinese were more accurate in replicating Korean stops than were speakers of English or Spanish, languages that lack phonemic voice quality and tone contrasts.

4aSCb20. Acoustic features of English sentences produced by native and non-native speakers. Yu-Fu Chen, Chang Liu, and Su-Hyun Jin (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX 78712)

Fundamental frequency (f0) contours and power envelopes of English sentences produced by young English-, Chinese-, and Korean-native speakers were measured. Sentences were selected from Hearing in Noise Test (HINT) and recorded from non-native speakers whose US residency within five years. Preliminary results showed that the f0 contours of non-native speakers were similar to those of native speakers, but with greater variation over the time, suggesting that non-native speakers may be able to follow native speakers' vocal pitch contour, but with higher temporal variability. Power envelopes of the sentences will be measured and the relationship between the acoustic features of speech and speech intelligibility will be discussed as well.

4aSCb21. Native English speakers learning German as a second language: Devoicing of word-final voiced stop targets. Bruce L. Smith and Elizabeth A. Peterson (Dept. of Commun. Sci. and Disord., Univ. of Utah, 390 S. 1530 E., Salt Lake City, UT 84107, bruce.smith@hsc.utah.edu)

In contrast to German and other languages that devoice underlying word-final, voiced obstruent targets, English has a surface contrast between voiced and voiceless obstruents. The present study investigated the issue of what occurs when native speakers of American English, in an early stage of learning German as a second language, produce word-final voiced and voiceless stop targets in German versus English. The fact that the underlying voicing contrast in German is reflected orthographically (e.g., "Tod" versus "tot") could make it difficult for native speakers of English to learn to devoice German word-final, voiced targets. The findings indicate that many of the 12 native English learners of German who were studied showed a tendency toward devoicing voiced targets in German relative to their productions of orthographically-similar words in English (e.g., "toad" and "tote"). In general, their partial devoicing in German (relative to their English productions) occurred due to a combination of producing somewhat shorter vowels before voiced consonant targets, reducing glottal pulsing during the closure of voiced consonant targets and/or shortening voiceless consonant closure durations. Subjects who produced more characteristically "voiced" consonants when speaking English (e.g., longer preceding vowel durations, etc.) tended to devoice German final stops to a lesser extent.

4aSCb22. Temporal characteristics of child-directed speech. Eon-Suk Ko (Dept. of Linguist., and Dept. of Communicative Disord. and Sci., Univ. at Buffalo, Buffalo, NY 14260, eonsukko@buffalo.edu) and Melanie Soderstrom (Univ. of Manitoba, Winnipeg, MB R3T 2N2)

Child-directed speech (CDS) is produced with a slower tempo compared to adult-directed speech (ADS). Yet the characterization of CDS as simply slowly spoken speech masks a number of underlying subtleties. We investigated temporal characteristics of CDS as a function of speech register based on a highly controlled set of elicited data. six sentence forms containing five monosyllabic words were read several times in declarative and question intonation with three focus conditions in CDS and ADS. After an evaluation of the data through a perceptual rating task, 2301 sentences produced by one mother and five theater students were segmented at the word-level using forced-alignment tools (Yuan and Liberman, 2008). We found strong effects of the CDS register on duration across the entire sentence. Additionally, elongation in CDS applied even to the syllables without an explicit focal accent and to function words. Our data also demonstrated a highly consistent ratio of the final syllable to the sentence duration both in CDS and ADS across all subjects. These results suggest that the slow speaking rate in CDS cannot be attributed to any single effect such as the exaggerated utterance-final lengthening (Church *et al.*, 2005) or effects of lexical categories (Swanson *et al.*, 1992).

4aSCb23. Developmental courses of infants' articulations estimated by acoustic-to-articulatory inversions. Hiroki Oohashi, Hama Watanabe, and Gentaro Taga (Graduate school of Education, The Univ. of Tokyo, Tokyo 113-0033 Japan)

Knowledge about the actual movements of articulators and their developmental changes in early childhood is crucial to a better understanding of the relationship between speech productions and perceptions. In this study, we attempted to recover the shapes of articulators from the acoustic properties of infants' vocalizations by using a variable linear articulatory model (Boe, 1999). We randomly selected 30 samples of 5 Japanese vowels at three age groups (8, 24, and 50 months) from the NTT Japanese infants speech database (Amano *et al.*, 2002) and those pronounced by adults. We modeled shapes of articulators and size of the vocal tract according to Maeda (1990) and synthesized vowels by forward transformation from the vocal tract cross-sectional area function to its acoustics. We performed inverse estimations of articulatory parameters from acoustic properties by using a pseudo-inverse of the Jacobian matrix. Statistical analysis of the estimated articulatory parameters revealed that the shapes of articulators during vowel pronunciations became closer to those of adults over developmental courses. Developmental courses of some articulators represent the u-shaped curve (e.g., tongue shape and tongue tip of /a/) and those of others represent the linear curve (e.g., tongue position of /o/).

4aSCb24. The acquisition of voiceless sibilant fricatives in children speaking Mandarin Chinese. Fangfang Li (Dept. of Psych., Univ. of Lethbridge, 4401 Univ. Dr. Lethbridge, AB, Canada, T1K 3M4)

The current study aims to describe Mandarin-speaking children's acquisition of voiceless sibilant fricatives, /s/, /ʂ/, and /ɕ/, as assessed by acoustics. Forty children, aged 2–5, participated in a word repetition task. The stimuli were fricative-initial words that are familiar to children. Children's speech sound productions were recorded and analyzed spectrally. Two acoustic parameters were obtained: the centroid frequency calculated over the middle 40-ms slice of the fricative noise spectrum and onset F2 frequency, the second formant frequency taken at the onset of the vowel following target fricatives. Centroid frequency indexes *where* the major lingual constriction is made in the oral cavity and is inversely related to the length of the front resonating cavity during the articulation of voiceless sibilant

fricatives. Onset F2 frequency indexes *how* the major constriction is made and is sensitive to the lingual posture during constriction. These two parameters have been demonstrated to capture adults' fricative distinctions successfully. The results indicated an early separation between /ʂ/ and the other two fricatives in the centroid dimension, and an early separation between /ɕ/ and other two fricatives in the onset F2 dimension, The results suggested that children gradually implement their motor control in different articulation/acoustic dimensions.

4aSCb25. The development of neutral tone in Mandarin-speaking children. Jie Yang and Barbara Davis (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, 2504 Whitis Ave., Austin, TX 78712, Babs@mail.utexas.edu)

Besides the four citation tones in Mandarin stressed syllables, neutral tone usually occurs in unstressed syllables. The fundamental frequency (F0) contour and height of neutral tone are determined by the preceding tones. Neutral tone was also considered to have lower intensity and shorter duration compared to citation tones (Cao, 1986; Chao, 1968). Li and Thompson (1977) suggested that neutral tone was not fully acquired by Mandarin-speaking children of age 2. However, the acoustic characteristics of neutral tone in the extended period of acquisition were not explored. The present study compared acoustic realizations of neutral tone in children's production with adults. Eight 5-yr-old and eight 8-yr-old mono-lingual Mandarin-speaking children and young adults participated. Bi-syllabic target words containing neutral tone in the second syllable was elicited by picture-naming tasks. F0, duration and intensity of neutral tone syllables were measured. The ratio of these acoustic parameters between the first and second syllable was calculated. Results indicated that 5-yr-old children started to produce F0 contour and height of neutral tone according to the preceding tone.

4aSCb26. Lexical effects in the production of emotional speech. Tatiana Kryuchkova and Benjamin V. Tucker (Dept. of Linguist., Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB, T6G 2E7, Canada, tatiana.kryuchkova@ualberta.ca)

Emotion in speech production has been shown to correlate well with fundamental frequency (F0), intensity, and duration [B. Zupan *et al*., J. Commun. Disord., **42**, 1–17 (2009)]. Studies in non-emotional speech have shown effects of lexical predictors on speech production (e.g., neighborhood density [B. Munson and N. P. Solomon. J. Speech, Lang. Hear. Res., **47**, 1048–1058 (2004)], and lexical frequency [R. H. Baayen *et al*., in *Proceedings of the Annual Meeting of the Chicago Linguistic Society* (2007), Vol. 43, pp. 1–29]. In the present study, we investigate the effects of lexical predictors in the production of emotional speech. Two professional male actors recorded 260 isolated words. Three emotional states were analyzed: neutral, anger, and joy. Measures of word frequency, morphological family size, number of synonyms and homophones, and ratings for danger and usefulness [L. Wurm, Psychon. Bull. Rev., **14**, 1218–1225 (2007)] were used as statistical predictors in modeling, using linear mixed-effects regression. Following previous literature, both speakers use F0, intensity, and duration to portray emotion. Lexical predictors such as word frequency, morphological family size, and danger ratings were found to significantly predict mean F0, mean intensity, and word duration across emotional types for both speakers.

4aSCb27. Executive abilities for spoken-word commands: Inhibiting conflicting responses in voice-tone classification by adolescents and adults. Blas Espinoza-Varas (Commun. Sci. Disord., OU Health Sci. Ctr., 1200 N. Stonewall Av., Oklahoma City, OK 73117), Hyunsook Jang (Hallym Univ., Chuncheon, South Korea), Caleb Lack (Univ. Central Oklahoma, Edmond, OK 73034), and Beatriz Luna (Western Psychiatric Inst. Clinic, Univ. Pittsburgh Medical Ctr., Pittsburgh, PA 15213)

The inhibition of responses prompted by conflicting laterality cues and response-mapping rules was studied while participants classified the voice-tone of word commands. Announced by the cue words /left/ or /right/, target word commands for impeding or instigating actions (e.g., /quit/ or /go/), spoken in stern or lenient tone, stimulated the left or right ear. Each trial presented a cue followed by a target (e.g., left-quit), and participants classified the target voice tone as lenient or stern with a left or right (or the reverse) response for impeding (or instigating) commands. Within a trial block, laterality-conflict conditions presented impeding or instigating words only and, on each trial, the cue and ear side were congruent or in conflict with the correct response side. Response-mapping conflict conditions presented impeding and instigating words within each trial block, and correct classification required adhering to the respective mapping. Without conflict, error percent was larger with instigating than impeding commands, and in adolescents than adults. With instigating commands, laterality conflict increased the errors in both groups but only in adolescents with impeding commands. Errors increased further with laterality and mapping conflict, especially in adolescents. The adolescent response inhibition is inferior to the adult. [Funded by ABMRF.]

4aSCb28. Normalized recognition of speech and audio events. Mark A. Hasegawa-Johnson, Jui-Ting Huang, Sarah King, and Xi Zhou (ECE Dept., Univ. of Illinois. Urbana, IL 61801)

An invariant feature is a nonlinear projection whose output shows less intra-class variability than its input. In machine learning, invariant features may be given *a priori*, on the basis of scientific knowledge, or they may be learned using feature selection algorithms. In the task of acoustic feature extraction for automatic speech recognition, for example, a candidate for *a priori* invariance is provided by the theory of phonological distinctive features, which specifies that any given distinctive feature should correspond to a fixed acoustic correlate (a fixed classification boundary between positive and negative examples), regardless of context. A learned invariance might, instead, project each phoneme into a high-dimensional Gaussian mixture supervector space, and in the high-dimensional space, learn an inter-phoneme distance metric that minimizes the distances among examples of any given phoneme. Results are available for both tasks, but it is not easy to compare them: learned invariance outperforms *a priori* invariance for some task definitions, and underperforms for other task definitions. As future work, we propose that the a priori invariance might be used to regularize a learned invariance projection.

4aSCb29. An automatic method for determining phonetic boundary for continuous speech utterances in an open source multi-language audio/video database. Montri Karnjanadecha and Stephen A. Zahorian (Dept. of Elec. and Comput. Eng., Binghamton Univ., P.O. Box 6000, Binghamton, NY 13902-6000)

Nine hundred video clips (approximately 30 h in each of English, Mandarin, and Russian) have been collected from Internet sources such as youtube.com and rutube.ru. This multi-language audio/video database has been orthographically transcribed by human listeners with time markers at the sentence level. However, the aim is to provide this database to the public with high accuracy time markers at the phonetic level, which will greatly increase the value of the database. This paper describes an approach to achieving high accuracy automatic phonetic labeling based on a Hidden Markov Model speech recognizer. This automatic method was developed due to the great length of time and tediousness of performing this task using only human listeners. One major challenge for the automatic method was that the audio data consists of spontaneous speech with unconstrained topics and the speech was spoken under various acoustic conditions. The approach begins with a well-trained acoustic model for each language. The acoustic model is then adapted to each passage and finally the phonetic labeling of the passage is determined. Comparison of the automatically determined phone time markers with those obtained by human listeners, for a subset of the speech materials, shows the accuracy of the automatic method.

4aSCb30. Spectral amplitude nonlinearities for improved noise robustness of spectral features for use in automatic speech recognition. Stephen Zahorian (Dept. of Elec. and Comput. Eng., Binghamton Univ., 4400 Vestal Parkway East, Binghamton, NY 13902, zahorian@binghamton.edu) and Brian Wong (Binghamton Univ., Binghamton, NY 13902)

Auditory models for outer periphery processing include a sigmoid shaped nonlinearity that is even more compressed than standard logarithmic scaling at very low and very high amplitudes. In some studies done at Carnegie Mellon University, it has been shown that this compressive nonlinearity is the most important aspect of the Seneff auditory model in terms of improving accuracy of automatic speech recognition in the presence of noise. However, in this previous work, the nonlinearity was trained for each frequency band of the Mel frequency cepstrum coefficients thus making it impractical to incorporate in automatic speech recognition systems. In the current study, a compressive nonlinearity is parametrically represented and constructed without training, to allow various degrees of steepness and "rounding" of corners for low and high amplitudes. Using this nonlinearity, experimental results for various noise conditions, and with mismatches in noise between training and test data, were obtained for phone recognition using the TIMIT and NTIMIT databases. The implications of the results are that a fixed compressive nonlinearity can be used to improve automatic speech recognition robustness with respect to mismatches between training and test data.

4aSCb31. Restoration of intermittent speech signal relying on auditory perceptual capability. Mitsunori Mizumachi (Kyushu Inst. of Technol., 1-1 Sensui-cho, Tobata-ku, Kitakyushu, Fukuoka 804-8550, Japan, mizumach@ecs.kyutech.ac.jp) and Toshiharu Horiuchi (KDDI R&D Labs. Inc., 2-1-15 Ohara, Fujimino, Saitama 356-8502, Japan)

Speech signals are frequently transmitted through the IP network. In such situation, a packet loss causes a temporal break in receiving speech signals. The intermittent speech signals make speech communication uncomfortable. However, the intermittent speech signal can be heard smoothly under certain conditions, because the brain reconstructs the missing speech packet unconsciously. This auditory illusory phenomenon, that is, the phonemic restoration effect, occurs under severe noisy conditions [Miller and Licklider, 1950]. In short, a speech signal with a heavy background noise, of which signal-to-noise ratio (SNR) is less than 0 dB, overcomes the intermittency in receiving speech signals, although the noisy speech signal makes us uncomfortable in speech communication. In this paper, we propose a speech signal restoration scheme with auxiliary signal processing to positively enhance our phonemic restoration capabilities under less noisy condition. First, we discuss the characteristics of the background noises. Reducing noisiness and enhancing the phonemic restoration effect should be compatible with each other for achieving comfortable speech communication. Second, we propose to predict and reconstruct the principal parts of the missing signal components from peripheral information. Finally, synergistic contribution is discussed between the above considerations. [Work partially supported by NEDO, Japan.]

4aSCb32. Acoustic evidence for protracted development of monosyllabic Mandarin tone production by Taiwanese children. Xin Yu (Dept. of Otolaryngol., The Ohio State Univ., 915 Olentangy River Rd., Columbus, Ohio 43212), Jing Yang (The Ohio State Univ., Columbus, OH 43210), and Puisan Wong (The Ohio State Univ., Columbus, OH 43212)

The current study examines the acoustic characteristics of monosyllabic Mandarin tones produced by 3-y-old children growing up in Taiwan. Four hundred monosyllabic tone productions were collected from 11 adults and 10 children and were judged by 5 Mandarin-speaking adults to determine tone accuracy. Seven acoustic parameters strongly associated with Mandarin tone perception were measured in these productions and compared among children's and adults correct and incorrect productions. The findings indicate that children do not produce the high level tone with fundamental frequencies (f0) as high or level as adults. Children's rising, falling, and dipping tones do not reach f0 ranges as low as adults. These results are largely consistent with the findings in our previous acoustic study on the monosyllabic Mandarin tones produced by 3-y-old children growing up in the U.S. Taken together, 3-y-old Mandarin-speaking children growing up in the U.S. and Taiwan do not produce adult-like monosyllabic Mandarin tones. Even children's productions in which the tonal targets are correctly perceived by adults are acoustically different than the adult forms. Children demonstrate more difficulties producing low f0 targets. The findings provide acoustic evidence to support a much more protracted process for Mandarin lexical tone acquisition than most studies have suggested.

THURSDAY MORNING, 3 NOVEMBER 2011 PACIFIC SALON 3, 8:00 TO 10:15 A.M.

Session 4aUWa

Underwater Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Characterization of Noise Radiation and Quieting Techniques for Unmanned Underwater Vehicles

Christopher Barber, Cochair
Applied Research Lab., Pennsylvania State Univ., P.O. Box 30, State College, PA 16804-0030

Jason D. Holmes, Cochair
Raytheon BBN Technologies, 10 Moulton St., Cambridge, MA 02138

Chair's Introduction—8:00

Invited Papers

8:05

4aUWa1. An international standard for the measurement of vessel underwater noise. Michael Bahtiarian (Noise Control Eng., Inc., 799 Middlesex Turnpike Billerica, MA 01821, mikeb@noise-control.com)

The development of a new international commercial standard for "Underwater Noise Measurement of Ships" started in April 2011. This effort follows on an American National Standard, ANSI/Atomic Security Agency (ASA) S12.64-2009/Part 1, Quantities and Procedures for Description and Measurement of Underwater Sound from Ships, Part 1; General Requirements. Currently, no international standards exist for performing underwater noise measurements of ships. For many years, the field of underwater noise from ships has been the exclusive specialty of the Navy. However, non-navy vessels are looking to be just as quiet so that they can perform better science. Green ships are being conceived in order to have less emission into the ocean. The goal of the project is to develop an International Organization for Standardization (ISO) standard for the measurement of underwater noise levels of ships using commercial technology. One aim is that the standard would be applicable to any open ocean site in the world and not require traveling to special acoustic test ranges. The committee' scope of work will include neither regulatory actions nor the development of any underwater noise level but will address both deep and shallow water sites. This presentation provides an update of the committee work to date and outreach to the acoustical community.

8:30

4aUWa2. Modeling noise from underwater vehicles. Raymond Fischer (Noise Control Eng., Inc., 799 Middlesex Turnpike Billerica, MA 01821, rayf@noise-control.com)

To predict and control the radiated noise from underwater vehicles can be a complicated task. As opposed to a surface ship with a hull surrounding the machinery sources, these vehicles have multiple acoustic sources in direct contact with water. Any surrounding hull-type structure, excited by the source, then also radiates into the acoustic media. This paper discusses these potential sources, e.g., propulsors, motors, pumps, controllers, high-pressure air system, and electronics. The transmission path from the sources to the ocean also needs to be defined and understood. Airborne, fluidborne, and structureborne paths exist for most vehicles and they are often cross-coupled. These vehicles can also have exotic hull materials whose radiation characteristics are quite different from the standard metallic hulls. Accurate models can identify which sources and paths are important and over which frequency ranges. To reduce the signature in an optimal manner and diminish the adverse impacts on space, payload, and cost of typical treatments, one must have a good understanding of the process of modeling the radiated noise of a vehicle. This paper discusses these critical factors.

8:55

4aUWa3. Underwater gliders as acoustic receiving platforms. Georges A. Dossot, James H. Miller, Gopu R. Potty, Kristy A. Moore (Univ. of Rhode Island, Dept. of Ocean Eng., Narragansett, RI), Scott Glenn (Rutgers Univ., NB, NJ), and James F. Lynch (Woods Hole Oceanograph. Inst., Woods Hole, MA 02882)

Acoustic data were collected on a single hydrophone towed by a Webb Slocum glider deployed by Rutgers University, during the shallow water experiment (SW06), on the continental shelf, of New Jersey. The geometry of the experiment provided for adequate recording of the 224 and 400 Hz tomography sources. A follow-up study of the New Jersey Tuckerton Field Station provided a rudimentary noise analysis showing the glider's capabilities as an acoustic receiving platform. The glider's saw-tooth glide profile allows for vertical sampling of the water column with periodic surfaces for GPS fixes and data transfer via satellite phone. The glider provides a low-noise and low-speed platform, potentially enabling detection of low level signals. [Work sponsored by the Office of Naval Research.]

9:20

4aUWa4. Application of boundary layer suction for reducing hydrophone sensing noise. Craig N. Dolder (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, dolder@utexas.edu), Meagan A. Villanueva (Dept. of Aerosp. Eng. and Eng. Mech., The Univ. of Texas at Austin, 1 University Station C0600, Austin, TX), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029), and Charles E. Tinney (Dept. of Aerosp. Eng. and Eng. Mech., The Univ. of Texas at Austin, 1 University Station C0600, Austin, TX)

One of the leading noise sources for hydrophone arrays on moving vessels is hydrodynamic noise which results from the presence of turbulent boundary layers. This ongoing experimental study explores the impact boundary layer suction has on reducing the hydrodynamic pressure fluctuations and thereby increasing the signal to noise ratio for sensing. A custom hydrophone array is used to capture the pressure signatures with and without boundary layer suction under 2D fully developed turbulent boundary layers with momentum thickness Reynolds numbers ranging from 2000–4000. The first generation suction device shows a reduction in noise of up to 50% with moderate suction intensities. The current focus is on observing the effect suction has on the boundary layer velocity field using both single point laser Doppler anemometry and 2D particle image velocimetry. This information will provide insight into how the hydrodynamic structures are being removed and will provide a basis for the development of future optimized suction devices.

Contributed Papers

9:45

4aUWa5. Radiated noise measurements in a harbor environment using a vertical array of omnidirectional hydrophones. Brian Fowler (Grad. Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, bef5000@arl.psu.edu) and Christopher Barber (Penn State Univ., P.O. Box 30, State College, PA 16804)

Measurement of the radiated noise of a ship or submerged vessel in shallow water is complicated by the presence of multiple surface and bottom reflections, requiring that environmental effects on propagation either be minimized in the measurement or accounted for in the result, or perhaps both. In a harbor environment, there are additional sources of reflection and noise that may degrade the ability to obtain meaningful measurements. The combination of multiple reflected paths, high background noise, and a possible inability to assume far-field behavior due to a shortened range present a significant challenge to acquiring high confidence measurements of the radiated acoustic field. This work presents preliminary results from a radiated noise measurement test conducted at the U.S. Navy's Acoustic Research Detachment in Bayview, Idaho during summer 2010. A line array of 14 equispaced omnidirectional hydrophones was deployed from a barge tied up adjacent to a moored test vessel to obtain radiated noise measurements. A series of test signals was also transmitted through a calibrated acoustic source deployed at various depths in the harbor to evaluate the effectiveness of vertical line array measurements in minimizing reflected path contributions and improving signal-to-noise ratio. Preliminary results and conclusions are presented. [Work sponsored by the Office of Naval Research, Code 331.]

10:00

4aUWa6. Review of methods for making radiated noise measurements of submerged vessels. Christopher Barber (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804, cbarber@psu.edu)

There are a variety of challenges associated with measuring the radiated noise of surface ships and submerged vessels in order to obtain a measurement and environment-independent parameter such as the far-field equivalent source level. While measurements of large ships and naval vessels have routinely been conducted at deep water test sites in what approximates a far field measurement in a free-field environment, open ocean, fixed range measurements are not always practical for smaller vessels and particularly for autonomous or unmanned undersea vehicles. This presentation provides a brief review of several existing methodologies and examines some of the specific challenges associated with obtaining high quality estimates of vessel radiated noise associated with various methods and measurement scenarios.

Session 4aUWb

Underwater Acoustics: Autonomous Underwater Vehicle Navigation

Brian T. Hefner, Chair
Applied Physics Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698

Contributed Papers

10:30

4aUWb1. Navigation and sonar applications of an acoustical spiral wave front beacon. Benjamin R. Dzikowicz (Naval Res. Lab., Physical Acoust. Branch Code 7136, 4555 Overlook Ave. SW, Washington, DC 20375, benjamin.dzikowicz@nrl.navy.mil), Brian T. Hefner (Univ. of Washington, Seattle, WA 98105), and Robert A. Leasko (Naval Surface Warfare Ctr., Panama City, FL 32407)

A spiral wave front beacon consists of a transducer whose phase depends on the azimuth at which it is received and a reference transducer whose phase does not. [J. Acoust. Soc. Am. **129**:(6), 2011]. A spiral wave front can be produced by using a cylindrical transducer whose radius advances by one wavelength over one revolution, or by phasing a circular array of elements out of phase such that they generate a spiral wave front. Several experiments using both of these types of beacons are carried out at the Navy's Dodge Pond facility in Connecticut. This facility provides a range of environments where the robustness of the signals in multipath and reverberation can be tested. Navigation experiments are carried out using a remotely operated USV equipped with a hydrophone and a data acquisition system which triggers upon receipt of an incoming signal. The vehicle is also equipped with a differential global positioning system (DGPS) receiver to determine its exact position. Several types of outgoing signals are employed. The beacons are also tested for use in passive and active sonar applications where the phase advance between the transducers gives an indication of direction. [Work supported by the Office of Naval Research.]

10:45

4aUWb2. A robust phase gradient bearing estimation algorithm for a tri-axis cross-dipole acoustic array, with application to a long range autonomous underwater vehicle homing and tracking system. Carmen Lucas, Garry Heard, Nicos Pelavas, and Richard Fleming (Defence RD Canada - Atlantic, P.O. Box 1012, Dartmouth, NS, Canada B2Y 3Z7, carmen.lucas@drdc-rddc.gc.ca)

A bearing estimation algorithm was developed as part of a long range acoustic bearing (LRAB) homing system implemented on an autonomous underwater vehicle (AUV). A tri-axis cross-dipole acoustic array with seven digital hydrophones was developed and mounted in the AUV for the homing system. The Phase Gradient algorithm was implemented on the vehicle's Acoustic Homing and Localization System processor and run in real-time. The algorithm was designed to estimate the bearing and elevation angles to a continuous wave (CW) signal from a beacon source. The algorithm directly estimates the three Cartesian components of the incoming signal wave-vector from estimated cross-spectra between the hydrophone elements. The algorithm is robust against hydrophone failure, and every hydrophone in the array is used to estimate each component of the wave-vector. In this paper, the theoretical development of the Phase Gradient algorithm is presented, as well as the results from real applications of the algorithm as part of an AUV homing and tracking system.

11:00

4aUWb3. Linear drift error for cornerstone autonomous underwater vehicle. Nicos Pelavas, Garry J. Heard, Carmen E. Lucas, and Derek Clark (Defence Res. Development Canada Atlantic, PO Box 1012 Dartmouth, NS, Canada B2Y 3Z7, nicos.pelavas@drdc-rddc.gc.ca)

Autonomous underwater vehicles (AUVs) have a promising future in their use of collecting bathymetric data in remote regions of the Arctic Ocean. In accordance with the United Nations Convention on the Law of the Sea, Defence Research & Development Canada Atlantic has partnered with Natural Resources Canada (NRCan) and Department of Fisheries and Oceans to use AUVs in support of Canada's Arctic claim. In this article, we investigate the AUV linear drift error that accumulates as a result of a misalignment between the Inertial Navigation Unit and the Doppler Velocity Log. Data collected during the 2010 Cornerstone Arctic field trial is used to quantify the linear drift error associated with one of the AUVs. The linear drift error was determined to be 0.67% to stern and 0.21% to starboard, and this result was then applied to correct the track for the AUV survey mission.

11:15

4aUWb4. Modeling a spiral wave front source in an ocean environment. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105 USA) and Benjamin R. Dzikowicz (Physical Acoust. Branch, Code 7136, Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375)

A spiral wave front source generates a pressure field which has a phase that depends on the azimuthal angle at which it is measured [J. Acoust. Soc. Am. **129**(6), (2011)]. This type of source can be used in conjunction with a reference source to form a navigation beacon. A remote receiver can determine the direction to the beacon from the phase difference between the pulses transmitted from each of the sources. To determine the accuracy of this navigation technique, it is necessary to model the output of the spiral wave front source in ocean environment. To this end, the spiral wave front analogue of the acoustic point source is examined and is shown to be related to the point source through a simple transformation. This makes it possible to transform the point source solution in a particular ocean environment into the solution for a spiral source in the same environment. This transformation is applied to simple cases, such as reflection from the sea surface, as well as to the more general case of propagation in a horizontally stratified waveguide. [Work supported by the Office of Naval Research.]

Session 4aUWc

Underwater Acoustics: Acoustic Propagation Modeling

Matthew A. Dzieciuch, Chair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

8:00

4aUWc1. Recent developments in the seismo–acoustic parabolic equation. Michael D. Collins (Naval Res. Lab., Stennis Space Ctr., MS 39529, mike.collins@nrlssc.navy.mil)

The development of the seismo–acoustic parabolic equation is currently proceeding in several directions. Since the stability of the parabolic equation is an issue for problems involving relatively thin elastic layers, special rational approximations are being designed for problems involving ice cover. Range dependence has previously been treated with approaches based on single scattering, energy conservation, and coordinate changes that are of limited use for certain cases. For test problems involving gradual range dependence (a convenient reference solution is available in this limit), promising results have been obtained with a single-scattering approximation that conserves quantities across a vertical interface in a mean sense. One of the advantages of this approach is that its simple physical interpretation facilitates generalizing to different cases, such as problems involving sloping fluid–solid interfaces and anisotropic and poro-elastic layers. [Work sponsored by the Office of Naval Research.]

8:15

4aUWc2. Three-dimensional underwater sound propagation using a split-step Padé parabolic equation solution. Jon M. Collis (Colorado School of Mines, 1500 Illinois St., Golden, CO 80401)

The majority of current three-dimensional (3-D) parabolic equation propagation model development work has been focused on implementations. Recent attention has been on the modeling of internal wave fields and propagation in the presence of strong internal tides and eddies. State of the art solutions are experimental water-tank comparisons against azimuthal schemes [Sturm *et al.*, JASA **113**] and split-step Fourier based propagation models [Lin *et al.*, JASA **126**]. A current thrust is to establish 3-D benchmarks for propagation simulations in range-dependent environments where effects due to generic bottom features are present. In this paper, a 3-D extension to current 2-D split-step Padé solutions is developed and benchmarked. Transverse and depth operator discretizations are performed using a Galerkin method. Cartesian versus cylindrical coordinate systems are discussed as is the nature of the acoustic source in either geometry.

8:30

4aUWc3. Propagation of coupled modes in three dimensions. Ronald F. Pannatoni (540 Mark Dowdle Rd., Franklin, NC 28734, elliptic@alum.mit.edu)

A time-harmonic acoustic field in an ocean can be represented in terms of local normal modes. This representation may include leaky modes to account for radiation into the bottom. A system of coupled elliptic equations governs the amplitudes of these modes. If there is a primary horizontal direction of propagation, it may be possible to approximate these equations by parabolic equations. The resulting system of coupled parabolic equations can be integrated in the direction of propagation only if a certain linear transformation is nonsingular. This constraint limits the size of the system and the amount of coupling among the modes that are consistent with the parabolic approximations. A method for integrating the coupled parabolic equations numerically is discussed, and it is applied to a problem of three-dimensional propagation and scattering around a conical seamount in a deep ocean.

8:45

4aUWc4. A three-dimension propagation model using stepwise coupled modes. Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

A propagation model has been developed that is applicable to shallow-water waveguides characterized by three-dimensional inhomogeneities which induce horizontal refraction and mode coupling. A normal-mode approach is chosen for this work because the field decomposition into modal amplitudes provides insight into the effects of the environment on the acoustic field. The model is based on the stepwise coupled-mode technique implemented with the single-scatter approximation [R. B. Evans, J. Acoust. Soc. Am. **74**, 188–195 (1983)]. The stepwise technique discretizes a range-dependent environment into a series of range-independent segments. The single-scatter solution is obtained by treating each pair of segments as an independent problem, thus neglecting the higher-order terms resulting from multiple scattering at other interfaces. The field is propagated using the parabolic equation in cylindrical coordinates. Thus, at each range step, horizontal refraction is accounted for in the angular direction and mode coupling is included in the radial direction. Examples illustrating the effects of horizontal refraction and mode coupling will be presented. [Work supported by ONR.]

9:00

4aUWc5. Discrete transparent boundary conditions for parabolic equations. Ronald F. Pannatoni (540 Mark Dowdle Rd., Franklin, NC 28734, elliptic@alum.mit.edu)

There are simple algorithms for constructing transparent boundary conditions (TBCs) for a partial discretization of the basic parabolic equation that is known as a "semi-discrete" parabolic equation. This equation and some of these algorithms are reviewed. Solutions of a semi-discrete parabolic equation in a long rectangular strip subject to TBCs at the long edges of the strip are then considered. These solutions can be computed accurately and efficiently with a pseudospectral method that is based on expansions in Chebyshev polynomials. It is beneficial to combine this method with a conventional split-step FFT solution of a parabolic equation subject to Neumann boundary conditions at the long edges of the strip. This hybrid approach will be called the "decomposition method." It is demonstrated in a computation of radiation modes from the termination of a truncated nonlinear internal gravity wave duct in a shallow water area.

9:15

4aUWc6. Equation, describing temporal evolution of the sound pulse in horizontal plane in shallow water. Boris Katsnelson (Voronezh State Univ., Universitetskaya Sq. 1, Voronezh 394006, Russia, katz@phys.vsu.ru)

It is shown that for comparatively narrow-band pulses in shallow water sound propagation, it is possible to introduce modal pulses, corresponding to variation (or evolution) of space-time dependence of the amplitude of separate waveguide modes. In ray approximation, in the presence of

perturbation depending on horizontal coordinates, these signals propagate along different trajectories in horizontal plane (space-time horizontal rays) depending on mode number and frequency. In the paper equation, describing evolution of amplitude of modal pulses as a function of time in horizontal plane outside the ray approximation is obtained. This equation can be considered as extension of well-known parabolic equation in horizontal plane. Examples of solutions for some shallow water models are shown; applicability of this equation is discussed.

9:30

4aUWc7. Seismo-acoustic propagation near thin and low-shear speed ocean bottom sediments. Jon M. Collis (Colorado School of Mines, 1500 Illinois St., Golden, CO 80401), Adam M. Metzler (Rensselaer Polytechnic Inst., Troy, NY 12180), Paden Reitz, and Rezwanur Rahman (Colorado School of Mines, Golden, CO 80401)

Accurate and efficient parabolic equation solutions exist for complex propagation environments featuring elastic and porous elastic sediment types. Because of numerical stability issues, areas of concern have been with thin and low-shear wave speed sediments. In certain situations, both of these common features of the seafloor can make obtaining an accurate solution difficult. At low frequencies, layers of this type can be treated as a massive interface between the water and higher-shear speed sediment basement layers. To satisfy interface conditions across the layer, Rayleigh jump conditions are imposed [F. Gilbert, Ann. Geofisica **XL**, 1211 (1997)]. This approach is only valid for a single layer, but is able to handle shear wave speeds as they tend to zero. In this talk, a massive elastic interface parabolic equation implementation is benchmarked along with classical bottom treatments to quantify the effects of ocean acoustic propagation over thin sediment layers. It is demonstrated that in certain situations, it is sufficient to consider the thin layer as part of adjacent, thicker layers.

9:45

4aUWc8. Robust computation of acoustic normal modes in attenuating ocean waveguides. Thomas J. Hayward and Roger M. Oba (Naval Res. Lab., Washington, DC 20375)

The normal mode representation [Jensen *et al*., *Computational Ocean Acoustics*, AIP Press, 1997] provides for the mathematical analysis and numerical computation of acoustic fields in a range-independent ocean waveguide. Existing algorithms [e.g., M. B. Porter and E. L. Reiss, J. Acoust. Soc. Am. **77**, 1760] provide for efficient computation of the mode eigenvalues and eigenfunctions for narrow-band acoustic fields. However, for extensive computations involving a large set of environmental parameter values or acoustic frequencies, reliability issues, such as eigenvalues omitted in the calculation, have been noted in the case of attenuating media. In this work, a computational method is presented that computes the normal modes by approximate solution of the mode depth-dependence equation on a discrete computational grid, using a selected discrete basis. Empirical evidence of the robustness of the method is provided by comparisons with established numerical benchmarks and by examining the acoustic parameter dependence of the mode spectra over thousands of parameter values. Theoretical support for the reliability of the computation is then discussed. [Work supported by the Office of Naval Research.]

10:00–10:15 Break

10:15

4aUWc9. An approach to computing acoustic wave propagation in shallow water. Cathy Ann Clark (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841)

A shallow water mode solution is presented, which approximates the water column sound speed variation using isovelocity layers. Within a layer, two fundamental solutions are seen to satisfy the separated depth-dependent wave equation and complex analogs apply in evanescent regions. Solutions which satisfy an upper boundary condition are extended continuously through isovelocity layers to the bottom, matching functions and derivatives at layer boundaries. The value of the Wronskian for the Green's function thus obtained is used to locate the eigenvalues of the normal modes comprising the propagating field. Acoustic mechanisms which are dominant in shallow water such as forward scattering and range dependence are incorporated as matrix multiplications to implement horizontal coupling between modes. Agreement between the shallow water approach and a benchmark deep water mode solution is shown for a number of shallow environments.

10:30

4aUWc10. Sonar propagation modeling using hybrid Gaussian beams in spherical/time coordinates. Sean M. Reilly (Dept. of Ocean Eng., Univ. of Rhode Island, Bay Campus, Narragansett, RI 02882)

This paper defines a new undersea acoustic transmission loss model that is optimized for real-time, sonar simulation/stimulation systems in littoral environments. The ray solutions to the eikonal equation are computed in latitude, longitude, and altitude coordinates to match wide-area environmental databases. Hybrid Gaussian beam techniques for transmission loss calculation are used to extend the applicability of ray theory to lower frequency regimes. Numerical integration of the wave equation is performed in the time dimension to support broadband signal modeling. This 3-D approach also supports out-of-plane reflection from the ocean bottom. This paper derives the eikonal solution from first principles to ensure a complete understanding of the coordinate system's impact.

10:45

4aUWc11. Underwater acoustic propagation in Arctic environments. Christie A. O'Hara (Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD 20723) and Jon M. Collis (Colorado School of Mines, Golden, CO 80401)

Developments in underwater acoustic modeling for the Arctic have been limited due to the complicated nature of the polar extreme. In Arctic regions, the sound speed minimum occurs at or near the ice-covered surface. The upward refracting sound speed profile causes any long-range propagation to repeatedly interact with the ice cover. This paper presents an overview of the derivation of a 2-D normal mode propagation model to the range-independent wave equation for a source and receiver in the water column. To more accurately model the Arctic ocean acoustic environment, we consider a modified Pekeris waveguide. As with the Pekeris waveguide, the bottom is considered as an infinite fluid halfspace and the top is considered to be a fluid layer of finite thickness overlying the water column. Results will be benchmarked against a fluid parabolic equation solution. An application is discussed as a means to track marine mammals using received signals.

11:00

4aUWc12. Perth-Bermuda revisited again: Global adiabatic mode parabolic equation results. Kevin D. Heaney and Richard L. Campbell (Ocean Acoust. and Instrumentation Systems, Inc., 11006 Clara Barton Dr. Fairfax Station, VA 22039, oceansound04@yahoo.com)

In 1960, a set of explosives were detonated off the coast of Perth Australia, and multi-pulse receptions were recorded from moored hydrophones off of Bermuda. The Perth-Bermuda experiment demonstrated the capability of trans oceanic acoustic propagation. The two-pulse arrival, separated by approximately 25 s, was explained by Heaney, *et al*. (JASA, **90**(5), 2586–2594) in terms of two disparate paths: a northern path refracting off islands in the southern Indian Ocean and the other refracting off the shelf-break on the coast of Brazil. In this paper a global adiabatic mode-parabolic equation hybrid model is used to compute the multi mode, broadband full-field pressure response from Perth to Bermuda. The PE field demonstrates that Bermuda is in the acoustic shadow of the refracted geodesics, yet observations of arrivals were made. PE results demonstrate that two significant scattered paths, one to the north passing by the Cape of Good Hope and one to the south, passing by the coast of Brazil yield strong strong arrivals, confirming the results of Heaney *et al*.

11:15

4aUWc13. Acoustic modes in a curved internal wave duct. Ying-Tsong Lin (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, ytlin@whoi.edu), Kara G. McMahon, William L. Siegmann (Rensselaer Poly. Inst., Troy, NY 12180), and James F. Lynch (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Observations and numerical simulations have shown that nonlinear internal waves in continental shelf and shelfbreak regions can form 3-D acoustic ducts. The strength of ducting depends on the size of internal waves, the width of the gap between waves and the curvature of the wave front, and also on the acoustic frequency and the vertical mode number. It has been seen in numerical simulations and simplified ray theory that for a given internal wave structure and a given frequency, higher vertical modes are easier being trapped in a curved internal wave duct. Also, the number of the lowest mode trapped between curved waves increases as the frequency goes up. In this talk, a 3-D normal mode theory is employed to analyze these observed characteristics. The analysis is carried out in a cylindrical coordinates, and two types of horizontal modes are found: whispering-gallery modes and full bouncing modes. Both types of modes can be described by Bessel functions, and the asymptotic formulas can be used in some limiting cases. [Work supported by the ONR.]

11:30

4aUWc14. Detection performance modeling and measurements for convergence zone (CZ) propagation in deep water. Kevin D. Heaney, Richard L. Campbell, James J. Murray (Ocean Acoust. and Instrumentation Systems, Inc. 11006 Clara Barton Dr. Fairfax Station, VA 22039, oceansound04@yahoo.com), Gerald L. D'Spain (Scripps Inst. of Oceanogr., UCSD), and Arthur B. Baggeroer (Massacheussets Inst. of Technol., Cambridge MA)

A novel parabolic equation algorithm in the C programming language has been developed based upon the RAM model. This model (CRAM) permits modeling of the full-field sonar equation to estimate towed array performance of the detection of quiet targets in a dynamic environment with both environmental range dependence and source/receiver kinematics. During an experiment in the northern Philippine Sea in 2009, a ship towing Penn State's Five-Octave Research Array (FORA) was towed at various depths in a star pattern about the station-keeping source ship, thereby sampling the first CZ in range, depth, and azimuth. Measurements and modeling of the CZ arrivals will be compared. A simple detection processor is applied to the CZ receptions. Comparison of passive ASW performance modeling results with measurements will be made. One of the primary science issues in the statistics associated with probabilistic detection is the time between independent samples, or the sample-to-sample correlation. This will be evaluated from the data for a portion of the test where the receiver was towed in an arch-fixing the source–receiver range for several hours. [Work supported by ONR.]

11:45

4aUWc15. An investigation of the effects of rough seas and bubble injections on high frequency propagation using a parabolic equation method. Joseph M. Senne, Aijun Song (CEOE, Univ. of Delaware, Robinson Hall, Newark, DE 19716, sennejm@udel.edu), Kevin Smith (Naval Postgrad. School, Monterey, CA 93943), and Mohsen Badiey (Univ. of Delaware, Newark, DE 19716)

High frequency underwater acoustic transmissions (>10 kHz) are heavily influenced by scattering from both rough surfaces and bubbles. These interactions are recorded through the prevalence of micro multi-paths in observed data. To study these scattering effects, a rough-surface variant of the Monterey Miami parabolic equation model was combined with a hydrodynamic surface model that produces non-linear waves along with depth- and range-dependent bubble distributions. Parabolic equation setup parameters were taken from collected environmental data, while a wave-rider buoy was used for time-evolving sea surface generation. Bubble plume densities were calculated using surface white-cap distributions along with a bubble evolution scheme. Comparisons of the simulated results are made against collected acoustic data for calm and rough sea states [Work supported by ONR Code 322OA.]

THURSDAY AFTERNOON, 3 NOVEMBER 2011 SUNSET, 1:00 TO 5:25 P.M.

Session 4pAAa

Architectural Acoustics, Noise, and Committee on Standards: Networking in Soundscapes—Establishing a Worldwide Collaboration II

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

Chair's Introduction—1:00

Invited Papers

1:05

4pAAa1. Getting it together—Interdisciplinary sound environment research. Frans Mossberg (Dept. of Cultural Studies, Lund Univ., Biskopsgat 7, 22100 Lund, Sweden, fransmo@glocalnet.net)

The Sound Environment Center at Lund university is an interdisciplinary center created to coordinate research on sound and soundscape issues and is known to be the first of its kind worldwide. Ranging from acoustics to medicine, psychology, and cognitive sciences, as well as humanities like musicology and linguistics, soundscape research adresses many interdependent areas and touches upon health as well as philosophical, aesthetic, and technical issues. To get a holistic comprehension, these perspectives need to be synchronized. Therefore, the center has an interdisciplinary board and a mission to study sound environments from multidisciplinary perspectives. Focus lies on research and contact between researchers. The center has external funding for larger research collaborations on topics such as teachers voice strain and rooms acoustics, health effects of combined exposure to noise and airborne particles, cognition, and sound exposure. In addition to initiating research projects, the center arranges symposiums adressing topics such as Noise and health, Seductive Sounds, Operational Sounds, Dangerous Sounds' and Sound, Cognition and Learning. Further topics have been Sound Design, Sounds and Silence for Mental Recreation, Teachers Voice Comfort, and recently Wind Turbine Noise. The symposiums facilitates cross disciplinary contacts and discussions, many of them producing published papers and reports.

1:25

4pAAa2. Soundscape ecology: A worldwide network. Catherine Guastavino (McGill Univ., School of Information Studies and CIRMMT, 3661 Peel, H3X 1X1, Montreal, QC, Canada, catherine.guastavino@mcgill.ca) and Bryan C. Pijanowski (Purdue Univ., West Lafayette, IN 47906)

The overarching objective of our network is to bring together acousticians, cognitive psychologists, ecologists, and creative artists to integrate how they study and perceive soundscapes and use this knowledge to help shape a research agenda for the conservation of soundscapes. Many natural soundscapes are being threatened from various directions, e.g., habitat destruction, climate change, invasive species. This project aims at recording and documenting soundscapes in remote locations and identifying conceptualizations of these soundscapes across different cultures and disciplines. The network will help to (1) foster open communication between different disciplines and communities about soundscapes; (2) coordinate soundscape monitoring sites where acoustic data are being collected long-term; (3) develop a common vocabulary, long-term monitoring standards, and metadata standards for acoustic data for use by ecologists; (4) increase awareness of this new field among ecologists and social scientists; and (5) increase public awareness of the importance of their acoustic connection to nature. This project on the soundscapes of natural ecosystems is a logical complement to research underway at the European level on urban soundscapes (COST action TD0804). Together, this worldwide network of researchers will be uniquely situated to contribute decisively to a cross-cultural conservation framework for soundscapes.

1:45

4pAAa3. Mapping the landscape of soundscape research. Hill Kobayashi and Saoru Saito (World Forum for Acoust. Ecology (WFAE) P.O. Box 268, Fairfield Victoria, 3078 Australia)

In this paper, I give a personal view on what could be the landscape of soundscape research. I will describe the philosophical content of this discipline, the current state of aesthetic aspect, the challenges in technical issues and the future direction of Soundscape activities among researchers, practitioners, designer, and composers. Given that to map a landscape of this discipline with discussion is a very challenging task. It requires us to connect knowledge from different disciplines with perceptional sense. The paper aims to address recent collaboration opportunities for interdisciplinary engagement and key topics for debate.

2:05

4pAAa4. Sounds in cause: Soundscape and evolution. José Manuel Berenguer (Caos-Sonoscop. CCCB. Montalegre, 5. 08001. Barcelona. Spain)

Beyond the therm of landscape, that has often been defined as "a painting, drawing, or photograph depicting natural scenery," the concept of soundscape should lack any aesthetic significance and be thought as a technical therm naming the whole sonic experience of animals having sense of hearing. In general, from a methodological point of view, approaches describing landscape as "a view of some natural place" does not seem to be useful, because artificial and natural are often indistinguishable. Soundscape does not need to be considered natural or artificial. For instance, it seems evident that in a city, most sources of sound objects in soundscapes are human activities; anyway, even in a city, sounds produced by non-human species can easily be found. Soundscapes are complex structures that can be considered in terms of evolution. They evolve sound sources adapt their sonic productions to sonic productions of other sound sources sharing the same environment. It happens in human and non-human soundscapes. This way of thinking underlies sounds in cause, a project that started building a database of soundscapes recorded following a strict methodology and now proposes an Internet2 Network of Laboratories and Stations for Permanent Listening to the Soundscape.

2:25

4pAAa5. On the soundscape of urban parks. J. Luis Bento Coelho and Mohammed Boubezari (CAPS, Instituto Superior Tecnico, TULisbon, Lisbon, Portugal)

Urban parks are parts of every city fabric and usually well appreciated by the citizens for providing restoration and some "quiet," at least when compared to most other city areas. Research on the soundscape of parks in cities in Portugal and in Brazil have been conducted in order to assess what makes such areas sonically interesting. Work is being directed to the differentiation of the sound components of the overall sound environment in different types of parks and on techniques for mapping the perceived sound components. The work also aims at understanding effects of climate and culture on the perception of the soundscape. Results are presented and discussed.

2:45

4pAAa6. A livingscape approach to characterize urban historical places. F. La Malva, A. Astolfi, P. Bottalico, and V.R.M. Lo Verso (TEBE Res. Group, Polytechnic of Turin, Dept. of Energetics, Turin, Italy)

The quality of life concerned with open spaces has more and more become an essential part of urban culture. The evaluation of environmental effects as perceived by people is primarily a subjective issue, rather than being simply based on objective parameters. This paper presents an approach called livingscape, used to assess the quality of urban spaces. It consists of analyzing and correlating psychometric tools to measure the perception of environmental quality with different aspects related to the urban blight (both in architectural and environmental terms) and objective investigation of environmental quality through the measurement of acoustic, visual, thermal, and IAQ physical parameters. Livingscape data were collected in 13 key-spaces of St. Salvario, an historical district in Turin (Italy), during summer 2010 and winter 2011, selected based on an historical analysis to characterize the past and present district soundscape and subdivided in nodes, paths, and edges. The subjective environmental perceptions are delineated through the analysis of the questionnaires submitted to the users of the area, outdoor. Objective measures (acoustical, lighting, and thermal parameters) were combined to subjective responses, thus providing a more complete key-spaces characterization. The investigation aims to describe the changes in the key-spaces characterization from 19th century to nowadays.

3:05–3:15 Break

3:15

4pAAa7. Using a soundscape approach to develop an acoustic ecology plan for a city. Lisa R. Lavia (Noise Abatement Society, Ste. 2, 26 Brunswick Terrace, Brighton, England, BN3 1HJ, United Kingdom), Max Dixon (Independent Consultant, London, England, United Kingdom), Osten Axelsson (Decorum Commun.s, Passvagen 30, SE-147 53 Tumba, Sweden), and Harry Witchel (Brighton and Sussex Med. School, Falmer, Brighton, BN1 9PS, United Kingdom)

Sounding Brighton is a collaborative project exploring practical approaches toward better soundscapes focusing on soundscape issues related to health, quality of life, and restorative functions of the environment. The project provides the opportunity to raise awareness and promote communication on soundscapes among the general public, stakeholders and those involved in policy, including encouraging exploration of new ways of listening in local soundscapes, and new ways of tackling noise and improving local soundscape quality. The project is working to provide opportunities to discuss how soundscape concepts might, alongside tackling conventional noise problems, contribute to local planning and environmental improvement as part of a city wide engagement process in the city of Brighton and Hove in England in the United Kingdom. A range of environments, e.g., seafront, foreshore, historic terraces, squares, lanes, parks, and gardens, are being considered. A soundmap of the city is being developed utilizing the Swedish Soundscape-Quality Protocol (developed by Osten Axelsson, Mats E Nilsson and Birgitta Berglund); a public outreach exhibition is being developed; and a night noise intervention study is planned to explore the relationship between soundscapes and the brain, community well being, social cohesion, and the physical and mental health of individuals.

3:35

4pAAa8. The introduction of the concept of soundscape to urban design analysis. Maria Tomalova MA (46 Leinster Gardens, W2 3AT, London, the UK, m_tomalova@hotmail.com)

The objective of this paper was to provide pragmatic approach through the analysis of Covent Garden Piazza's soundscape with intention to show the way of its implementation into urban design analysis, whereas analysis of soundscape is missing part of urban design analysis. So, soundscape suppose to be a new tool for designer to create more pleasant sound ambient environments not only for users who are able to perceive urban space visually but also for those who can use another senses only like aural and tactile to identify relevant quality of urban places. In this research were employed methods of subjective evaluation and spatial analysis of soundscape (entirely qualitative research). The data were collected through exploration of the soundscape preference and level of awareness of ambient sound environment within explorative single case study and thorough observation of activities on the study site and investigation of urban form. Thus, the urban uses and activities of the location and its acoustic identification are closely linked. Recent urban analyses generally take only human and technical noises into consideration, excluding the rest of noises of sonic environment and way of their perception by human beings. Acoustic dimensions must be available for design.

3:55

4pAAa9. Urban sound design—Utopia or urgent need. Nina Hllgren (Dept. of Architecture, KTH Royal Inst. of Technol./Konstfack Univ. of Arts Crafts & Design, Box 3601, 12627 Stockholm, Sweden, nina.hallgren@konstfack.se)

Today we are facing the consequences of about 100 years of urbanization, confronting questions about quality of life in relation to efficiency and economical benefits. Hard facts are considered to be more reliable than values which are not so easily measurable. The quality of sound is one of them. Urban designers and architects are currently not fully aware of the interaction between the built outcomes of their work and the process of propagation and perception of sound. Designing houses, relations between houses, connections between places, whole neighborhoods, and cities is a serious task affecting many different aspects of life. But we can still note a recurring absence of knowledge regarding the complex relation between visual and sonic realities. This reality is in fact what surrounds the urban inhabitant for an entire lifetime. But what can or should the architects and planners do? As has been recently pointed out in a report on the subject, this professional group is lacking the tools, language, and guiding examples for being able to implement anything in reality. Implementing what? What can really be done or designed to improve the sonic environment and for what purpose?

4:15

4pAAa10. Piazza del Marchese Paolo: An architectural and soundscape design to redevelop an outdoor public space. Achille Sberna, Francesco Asdrubali (Dept. of Industrial Eng., Univ. of Perugia, Via G. Duranti 67, 06125 Perugia, Italy), and Brigitte Schulte-Fortkamp (Technische Universitaet Berlin, Germany)

This paper will report about a design procedure regarding the redevelopment of an open square located in the historical center of Città di Castello, Italy. The square has five entrances and it is surrounded by old buildings. The Public Library of the town is also located at this place. Currently, the square is used as a parking lot. The goal of the design is to redevelop this square matching the given context. For the design procedure, first, the visual and acoustical status of the place will be described. Second, binaural recordings will be carried out to measure the acoustical climate and third, soundwalks will be conducted to help to detect the soundmarks of the area. Moreover, the idea is to transform the space in a pedestrian area and to consider the square as an acoustical "outdoor floor" for the library. The design process will be focused on the preservation of the genuine Soundscape.

4:35

4pAAa11. How the soundscape approach enlightens the knowledge about the feeling of safety in urban spaces: A matter of networking and interdisciplinarity. Kay Sebastian Voigt and Brigitte Schulte-Fortkamp (Technische Universite, Inst. Fluid Mech. and Eng., Secr TA7, Einsteinufer 25, Berlin, Germany)

In a current project on dynamics of urban safety and its arrangements funded by the Federal Ministry of Education and Research, Germany http://www.dynass-projekt.de/projekt-dynass/ Dynamische Arrangements städtischer Sicherheitskultur/, the Soundscape approach is one of diverse approaches to investigate areas of different cities with respect to the perception and production of safety in such environments. As to the perception of safety, areas as well as its production special attention will be given to the reconstruction of decisive factors. The current status of the international standardization process of the ISO/TC 43/SC 1/WG 54 and the contribution of the COST network are used here as the basis to improve procedures and measures in the applied Soundscape approach. Moreover, this paper will discuss the Soundscape research as an interdisciplinary approach and in an interdisciplinary approach with regard to investigations upon safety management in urban areas. First results give hints to the importance of the acoustic, visual, and social structures of an area in the face of the Soundscape procedure.

Contributed Papers

4:55

4pAAa12. Water features and acoustic diversity of urban parks. Osten Axelsson (Passvaegen 30, SE-147 53 Tumba, Sweden) and Mats E. Nilsson (Stockholm Univ., SE10691 Stockholm, Sweden)

Water features are well-acknowledged in architecture and urban planning for their visual characteristics. But, how do water features contribute to acoustic diversity and soundscape quality? Visitors in an urban park were recruited to complete a questionnaire on how they perceived the park including its soundscape. Meanwhile, the soundscape was manipulated by turning a fountain on or off at irregular hours. The fountain sounds had a positive effect on soundscape quality in an area close to the fountain, by masking background road-traffic noise. The fountain sound also masked other natural sounds, which may have a negative influence on acoustic diversity and soundscape quality. In addition, some participants may have mistaken the fountain sounds for distant road-traffic noise. Hence, when introducing a water feature in an urban park it is necessary to consider the acoustic characteristics of the water sounds, as well as the placement of the water feature.

5:10

4pAAa13. Researching sound in silence. Jurgen De Blonde (Aifoon vzw, Nieuwevaart 117a, B-9000 Gent, Belgium, jurgen@aifoon.org)

Aifoon is an educational arts organization that investigates sound in silence. We investigate poetic and communicative possibilities of everyday sounds. Our aim is to have people communicate about their immediate surroundings in a non-verbal way by using sounds taken from those surroundings. In our workshops, we teach people to record sounds, we teach microphone awareness, we open ears, draw sounds, and compose with these drawing, compose with sound, make a montage on a computer, and talk about the results. We avoid using music and words since these two elements are too coded. We take the results and the questions that rise from our work back to artists, researchers, and the public for further interpretation and investigation. We have taken our core philosophy as a basis for an exposition, a number of events, an ensemble, and a couple of installations in an attempt to take our educational role beyond the workshop session into the public space. This has proven successful and has often helped us to explain our ideas and outsiders to understand them.

THURSDAY AFTERNOON, 3 NOVEMBER 2011 SUNRISE, 1:25 TO 5:45 P.M.

Session 4pAAb

Architectural Acoustics: Variable Acoustics—Methods for Effective Collaboration

Roger W. Schwenke, Chair
Research and Development, Meyer Sound Laboratories, 2832 San Pablo Ave., Berkeley, CA 94702

Chair's Introduction—1:25

Invited Papers

1:30

4pAAb1. What you call it does matter: A vocabulary for active acoustics. Kurt M. Graffy (Arup Acoust., 560 Mission St., San Francisco, CA 94105, kurt.graffy@arup.com)

Active acoustics (Electronic Architecture) systems have reached a maturity in terms of their acoustic performance which makes them a viable option for multi-use venues as well as renovations of existing facilities. However, the awareness on the part of owners and facilities regarding the capabilities and potential acoustic benefits of such systems is still developing. More significantly, the corresponding level of awareness for designers and architects, which active acoustics can actually represent options for how architectural, structural, and building service designs may proceed, is even less developed. At this intersection of past techniques versus current abilities, we are in need of a common vocabulary for active acoustics; a common vocabulary to interact with the design team, a common vocabulary to assist venue operators and owners understand the benefits, and a common vocabulary to interact with the musicians and artists in tuning and configuration. The presentation references some recent projects utilizing active architecture primarily focused on the interaction with the design team and subsequently with the artists, and the development of a common language as the projects moved forward.

4p THU. PM

1:50

4pAAb2. 2006 International Music Festival. Edward Dugger III (4490 SW Long Bay Dr. Palm City, FL 34990, edward@edplusa.com)

Throughout the 3-week run of the 2006 International Music Festival held in Boca Raton, Florida, the 4000 seat Mizner Park Amphitheatre was fitted with an active acoustic system. We will present a case study that explores many of the elements of the project including the initial audio system selection, installation, rehearsals, and performances. Since this was an unusual project both in its exterior application and large scale, we will present the design features that had significant implications for the musicians, concert presenters,

stage hands, and audience. At the request of the venue operators, a new design has been developed to install a permanent active acoustic system for the amphitheater. This design will also be presented.

2:10

4pAAb3. Successfully merging architectural and electronic acoustical treatments. Steve Barbar (30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

In enclosed volumes, the integration of electronic acoustical components with architectural surface treatments forms a hybrid system that produces the perceived acoustical conditions. Since the underlying operating principles for electro-acoustic enhancement systems differs considerably between manufacturers, the requirements for system infrastructure are not germane, nor is the optimum integration of architectural treatments. As a result, the nature of the work performed by the acoustical consultant changes to accommodate optimum performance of the specific "hybrid" system, which may also include other forms of variable treatments.

2:30

4pAAb4. Working with musicians to increase low-frequency performance of an active acoustic system in a music practice room. Ron Freiheit (Wenger Corp., 555 Park Dr. Owatonna, MN 55060, ron.freiheit@wengercorp.com)

To enhance the performance of an active acoustic system for music practice rooms, a new speaker was developed with extended low-frequency response. To better understand the performance desired from certain musicians, a small number of cellists provided observations and opinions about what environment was most pleasing to them, specifically related to the expanded low-frequency response of the active system. A system was developed that allowed the tuning of the low-frequency response to better ascertain at which point it was optimized and below or above which very little improvement was noted. Once these parameters were optimized, design criteria for the speakers were determined. Also discovered during this research were challenges musicians faced in discriminating between the direct sound and active sound field.

2:50

4pAAb5. The care and feeding of clients with variable room acoustics systems. Edward Logsdon (D. L. Adams Assoc., Inc., 1701 Boulder St., Denver, CO 80211, elogsdon@dlaa.com)

The care and feeding of clients with variable room acoustics systems. Now that the Variable Room Acoustics System (VRAS) is designed and installed in the room, what do you do? An Acoustical Consultant's Perspective One of the primary educational goals for the Colorado College, Edith Kinney Gaylord Cornerstone Arts Center in Colorado Springs, CO, is to encourage collaboration between the music, drama, dance, film, and visual artists who perform and display their work in the facility. The building offers many opportunities for film/video and theater students, or music and sculptural artists to work together on multimedia presentations. We used "collaboration" to further encourage the student and faculty performers and artists to communicate their needs and priorities for use of the multipurpose main theatre early on in the design. Subjective opinions of how the room sounds, or how best to control the VRAS system, had to be carefully addressed and assessed to avoid confusion and to achieve unanimity. We will review our process of establishing an artistic vocabulary that allowed a jury of users to advise us on the types and number of room presets needed for the various uses of the theatre and the validation and approval of the system.

3:10–3:30 Break

3:30

4pAAb6. Various applications of active field control. Takayuki Watanabe and Masahiro Ikeda (Spacial Audio System Group, Yamaha Corp., 10-1 Nakazawa-cho, Nakaku, Hamamatsu, Japan, watanabe@beat.yamaha.co.jp)

Several types of various active field control (AFC) applications are discussed, while referring to representative projects for each application. (1) Realization of acoustics in a huge hall to classical music program, E.g., Tokyo International Forum: This venue is a multi-purpose hall with approximately 5000 seats. AFC achieves "loudness" and "reverberance" equivalent to those of a hall with 2500 seats or fewer. (2) Compensation of acoustics on stage without rigid shell using the electro-acoustic method. E.g., High school auditoriums: In these renovation projects, AFC achieves "acoustical support" for performer on stage and "uniformity" throughout the auditorium from the stage to the audience area, etc. (3) Improvement of the acoustics under the balcony in auditoria. E.g., Experiments on a full-scale model and the school auditorium. The system is a non-regenerative system, and the loudspeakers, located at positions corresponding to measurement points across the balcony, recreate the reflecting sound from above the balcony area, which otherwise fail to reach to the listeners under the balcony. The results of the experiment show that the system is significantly better for all tests to the use of no system and that the system is superior to a standard PA (delay system).

3:50

4pAAb7. Acoustic repair: Recent experience with the acoustic control system (ACS) for improving acoustic conditions in two existing venues. Timothy E. Gulsrud (Kirkegaard Assoc., 954 Pearl St., Boulder, CO, 80302 tgulsrud@kirkegaard.com) and Arthur van Maurik (Acoust. Control Systems BV, Speulderweg 31 3996 LA Garderen, The Netherlands)

Active acoustics systems are becoming more prevalent in architectural acoustics practice, particularly in the context of repairing or improving acoustics in existing venues. Governmental policies to reduce funds and subsidies put into new facilities for the performing arts are another reason for designers to consider the use of active acoustics. This paper highlights two recent examples of such installations of ACS systems, one at the Sydney Opera House Concert Hall, and the other at MBCCH, Winnepeg, Canada. Collaboration between the system designer, the musicians, and the acoustics consultant will be emphasized, along with techniques used to evaluate the systems' performance in the halls.

4:10

4pAAb8. Indoor outdoor acoustics: Active acoustics at the New World Center, Miami Beach Soundscape. Frederick R. Vogler (Sonitus, fred@sonitusconsulting.com), John Pellowe (Meyer Sound Labs, Inc.), and Steve Ellison (Meyer Sound Labs, Inc.)

The Frank Gehry designed New World Center is home to the New World Symphony in Miami Beach, FL. This facility includes an intimate 756 seat concert hall and is used as a training platform both for symphonic conductors and musicians. The adjacent park uses an active acoustics system to allow a similar number of people in the park to simultaneously experience the indoor concert experience in an open air environment. The immersive sound of the reproduced concert experience is accompanied by a 7000-square-foot projection wall that carries live video of the performance. The system captures the natural acoustic of the concert hall using microphones distributed throughout, and these signals are processed and then transmitted to the park utilizing a set of 160 distributed loudspeakers. The successful design, commissioning, and tuning of the system relied on a team approach between the architect, consultants, manufacturer, installer, and venue operators. Scope within the team is explored, challenges revealed, and suggestions offered to help ensure the success of new multi disciplinary ventures such as this. Similarities and differences to a surround sound broadcast transmission of the Los Angeles Philharmonic are also reviewed.

Contributed Papers

4:30

4pAAb9. Acoustical design of New World Center, Miami Beach, FL. Daniel F. Beckmann, Kayo Kallas, Motoo Komoda, and Yasuhisa Toyota (Nagata Acoust., 2130 Sawtelle Bl. Ste 308, Los Angeles, CA 90025, beckmann@nagata.co.jp)

The New World Center opened in January 2011 after an 8-year process to design and build the $160 million facility. The Frank Gehry-designed facility for the New World Symphony, "America's Orchestral Academy," has at its heart the 756-seat auditorium, designed in the arena style. Also in the facility are one large orchestral rehearsal room, seven medium-sized Chamber/Ensemble rehearsal rooms, and 24 individual coaching/practice rooms. Administrative offices and production spaces complement the spaces for music to bring the building to 100 641 square feet. The acoustical design of the 756-seat auditorium was performed in close collaboration with Gehry and the founder of the New World Symphony, acclaimed conductor Michael Tilson Thomas. To ensure validity of the acoustical design, a 1:24 scale model test was performed. Flexibility in rehearsal and performance was one of the prime requirements for the space as the program includes much more than just orchestral performance, which is met by a curtain system for variable acoustics, a highly flexible stage lift system, 247 retractable seats, and four alternate "Performance Platforms." Five large "sails," which double as projection surfaces, a proper ceiling above the stage, and a steeply raked audience area are amongst the acoustical design elements that are reported.

4:45

4pAAb10. Acoustical design of Soka University Performing Arts Center. Kayo Kallas, Motoo Komoda, and Yasuhisa Toyota (Nagata Acoust. America, 2130 Sawtelle Blvd., Ste. 308, Los Angeles, CA 90025, kimotsuki@nagata.co.jp)

The Soka University Performing Arts Center and Academic Building will open in September 2011. The $73 million performing arts center is open to the public, hosting various types of performing arts. For lovers of the performing arts, the center will become another choice among the many fine venues in Orange County. The building houses a 1000-seat multipurpose hall, a 150-seat black box theater, support spaces, and classrooms. The multipurpose hall was designed primarily as a concert hall, and later to become suitable for dance, plays and musicals. To satisfy these flexible programs, the design features a curtain system for variable acoustics and an automated stage lift to accommodate concert, thrust, and convocations stage settings. The seating layout of the multipurpose hall was arranged in the arena style. The room shape and interior materials were carefully selected to optimize the acoustics of the space, including the two layered ceiling design: one for aesthetics and the other for room acoustics. Acoustical design and characteristics of the new multipurpose hall will be reported.

5:00

4pAAb11. The effect of reverberation enhancement on the diffusion of the sound field. Hugh Hopper, David Thompson, and Keith Holland (I.S.V.R., University of Southampton, University Rd., Southampton, SO17 1BJ, UK)

Reverberation enhancement is a technology which allows the reverberation time of a room to be increased. It is important to consider the effect of this technology on the other measurable attributes of the room response. The spatial variation of steady state sound pressure level and reverberation time within the room can be used to measure the extent to which the room approximates a diffuse field. A theoretical value of these quantities can be predicted for an ideal diffuse field, and the ratio between the measured and theoretical values gives a normalized measure of the diffusion of the sound field. This work investigates the changes in these measures when reverberation enhancement is applied to a room. Experimental results have shown that the normalized measures of diffusion increase with the introduction of reverberation enhancement. This implies a reduction in the homogeneity and isotropy of the sound field which may be perceived as a reduction in subjective quality.

5:15

4pAAb12. Variable acoustics at hauppauge high school. Richard F. Riedel (Riedel Audio & Acoust., LLC, 443 Potter Blvd., Brightwaters, NY 11718)

The author will discuss how variable acoustic solutions were used to solve room anomalies and provide a means to control room size. Once plagued by a significant rear wall reflection that interfered with stage musicians' timing, due to its later arrival time, the author will explain how a unique custom designed retractable acoustical diffuser reduced this problem. Another recurring issue in the space was the need to reduce the size of the seating area, normally 1100 seats, to one which provided a more intimate setting for dramatic presentations, which were normally not attended by large audiences. Using a commercially available acoustical product, the issue was solved and a means provided to alter the room's acoustical environment. This paper presents details about the methods used to provide variation to the room's acoustics and cites specific measurements of the space with and without the variable acoustical elements.

5:30

4pAAb13. Sound absorbing vertically retracting drapery: A comparative study. Liz L. Lamour (Univ. of Kansas, School of Architecture, Design and Planning, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, lizlamour@gmail.com), Ben Bridgewater (Univ. of Kansas, Lawrence, KS 66045), and Ben Brooks (Univ. of Kansas, Lawrence, KS 66045)

The use of drapery as a variable sound absorbing material is widespread in theaters, university music halls, and other spaces where variable reverberation time is desired. As a class project at The University of Kansas, the authors and their classmates tested two types of sound absorbing vertically retracting drapery manufactured by a theatrical contracting firm using reverberant room methods. Fabric used in these two tests was cotton velour and acrylic velour. Coefficients of absorption were compared with coefficients of absorption published by a firm specializing in the fabrication of vertically retracting sound absorbing velour drapery. The measured and published data allowed both types of vertical retracting drapery to be specified for a multipurpose auditorium renovation project at a college in Kansas. The college auditorium renovation project will use the drapery fabricated by the specialty firm and the on-site determined coefficients of absorption will be presented and compared with the manufacturer's published data and also compared with the data obtained in the reverberant room for the somewhat different retracting drapery design by the theatre contracting firm. And if it proves to be possible, the on-site sound absorption measurements will be made at another college auditorium which uses cotton velour retracting drapery produced by the theatrical contracting firm. These data will be useful in specifying sound absorbing vertically retracting drapery for future projects.

THURSDAY AFTERNOON, 3 NOVEMBER 2011 PACIFIC SALON 1, 1:30 TO 3:15 P.M.

Session 4pAB

Animal Bioacoustics: Long-Term Acoustic Monitoring of Animals II

Simone Baumann-Pickering, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Marie A. Roch, Cochair
Dept. of Computer Science, San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720

Invited Papers

1:30

4pAB1. Diel and lunar variations of marine ambient sound in the North Pacific. Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Ana Širović (Scripps Inst. of Oceanogr., La Jolla, CA 92093), Marie A. Roch (San Diego State Univ., San Diego, CA 92182), Anne E. Simonis, Sean M. Wiggins (Scripps Inst. of Oceanogr., La Jolla, CA 92093), Erin M. Oleson (Pacific Islands Fisheries Sci. Ctr., NOAA, Honolulu, HI 96822), and John A. Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

Marine ambient sound was recorded on autonomous high-frequency acoustic recording packages (bandwidth 10 Hz to 100 kHz) during long term deployments at multiple sites across the North Pacific, from the high latitude Aleutian Islands to tropical Palmyra Atoll in depths of 600–1 000 m. Most intertropical but no temperate locations showed a distinct diel pattern in ambient sound. The soundscape at each location was unique, yet there was a similar recurring sound of unknown origin in lower latitude locations. This sound had a peak frequency around 3–5 kHz and was recorded only for several hours after sunset. Additionally, at some locations, a broadband acoustic signal with bandwidth up to 60 kHz was recorded at night with crepuscular peaks. Both sound patterns were lunar dependent with lower acoustic levels during full moon phases. Site-specific diel and seasonal acoustic patterns have been observed for various odontocete species. Correlations between odontocete presence and levels of ambient sound are investigated. [Work supported by NOAA-Pacific Islands Fisheries Science Center, US Navy-N45/PACFLT, ONR, Pacific Life, Ocean Foundation, University of California, San Diego.]

1:50

4pAB2. Long-term passive acoustic monitoring of nearshore ecosystems in the Northwestern Hawaiian Islands. Marc O. Lammers, Lisa Munger (Hawaii Inst. of Marine Biology, P.O. Box 1346, Kaneohe, HI 96744, lammers@hawaii.edu), Pollyanna Fisher Pool (Univ. of Hawaii, Honolulu, Hawaii), Kevin Wong (Pacific Islands Fisheries Sci. Ctr., Honolulu, Hawaii), Whitlow W. L. Au (Hawaii Inst. of Marine Biology, Kaneohe, HI 96744), and Russell E. Brainard (Pacific Islands Fisheries Sci. Ctr., Honolulu, Hawaii)

Monitoring the changing state of marine habitats in remote areas is, in most cases, a challenging task due to limited and/or infrequent opportunities to make direct observations. Passive acoustic monitoring is sometimes the best means of establishing long-term biological trends in such areas. Since 2006, an effort has been underway to monitor the neashore ecosystems of the Northwestern Hawaiian Islands (NWHI) using a network of Ecological Acoustic Recorders. A wide range of acoustic signals are being monitored to infer biological trends and to gauge the relative stability of the ecosystem. Among the variables measured are the acoustic activity of snapping shrimp, the incidence of cetaceans and the extent of spectral and temporal partitioning of the acoustic space by different taxa, measured as the "acoustic entropy" of the habitat. Multiyear time series of the different measures provide baseline levels of biological activity at each location and also reveal periods of anomaly. Observed trends are then examined for corollary relationships with oceanographic and meteorological parameters measured both in situ and remotely. The data obtained thus far are providing valuable insights that will help assess the long-term response of ecosystem in the NWHI to both natural and anthropogenic factors

2:10

4pAB3. Eavesdropping on coconut rhinoceros beetles, red palm weevils, Asian longhorned beetles, and other invasive travelers. Richard W Mankin (USDA-ARS-CMAVE, 1700 SW 23rd Dr., Gainesville, FL 32608, richard.mankin@ars.usda.gov)

As global trade increases, invasive insects inflict increasing economic damage to agriculture and urban landscapes in the United States yearly, despite a sophisticated array of interception methods and quarantine programs designed to exclude their entry. Insects that are hidden inside soil, wood, or stored products are difficult to detect visually but often can be identified acoustically because they produce 3–30-ms, 200–5 000-Hz impulses that are temporally grouped or patterned together in short bursts. Detection and analysis of these sound bursts enables scouts or inspectors to determine that insects are present and sometimes to identify the presence of a particular target species. Here is discussed some of the most successful acoustic methods that have been developed to detect and monitor hidden

insect infestations. Acoustic instruments are currently available for use in rapid surveys and for long-term monitoring of infestations. They have been useful particularly for detection of termites, coconut rhinoceros beetles, red palm weevils and Asian longhorned beetles in wood, white grubs and Diaprepes root weevil in soil, and stored product insects.

Contributed Papers

2:30

4pAB4. Acoustic monitoring of dolphin populations in the Gulf of Mexico. Kaitlin E. Frasier (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr. La Jolla, CA 92093, kefrasie@ucsd.edu), Melissa S. Soldevilla (Protected Resources and Biodiversity Div, NMFS/SEFSC, Miami, FL 33149), Mark A. McDonald (WhaleAcoust., Bellvue, CO 80512), Karlina P. Merkens, Sean M. Wiggins, John A. Hildebrand (Univ. of California, San Diego, La Jolla, CA 92093), and Marie A. Roch (San Diego State Univ., San Diego, CA 92182)

High-Frequency Acoustic Recording Packages (HARPs) continuously monitored delphinids at five sites in the northeastern Gulf of Mexico during and after the *Deepwater Horizon* oil spill. Surface oil reached two sites, while the three unexposed sites functioned as “controls.” Presence of dolphin vocalizations (clicks, whistles, and burst pulses) was documented at exposed and unexposed sites over the course of a year following the oil spill. These sites are within the known habitat ranges of 11 species of delphinids. Broadband towed array recordings with visual identifications were used to determine species-specific vocalization characteristics, which were then compared with autonomously recorded vocalizations. Two species have distinctive vocalizations that match between towed array and autonomous recordings. At least four more unique vocalization patterns were detected autonomously, which may be species-specific. Both clicks and whistles were explored for identifying features. The data provide a comparative view of delphinid presence relative to the oil spill.

2:45

4pAB5. Passive acoustic monitoring of sperm whales during and after the Deepwater Horizon oil spill. Karlina Merkens (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr. MC 0205, La Jolla, CA 92093-0205, kmerkens@ucsd.edu), Mark A. McDonald (Whale Acoust., Bellvue, CO 80512), Simone Baumann-Pickering, Kaitlin Frasier, Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0205)

The Deepwater Horizon oil spill during the summer of 2010 impacted a region of sperm whale habitat along the continental slope and deep waters of the Gulf of Mexico. Passive acoustic monitoring was used to study the potential impact of the oil spill on sperm whales by recording trends in their characteristic sounds, such as echolocation clicks and foraging creaks. High-frequency Acoustic Recording Packages (HARPs) were deployed shortly after the oil spill began; one was located close to the Deepwater Horizon well, above which the sea surface was contaminated by oil throughout the summer of 2010, and another was deployed in a region of sperm whale habitat that remained unexposed to surface oil to function as a “control” site. At both sites, sperm whales were detected on a majority of days during the nearly year-long recording period. Sperm whale presence was evaluated from detected clicks and creaks, and changes in these sounds over time and between sites were compared.

3:00

4pAB6. Long-term acoustic monitoring of marine mammal response to the 2010 oil spill in the Northern Gulf of Mexico. Natalia Sidorovskaia (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA 70504-4210, nas@louisiana.edu), Azmy S. Ackleh, Baoling Ma (Univ. of Louisiana at Lafayette, LA 70504), Christopher Tiemann (Univ. of Texas at Austin, Austin, TX 78713), George E. Ioup, and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA 70148)

The 2010 deep water horizon (DWH) oil spill in the Northern Gulf of Mexico brought a need for assessing spill impact and recovery timeline for the deepwater ecosystem, including marine mammals. Passive acoustics is emerging as a viable technology to monitor short-term and long-term abundance dynamics and to assess different factors that may cause an observed response. Multi-year pre-spill and post-spill acoustic data collected at different distances from the DWH incident site by the Littoral Acoustic Demonstration Center (LADC) are used to compare first-year oil spill response by three different groups of marine mammals: sperm whales, beaked whales, and dolphins. Densities of acoustic phonations by these animals are extracted from collected data and used for point estimates of the resident population density. As an example, a regional abundance estimate shows a decrease in the number of sperm whales at the site nearest to the DWH (9 mi away) which exceeds statistical uncertainties and can be accepted as an existing trend. The use of acoustic data to extract information about environmental factors, such as anthropogenic noise level or food call densities, that may contribute to the explanation of existing trends is also discussed. [Work is partially supported by NSF.]

Session 4pBA

Biomedical Acoustics: Therapy and Applications

Thomas J. Matula, Chair
Applied Physics Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698

Contributed Papers

1:30

4pBA1. Effects of physical properties of the skull on high intensity focused ultrasound for transcranial sonothrombolysis. Prashanth Selvaraj (Dept. of Mech. Eng., Etcheverry Hall, Univ. of California, Berkeley, CA pselvaraj@me.berkeley.edu), Kohei Okita (Ctr. for Intellectual Property Strategies RIKEN 2-1 Hirosawa, Wako, Saitama 351-0198, Japan), Yoichiro Matsumoto (Univ. of Tokyo, 7-3-1 Hongo, Bunkyo, Tokyo 113-8656, Japan), Arne Voie, Thilo Hoelscher (Univ. of California, San Diego, 212 West Dickinson St.), Hope Weiss, and Andrew J. Szeri (Etcheverry Hall, Univ. of California, Berkeley, CA)

The use of high intensity focused ultrasound (HIFU) in transcranial sonothrombolysis is emerging as a promising therapeutic intervention after stroke. Of interest in the present study is the evolution of the wave from transducer to focus, with special attention to two aspects. One is the attenuation of the wave before it reaches the focus, the other is the scattering of the wave at tissue interfaces leading to alteration of the focus. A code developed for tissue ablation (Kohei Okita, Kenji Ono, Shu Takagi, and Yoichiro Matsumoto, Int. J. Numer. Methods Fluids 65:43–66 (2011)), has been modified to study the effect of the physical properties of the skull on the focusing of the HIFU waves. Phase delay of the array transducer is employed to focus the waves. A basic model illustrative of the calvaria of the skull has been used as only the physical properties of the bone are of interest here. Microbubble cavitation has been shown to enhance sonothrombolysis; hence, the altered wave is examined from the point of view of the bubble dynamics it engenders.

1:45

4pBA2. Vascular permeability with targeted contrast agents—The effect of physiologically relevant dynamic shear stress. Pavlos Anastasiadis (Dept. of Mech. Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 302, HI 96822, pavlos@hawaii.edu), Joshua J. Rychak (Targeson, 3550 General Atomics Court, San Diego, CA 92121), and John S. Allen III (Dept. of Mech. Eng., Univ. of Hawaii at Manoa, HI 96822)

Targeted ultrasound contrast agents (UCAs) may be able to facilitate an early noninvasive diagnosis of atherogenesis. The coronary arterial branches might be routinely scanned for a clinical diagnosis since plaques often first form at disturbed flow regions within bifurcations. However, many outstanding questions exist on the targeting efficacy subject to pulsatile and potentially pathological flow in these areas. Targeted ultrasound contrast agents conjugated to the antibody of the intercellular adhesion molecule-1 (ICAM-1) are injected into y-shaped flow chambers that model the coronary heart artery bifurcations. Simultaneous measurements of the barrier function in physiologically relevant dynamic shear stresses are monitored with electric cell impedance (ECIS) measurements. The study aims at quantifying variations of the barrier function at regions with exposure to different shear stresses and correlating this information to the binding efficacy of targeted UCAs conjugated to the antibody of ICAM-1. These results are discussed with respect to reported targeted ultrasound contrast agent studies of inflammation and plaque in Apolipoprotein E-deficient mice.

2:00

4pBA3. Synergistic interaction between stress waves and cavitation is important for successful comminution of residual stone fragments in shock wave lithotripsy. Jaclyn Lautz, Georgy Sankin, and Pei Zhong (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708, jaclyn.lautz@duke.edu)

To assess the role of stress waves and cavitation in comminuting residual fragments during shock wave lithotripsy (SWL), cylindrical 4×4 mm BegoStone phantoms were treated in an electromagnetic lithotripter either at the focus ($z = 0$, $p_+ = \sim 45$ MPa) or pre-focally ($z = -30$ mm, $p_+ = \sim 24$ MPa). The treatment was performed with the stone immersed either in degassed water or in Butanediol, which has similar acoustic impedance to water but much higher viscosity to suppress cavitation. At the focus, the first fracture was observed after 26 ± 9 shocks, both in water and Butanediol ($p = 0.7$). However, when stones were moved pre-focally where comparable cavitation is produced (based on high speed imaging), the average shock number required for the initial fracture was increased to 66 ± 10 in water and 122 ± 20 in Butanediol ($p = 0.002$). Below -40 mm prefocally ($p_+ < 20$ MPa), stones did not fracture in water even after 2,000 shocks, although cavitation was observed. Furthermore, stone comminution at the focus after 250 shocks was $\sim$35% in water compared to $\sim$5% in Butanediol ($p < 0.001$). Altogether, these findings suggest that a synergistic interaction between stress waves and cavitation is critical in producing effective stone comminution during SWL. [Work supported by NIH and NSF GRFP.]

2:15

4pBA4. Investigation on the effect of specular reflections from stone surface on twinkling artifact. Wei Lu, Oleg A. Sapozhnikov, John C. Kucewicz, Bryan W. Cunitz, Peter J. Kaczkowski, Lawrence A. Crum, and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle WA 98105)

The twinkling artifact can highlight kidney stones during ultrasound color Doppler imaging with high sensitivity for stone detection. The mechanism of the twinkling artifact is still under debate. It was reported previously that twinkling appeared distal to the echogenic reflection from the stone surface in cases with no signal saturation. [Lu *et al.*, JASA **129**(4), p. 2376]. In this report, the effect of specular reflections on twinkling was investigated. Human kidney stones (5-9 mm in length) were embedded in a polyacrylamide gel phantom. Radio-frequency (RF) data were recorded from pulse-echo ensembles using a software-programmable ultrasound system. The variability within the beamformed Doppler ensemble, which is responsible for twinkling, was traced back to the unbeamformed RF channel data to identify whether variability arose disproportionately on channels receiving the specular reflection. The results showed that the specular reflection did not saturate individual channels and that the variability was observed on most channels with similar magnitude, which indicates that the appearance of twinkling does not rely on the specular reflection from the stone surface. Instead in the beamformer, the varying signals have the appearance of arising from a point source within the stone. [Work supported by NIH DK43881, DK086371, DK092197, and NSBRI through NASA NCC 9-58.]

2:30

4pBA5. Inhibition of breast cancer cell proliferation by low-intensity pulsed ultrasound (LIPUS). Amit Katiyar, Kausik Sarkar (Mech. Eng., Univ. Of Delaware, Newark, DE 19716), and Krishna Sarker (Biological Sci. , Univ. of Delaware, Newark, DE 19716)

Cancer is the second leading cause of death in the United States, preceded only by heart disease. Cancer cells display an uncontrolled proliferation, controlling which has been a big challenge for cancer treatment. Ultrasound is best known for its application in diagnostic imaging; it is also a vehicle for delivering high frequency mechanical stimulation toward beneficial bio-effects. Unlike high intensity focused ultrasound, which is recently being investigated for thermal ablation of solid tumors, low intensity pulsed ultrasound (LIPUS) is directed toward cellular mechanisms. The effects of LIPUS on cancer cell proliferation are not known. Here, we demonstrate that LIPUS dose-dependently inhibits proliferation of breast cancer cell T47D as determined by several biochemical assays such as MTS, Alamar Blue, and BrdU assay. Statistically significant inhibition of T47D cell proliferation is observed when cells are exposed to 50–100 mW/cm^2. For this intensity range, LIPUS excitation inhibits the proliferation of T47D cells upto 50%. We also notice that inhibition of cell proliferation by LIPUS depends on its exposure time on cells. Minimum exposure time of LIPUS excitation for pronounced inhibitory effects on T47D cell proliferation is approximately 10 min.

2:45

4pBA6. Acoustical assessment of body water balance. Armen Sarvazyan (Artann Labs., 1459 Lower Ferry Rd., Trenton, NJ 08618)

A new medical application of acoustics has recently emerged: assessment of body hydration status by ultrasonic measurement of muscle water content. The need for an easy-to-perform method for the detection of water imbalance is of the utmost clinical importance. Body hypohydration may cause severe health and performance problems, decreasing cognitive and physical work capabilities, while excessive hydration is a common symptom of many other diseases. The speed of longitudinal acoustic waves in muscle, as well as in other soft tissues, is defined by tissue molecular composition because both the density and bulk compressibility of tissue depend mainly on short range molecular interactions. Skeletal muscle is the largest water compartment in the body; it comprises 40% of body mass and 75% of muscle is water. The ultrasound velocity in muscle is a linear function of water content with the slope of about 3.0 m/s per 1% change in water content. We will describe the design, measurement principles, and testing results for new acoustic devices for assessment of hydration status of elderly and infants, two most vulnerable groups of population. Advantages and disadvantages of acoustical method of hydration over currently available methods are discussed.

3:00

4pBA7. Use of highly nonlinear solitary waves for the assessment of dental implants. Bruk Berhanu (942 Benedum Hall, Dept. of Civil and Environ. Eng., 3700 O'Hara St., Pittsburgh, PA 15261, bruk.berhanu@gmail.com)

This paper presents a noninvasive technique based on the propagation of highly nonlinear solitary waves (HNSWs) to monitor the stability of dental implants. HNSWs are mechanical waves that can form and travel in highly nonlinear systems, such as one-dimensional chains of contacting spherical particles (i.e., granular crystals). In this study, a granular crystal-based actuator/sensor, designed and built at the University of Pittsburgh, was used to introduce HNSWs into dummy implants that were inserted into either hardened plaster or treated beef bones. The waves reflected at the interface between the particle and implant were monitored to estimate the change in stiffness of the material. The hydration of the plaster was monitored because it can be considered largely similar to the osseointegration process that occurs in the oral connective tissue once a dental-endosteal threaded implant is surgically inserted. In the experiment using bone, the implant-bone system was immersed in an acid bath causing decalcification of the bone and, therefore, reduced stiffness of the bone itself, simulating the inverse of osseointegration. Positive correlations were found, in both experiments, between certain properties of the HNSWs and the stiffness of the test object, demonstrating that HNSWs show promise for use in assessment of dental implants.

3:15

4pBA8. Artificial flesh material selection for hearing protection evaluation system. Mehmet M. A. Bicak, Josiah M. Oliver, and Kevin R. Shank (Adaptive Technologies. Inc., 2020 Kraft Dr. Ste. 3040, Blacksburg, VA 24060)

Developing an advanced hearing protection evaluation system (HPES) in the form of an acoustic test fixture (ATF) allows for characterizing either circumaural or insert-type hearing protection devices (HPDs) in both impulse and continuous noise environments across the dynamic range of human hearing. This is a challenging task since the acoustical transfer paths through flesh contribute to the dynamic response of the system. Current ATFs do not account for the transfer paths through flesh to ear canal. In this study, we investigated several visco-elastic flesh materials numerically using coupled vibro-acoustic simulations, and experimentally using vibration and acoustic excitation methods. Geometrically representative prototypes are being developed using volume computed tomography (VCT) that include detailed features of the skull and flesh structure, so that flesh conducted sound transmission paths can be physically modeled. The HPD on ATF dynamic behavior is compared with the HPD on subject behavior using finite element simulation models developed using the VCT images. The material selection is validated using noise reduction and vibration experiments on the subjects.

3:30

4pBA9. Cough count as a marker for patient recovery from pulmonary tuberculosis. Brian H. Tracey (ECE Dept., Tufts Univ., 196 Boston Ave., Rm., 4330 Medford, MA 02155, brian.tracey@tufts.edu), German Comina (Laboratorio de Ingeniería Fsica, Facultad de Ciencias, Universidad Nacional de Ingeniería, Rimac, Lima, Per), Sandra Larson (Michigan State Univ., College of Osteopathic Medicine, East Lansing, MI), Marjory Bravard (Massachusetts General Hospital, Boston, MA 02114), Jose W. Lopez (Unidad de Epidemiología, Hospital Nacional Dos de Mayo, Lima, Per), and Robert H. Gilman (Johns Hopkins Bloomberg School of Public Health, Baltimore, MD)

In regions of the world where tuberculosis (TB) poses the greatest disease burden, clinicians often lack access to skilled laboratories. This is particularly problematic for patients with drug-resistant tuberculosis, as these patients will otherwise receive standard TB medication and will not respond to treatment. Thus, a lab-free method for assessing patient recovery during treatment would be of great benefit. We hypothesize that cough analysis may provide such a test, and have carried out a pilot study to record coughs from a cohort of patients in Lima, Peru. We describe algorithm development for cough data analysis and compare several event detection and classification strategies. Results from our ongoing validation efforts suggest that cough count (cough/hour) decreases noticeably after the start of treatment in drug-responsive patients. Our long-term goal is development of a low-cost ambulatory cough analysis system that will help identify patients with drug-resistant tuberculosis.

Session 4pEA

Engineering Acoustics and Underwater Acoustics: Vector Sensors, Projectors, and Receivers II: Receivers, Reception, and Transmission

Stephen C. Butler, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02840

Roger T. Richards, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02840

Invited Papers

1:05

4pEA1. Acoustic intensity vector probes. Gary W. Elko (Mh Acoust. LLC, 25A Summit Ave., Summit, NJ 07901)

Modern acoustic intensity measurement techniques began in the early 1980's with the realization that the imaginary part of the cross-spectral density was directly related to the active intensity component along the axis between two closely-spaced pressure microphones. Since acoustic intensity is a vector quantity, it was obvious that one would like to measure all three orthogonal components and to graphically represent acoustic power flow through space. Needing an interesting topic for a Ph.D. thesis at Penn State (and a way of funding it ..., thank-you US Navy), the task of investigating and developing the estimation of the acoustic intensity vector field fortunately came my way. This talk will present some of the early interesting and fun things that came out of working on in-air acoustic vector sensors. It will conclude with some more recent developments that have direct connections to the early acoustic vector probes that were built, tested, and used at Penn State.

1:25

4pEA2. Vector sensors for airborne surveillance applications. James McConnell, Scott Jensen, Thomas McCormick, and Brendan Woolrich (Appl. Physical Sci. Corp., 475 Bridge St., Groton CT 06340)

The use of vector sensors for airborne surveillance applications (e.g., frequencies below 500 Hz) is discussed with emphasis on transducers that measure the acoustic pressure-gradient. Traditional approaches such as ribbon microphones, hot-wire anemometers, and finite-difference techniques will be reviewed. The crux of the presentation concerns a discussion of diffraction type pressure-gradient microphones that go beyond the classic ribbon microphone and utilize large format membrane transduction elements comprised of piezoelectric and electret materials. The results of analytical, numerical, and experimental evaluations will be presented.

1:45

4pEA3. Estimation of sea floor properties using acoustic vector sensors. Steven E. Crocker (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841), James H. Miller, John C. Osler (NATO Undersea Res. Ctr., La Spezia, Italy), Gopu R. Potty (Univ. of Rhode Island, Narragansett, RI 02882), and Paul C. Hines (DRDC Atlantic, Dartmouth, NS B2Y3Z7, Canada)

Information in the acoustic vector field can be used to estimate properties of the environment with which the field interacts. A study was performed to understand the value of vector field data when inverting for sea floor geoacoustic properties. The study compared results obtained with new inverse methods based on measurements of complex acoustic transfer functions and specific acoustic impedance. Acoustic field data consisted of gated continuous wave transmissions acquired with four acoustic vector sensors that spanned the water-sediment interface during the Sediment Acoustics Experiment 2004 (SAX04). Motion data provided by the buried vector sensors were affected by a suspension response that was sensitive to the sediment density and shear wave speed. The suspension response for the buried vector sensors included a resonance within the analysis band of 0.6–2.4 kHz. The response was sufficiently sensitive to the local geoacoustic properties, that it was exploited by the inverse methods developed for this study. Inversions of real and synthetic data sets indicated that information about sediment shear wave speed was carried by the suspension response of the buried sensors, as opposed to being contained inherently within the acoustic vector field. [Work supported by ONR.]

2:05

4pEA4. Comparing vector- and pressure-sensor arrays: The 2009 cooperative array performance experiment. Daniel Rouseff, Russell Light (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105), Zhongkang Wang, and Shihong Zhou (Hangzhou Appl. Acoust. Res. Inst., Hangzhou, China)

The 2009 cooperative array performance experiment (CAPE'09) was designed to compare performance between vector- and pressure-sensor arrays. The experiment was a joint effort of Chinese and American investigators; both arrays were designed and assembled by the Hangzhou Applied Acoustics Research Institute (HAARI), while the source systems and signal processing/recording systems were designed and assembled by Applied Physics Laboratory, University of Washington (APL-UW). The two arrays, both approximately 7 m in length, were deployed vertically off the stern of the APL-UW's R/V Robertson in Lake Washington, Seattle. Various transmitted signals in the 1.5–4 kHz band were recorded simultaneously on the two arrays at ranges between 10 m and 4 km. The signals included repeated linear frequency-modulated chirps and communications sequences. The pressure- and vector-sensor arrays had 32 and 8 uniformly spaced elements, respectively. Because each element in the vector-sensor array recorded both pressure and the three components of particle velocity, the two arrays made the same number of measurements over a similar vertical aperture. In the present talk, the design features of the vector-sensor array are emphasized. Sample results for both arrays are presented. [Work supported by ONR.]

2:25

4pEA5. Acoustic particle velocity amplification with horns. Dimitri M. Donskoy (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030)

Previously [Donskoy and Cray, J. Acoust. Soc. Am., **129**(4), Pt. 2, 2644 (2011)] the authors numerically investigated an acoustic particle velocity amplification effect with conical open-ended horns. Here, using Webster's approach, an analytical solution for the particle velocity response of conical horns (single and double) is derived and analyzed. The solutions are verified by comparison with direct numerical computations and supported with experimental measurements. It is shown that small horns, compared to the acoustic wavelength, are capable of providing substantial particle velocity amplification. For example, a 20 cm horn, in-water, can deliver nearly 10 dB of amplification over a very broad frequency range (from zero to 1000 Hz) without significant amplitude and phase distortion. Another unique feature of the velocity horn is its dipole directionality. The paper presents a thorough analysis of horn's amplification versus geometrical parameters. [This work is supported in part by the ONR summer fellowship program].

Contributed Papers

2:45

4pEA6. Analysis of the advantages and complexities of acoustic vector sensor arrays. Andrew J. Poulsen (Appl. Physical Sci. Corp., 49 Waltham St., Ste. 4, Lexington, MA 02421, apoulsen@aphysci.com) and Arthur B. Baggeroer (MIT, Cambridge, MA 02139)

The hydrophone, an omnidirectional underwater microphone, is the most common sensor for listening to underwater sound. Directional sensors, however, have many important applications. Acoustic vector sensors, one important class of directional sensors, measure acoustic scalar pressure along with acoustic particle motion. With this additional vector measurement, vector sensors feature many advantages over conventional omnidirectional hydrophone sensors: improved array gain/detection performance, enhanced bearing resolution, the ability to "undersample" an acoustic wave without spatial aliasing, and the capability of attenuating spatial ambiguity lobes, e.g., left/right ambiguity resolution for a linear array. Along with their advantages, however, vector sensors also pose additional practical complexities: greater sensitivity to non-acoustic, motion-induced flow noise at low frequencies, requisite knowledge/measurement of each sensor's orientation, management of different sensor types (pressure and particle motion) that each with different noise properties/calibration requirements, and adaptive processing can become difficult in a snapshot limited regime since each vector sensor is made up of up to four data channels. This paper will explore the virtues and limitations of vector sensor arrays in the presence of realistic ocean noise fields and system imperfections, including their effects on array performance (gain, beampatterns, etc.) supported by theoretical analysis and illustrative examples.

3:00

4pEA7. Vector wave measurements on landmine detection with an array of loudspeakers focused on the ground. Martin L. Barlett, Justin D. Gorhum, Wayne M. Wright, Mark F. Hamilton, and Thomas G. Muir (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

An array of 16 loudspeakers, deployed along a segment of the base of a right circular cone, was used to focus sound on soils overlying buried targets lying along the conical axis of the source. Measurements were made at several incident angles tending toward grazing to examine long range detection for humanitarian de-mining applications. Targets and soils were instrumented with triaxial geophone and accelerometer sensors. The transmission of airborne sound into the soils produced vertical and radial vibrations in both the soil and the targets, which included rigid and compliant mine cases. Several waveform transmission types were utilized. The compliant targets provided resonances amenable to optical detection, depending on the acoustic, geometric and environmental parameters, which are discussed. [Work supported by the IRD program at ARL:UT Austin, in cooperation with the National Center for Physical Acoustics.]

3:15

4pEA8. Capacity and statistics of measured underwater acoustic particle velocity channels. Huaihai Guo, Chen Chen, Ali Abdi (Elec. & Comput. Eng. Dept., New Jersey Inst. of Technol., Newark, NJ 07102, hg45@njit.edu), Aijun Song, Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE 19716), and Paul Hursky (Heat, Light, and Sound Res., Inc., La Jolla, CA 92037)

Acoustic particle velocity channels can be used for communication in underwater systems [A. Abdi and H. Guo, IEEE Trans. Wireless Communi. **8**, 3326–3329, (2009)]. In this paper, the information (Shannon) capacity of underwater acoustic particle velocity channels is studied using measured data. More specifically, the maximum achievable data rates of a compact vector sensor communication receiver and another communication receiver with spatially separated scalar sensors are compared. Some statistics of particle velocity channels such as amplitude distribution and power delay profile are investigated using measured data and proper models are suggested as well. The results are useful for design and simulation of vector sensor underwater communication systems in particle velocity channels. The work is supported in part by the National Science Foundation (NSF), Grant CCF-0830190.

Session 4pNS

Noise and Physical Acoustics: Launch Vehicle Noise II

R. Jeremy Kenny, Cochair
Marshall Space Flight Center, Bldg. 4203, Huntsville, AL 35812

Kent L. Gee, Cochair
Dept. of Physics and Astronomy, Brigham Young Univ., N243 Eyring Science Center, Provo, UT 84602

Invited Papers

1:10

4pNS1. Overview of the Ares I scale model test program. Douglas D. Counter (Bldg. 4203, M.S. ER42, Marshall Space Flight Ctr., Huntsville, AL 35812, douglas.d.counter@nasa.gov) and Janice Houstion (Jacobs Eng., Huntsville, AL 35812)

Launch environments, such as lift-off acoustic (LOA) and ignition overpressure (IOP), are important design factors for any vehicle and are dependent upon the design of both the vehicle and the ground systems. LOA environments are used directly in the development of vehicle vibro-acoustic environments and IOP is used in the loads assessment. The NASA Constellation Program had several risks to the development of the Ares I vehicle linked to LOA. The risks included cost, schedule, and technical impacts for component qualification due to high predicted vibro-acoustic environments. One solution is to mitigate the environment at the component level. However, where the environment is too severe for component survivability, reduction of the environment itself is required. The Ares I scale model acoustic test (ASMAT) program was implemented to verify the Ares I LOA and IOP environments for the vehicle and ground systems including the mobile launcher (ML) and tower. An additional objective was to determine the acoustic reduction for the LOA environment with an above deck water sound suppression system. ASMAT was a development test performed at the Marshall Space Flight Center (MSFC) East Test Area (ETA) Test Stand 116 (TS 116). The ASMAT program is described in this presentation.

1:30

4pNS2. Ares I Scale Model Acoustic Tests Instrumentation for Acoustic and Pressure Measurements. Magda B. Vargas (Bldg. 4203, M.S. ER42, MSFC, Huntsville, AL 35812, magda.b.vargas@nasa.gov) and Douglas D. Counter (MSFC, Huntsville, AL, 35812)

The Ares I Scale Model Acoustic Test (ASMAT) was a development test performed at the Marshall Space Flight Center (MSFC) East Test Area (ETA) Test Stand 116. The test article included a 5% scale Ares I vehicle model and tower mounted on the Mobile Launcher. Acoustic and pressure data were measured by approximately 200 instruments located throughout the test article. There were four primary ASMAT instrument suites: ignition overpressure (IOP), lift-off acoustics (LOA), ground acoustics (GA), and spatial correlation (SC). Each instrumentation suite incorporated different sensor models which were selected based upon measurement requirements. These requirements included the type of measurement, exposure to the environment, instrumentation check-outs and data acquisition. The sensors were attached to the test article using different mounts and brackets dependent upon the location of the sensor. This presentation addresses the observed effect of the sensors and mounts on the acoustic and pressure measurements.

1:50

4pNS3. Measurements of the ground acoustic environments for small solid rocket motor firings. Bruce T. Vu (NASA Kennedy Space Ctr., NE-M1, KSC, FL 32899, Bruce.T.Vu@nasa.gov) and Kenneth J. Plotkin (Wyle Labs., Arlington, VA 22202, Kenneth.Plotkin@wyle.com)

During the ground launch of a space vehicle, the mobile launcher deck and tower are exposed to severe acoustic environments. These environments, if not properly managed, can weaken ground support equipment and result in structure failure. The ground acoustic environments are different than the vehicle acoustic environments. They are typically more severe because of the close proximity of the rocket plume, which often involves direct impingement. They are more difficult to predict, and their measurement and data reduction remain challenging. This paper discusses these challenges and describes the methods of processing ground acoustic data during a series of static firings of a 5-percent scale solid rocket launch vehicle and mobile launcher, known as the Ares Scale Model Acoustic Test.

2:10

4pNS4. Ares I scale model acoustic test lift-off acoustics. Douglas D. Counter (MSFC, M.S. ER42, Huntsville, AL, 35812, douglas.d.counter@nasa.gov) and Janice Houston (Jacobs Egnineering, Huntsville, AL, 35812)

The lift-off acoustic (LOA) environment is an important design factor for any launch vehicle. For the Ares I vehicle, the LOA environments were derived by scaling flight data from other launch vehicles. The Ares I LOA predicted environments are compared to the Ares I scale model acoustic test (ASMAT) preliminary results.

2:30

4pNS5. Ares I Scale Model Acoustic Test above deck water sound suppression results. Douglas D. Counter (MSFC, M.S. ER42, Huntsville, AL 35812, douglas.d.counter@nasa.gov) and Janice Houston (Jacobs Eng., Huntsville, AL 35812)

The Ares I Scale Model Acoustic Test (ASMAT) program test matrix was designed to determine the acoustic reduction for the LOA environment with an above deck water sound suppression system. The scale model test can be used to quantify the effectiveness of the water suppression system as well as to optimize the systems necessary for LOA noise reduction. Several water flow rates were tested to determine which rate provides the greatest acoustic reductions. Preliminary results are presented.

2:50

4pNS6. Recovering the spatial correlation of liftoff acoustics from the Ares Scale Model Acoustics Test. Brian Prock, Paul Bremner (ATA Eng., Inc.), and Thomas L. Philley (NASA JSC)

Accurately predicting structural vibrations due to acoustic loads requires knowledge about the overall sound pressure levels, the amplitude of the sound pressure levels as a function of frequency (auto-spectra), and the spatial correlation of the sound pressure levels (cross-spectra). When dealing with the liftoff acoustics of launch vehicles, a large amount of historical data is available in terms of overall levels and auto-spectra, but only a limited amount of data exists for cross-spectra. ATA Engineering has used data taken during NASAs Ares Scale Model Acoustic Test to recover the spatial correlation of liftoff acoustics for a typical launch vehicle. By assuming the measured liftoff acoustics that are a combination of propagating waves and diffuse acoustic fields, curve-fitting algorithms are used to recover spatial correlation parameters required by modern vibro-acoustic analysis software.

3:10–3:30 Break

3:30

4pNS7. 5% Ares I scale model acoustic test: Overpressure characterization and analysis. David Alvord (Jacobs Eng., M.S. ER42, MSFC, Huntsville, AL 35812, david.alvord@nasa.gov), Matthew Casiano, and David McDaniels (MSFC, Huntsville, AL, 35812)

During the ignition of a ducted solid rocket motor (SRM), rapid expansion of injected hot gases from the motor into a confined volume causes the development of a steep fronted wave. This low frequency transient wave propagates outward from the exhaust duct, impinging the vehicle and ground structures. An unsuppressed overpressure wave can potentially cause modal excitation in the structures and vehicle, subsequently leading to damage. This presentation details the ignition transient findings from the 5% Ares I scale model acoustic test (ASMAT). The primary events of the ignition transient environment induced by the SRM are the ignition overpressure (IOP), duct overpressure (DOP), and source overpressure (SOP). The resulting observations include successful knockdown of the IOP environment through use of a space shuttle derived IOP suppression system, a potential load applied to the vehicle stemming from instantaneous asymmetrical IOP and DOP wave impingement, and launch complex geometric influences on the environment. The results are scaled to a full-scale Ares I equivalent and compared with heritage data including Ares I-X and both suppressed and unsuppressed space shuttle IOP environments.

3:50

4pNS8. Simulation of acoustics for Ares I scale model acoustic tests. Gabriel C. Putnam and Louise L. Strutzenburg (Marshall Space Flight Ctr., MSFC / ER42, Huntsville, AL, 35812)

The Ares I scale model acoustics test (ASMAT) is a series of live-fire tests of scaled rocket motors meant to simulate the conditions of the Ares I launch configuration. These tests have provided a well documented set of high fidelity acoustic measurements useful for validation including data taken over a range of test conditions and containing phenomena like ignition over-pressure and water suppression of acoustics. To take advantage of this data, a digital representation of the ASMAT test setup has been constructed and test firings of the motor have been simulated using the LOCI/CHEM computational fluid dynamics software. Results from ASMAT simulations with the rocket in both held down and elevated configurations, as well as with and without water suppression have been compared to acoustic data collected from similar live-fire tests. Results of acoustic comparisons have shown good correlation with the amplitude and temporal shape of pressure features and reasonable spectral accuracy up to approximately 1000 Hz. Major plume and acoustic features have been well captured including the plume shock structure, the igniter pulse transient, and the ignition overpressure.

4:10

4pNS9. Hybrid computational fluid dynamics and computational aero-acoustic modeling for liftoff acoustic predictions. Louise L. Strutzenberg (MSFC, M.S. ER42, MSFC, Huntsville, AL 35812, louise.s@nasa.gov) and Peter A. Liever (CFD Res. Corp., Huntsville, AL 35812)

This paper presents development efforts at the NASA Marshall Space flight Center to establish a hybrid computational fluid dynamics and computational aero-acoustics (CFD/CAA) simulation system for launch vehicle liftoff acoustics environment analysis. Acoustic prediction engineering tools based on empirical jet acoustic strength and directivity models or scaled historical measurements are of limited value in efforts to proactively design and optimize launch vehicles and launch facility configurations for liftoff acoustics. CFD based modeling approaches are now able to capture the important details of vehicle specific plume flow environment, identify the noise generation sources, and allow assessment of the influence of launch pad geometric details and sound mitigation measures such as water injection. However, CFD methodologies are numerically too dissipative to accurately capture the propagation of the acoustic waves in the large CFD models. The hybrid CFD/CAA approach combines the high-fidelity CFD analysis capable of identifying the acoustic sources with a fast and efficient boundary element method (BEM) that accurately propagates the acoustic field from the source locations. The BEM approach was chosen for its ability to properly account for reflections and scattering of acoustic waves from launch pad structures. The paper will present an overview of the technology components of the CFD/CAA framework and discuss plans for demonstration and validation against test data.

4:30

4pNS10. Validation study on computational aeroacoustics of acoustic waves from sub-scale rocket plumes. Seiichiro Morizawa (Dept. of Aerosp. Engineering, Tohoku Univ., 3-1-1 Yoshinodai, Chuo-ku, Sagamihara 25205210, morizawa@edge.ifs.tohoku.ac.jp), Taku Nonomura (Inst. of Space and Astronautical Sci., Sagamihara 2525210), Seiji Tsutsumi (JAXA's Eng. Deigital Innovation, JAXA, Sagamihara 2525210), Nobuhiro Yamanishi, Keita Terashima (Space Transportation Mission Directorate, Tsukuba 3058505), Shigeru Obayashi (Tohoku Univ., 2-1-1 Katahira Aoba-ku Sendai, 9808577), and Kozo Fujii (Inst. of Space and Astronautical Sci., Sagamihara 2525210)

In this paper, the comparative study of prediction based on computational aeroacoustics (CAA) and experimental results for acoustic waves from modeled rocket motors is conducted, and prediction accuracy of CAA is discussed in the framework of JAXA-CNES collaboration. Experimental data of flow and acoustic fields of solid motor by JAXA and H2-AIR liquid motor by CNES are used as the reference. Two types of computational codes are adapted in this study. The predictions of sound pressure level by both computational codes agree reasonably with corresponding experimental data, whereas the errors are approximately less than 5 dB. In addition, each aeroacoustic field of CAA results in this study is discussed in detail.

Contributed Papers

4:50

4pNS11. Measurement and propagation of supersonic aeroacoustic noise sources using continuous scanning measurement technologies and the fast multipole boundary element method. Michael Y. Yang, Havard Vold, and Parthiv N. Shah (11995 El Camino Real, San Diego, CA 92130)

ATA Engineering has developed a technique which uses a continuous scanning robot to take high-resolution measurements of supersonic jet plumes. The jet noise was modeled using a reduced-order model and propagated to far field microphone locations in the free-field. It is shown that the pressure at these microphones was successfully reconstructed across a range of frequencies. The capability to make predictions when scattering surfaces are present is also demonstrated using the fast multipole boundary element method in VA One. This work was originally designed for supersonic jets but can also be used for static firing tests of launch vehicle engines. The measured data could then be used for analytic predictions of the liftoff environment.

5:05

4pNS12. On the jet-wake similarity. Ballard W. George (1367 Boblink Circle, Sunnyvale, CA 94087)

This paper is concerned with similarities and differences between jets and wakes, as indicated by a sampling of the literature, including what tests and studies have been made in each case. As noted by Franken, jets and wakes are both characterized by a region of shear and (depending on the Reynolds number) high turbulence. Jet noise has been extensively studied under the impetus of a strong concern with community noise. Jet noise studies have examined numerous quantities including, for example, spectra, directivity, source location, and total power. Lighthill published widely quoted papers on sound generated aerodynamically, primarily in the context of jet noise and without solid boundaries. Literature referred to for this paper was primarily for airborne sound and included underwater sound, in which case cavitation plays a significant role and tends to mask turbulent wake noise due to the vehicle body. Propellers are a source of wake as well as lift noise. A readily noticeable difference between jets and wakes involves the fact that jets can be tested statically, while for wakes there has to be relative motion. Also jets, though not jet engines, can be tested "cold."

THURSDAY AFTERNOON, 3 NOVEMBER 2011 ESQUIRE, 1:55 TO 4:45 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Perceptual Aspects of Sound

Elizabeth A. Strickland, Chair
Dept. of Speech, Language, and Hearing Sciences, Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907

Chair's Introduction—1:55

Contributed Papers

2:00

4pPP1. Mistuned harmonic detection and the role of tonotopically local neural synchrony. William M. Hartmann (Depatment of Phys. and Astronomy, Michigan State Univ., East Lansing, MI 48824)

A mistuned harmonic in an otherwise periodic complex tone is a simple stimulus for the study of segregation and integration of complex tones by human listeners. In a mistuned harmonic *detection* experiment, a listener must discriminate between two tones—one perfectly periodic and the other with a mistuned harmonic. This detection paradigm is subject to known artifacts. However, with careful experiment design the artifacts can be controlled, and the paradigm becomes an efficient way to explore the emergence of segregated mistuned spectral components. The results of mistuned harmonic detection experiments support a tonotopically local model in which detection is mediated by the dephasing of a neural spike sequence in a tonotopically tuned channel. Evidence for this model comes from (1) the functional dependences of detectability on the amount of mistuning and the tone duration, (2) the non-monotonic level dependence of detectability, (3) the lowpass character of detection—indicating an essential role for neural synchrony, and (4) the need for tonotopically local interaction as evidenced by experiments using mistuned distracters, mistuned harmonics in spectral gaps, and dichotic presentation. [Work supported by the NIDCD and the AFOSR.]

2:15

4pPP2. Parametric issues in measuring the olivocochlear reflex with a masking technique. Elin M. Roverud and Elizabeth A. Strickland (Dept. of Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr. West Lafayette, IN 47907, eroverud@purdue.edu)

Previous studies in this laboratory have suggested that forward masking occurs by two mechanisms, excitatory masking and gain reduction via the medial olivocochlear reflex (MOCR), that operate over different time courses. In this laboratory, a forward masking technique is used in which a precursor, intended to elicit the MOCR, is followed by a fixed off-frequency forward masker and signal. A prior study [Roverud and Strickland, J. Acoust. Soc. Am. **128**, 1203–1214] found that for short precursors, signal threshold increased then decreased (buildup) as the signal and masker were delayed from the precursor. This result could be consistent with the sluggish onset of the MOCR. When the masker is removed, however, buildup is not observed as the signal is delayed from a precursor, even though the precursor should still be eliciting the MOCR. This raises the question, what is the role of the masker and why is it necessary to observe buildup? The current study examines the frequency and level characteristics of the masker required to observe the buildup effect. Results will be discussed in terms of gain reduction and temporal window models of forward masking. [Research supported by a Grant from NIH(NIDCD) R01 DC008327.]

2:30

4pPP3. Perceiving auditory distance using level and direct-to-reverberant ratio cues. Andrew J. Kolarik, Silvia Cirstea, and Shahina Pardhan (VERU, Eastings 204, Anglia Ruskin Univ., East Rd., Cambridge, CB1 1PT, United Kingdom, andrew.kolarik@anglia.ac.uk)

The study investigated how level and reverberation cues contribute to distance discrimination, and how accuracy is affected by reverberation cue strength. Sentence pairs were presented at distances between 1 and 8 m in a virtual room simulated using an image-source model and two reverberation settings (lower and higher). Listeners performed discrimination judgments in three conditions: level cue only (Level-Only), reverberation only (Equalized), and both cues available (Normal). Percentage correct judgment of which sentence was closer was measured. Optimal distance discrimination was obtained in the Normal condition. Perception of the difference in distance between sentences had a lower threshold (i.e., performance was significantly better, $p<0.05$) for closer than further targets in Normal and Level-Only conditions. On the contrary, in the Equalized condition, these thresholds were lower for further than closer targets. Thresholds were lower at higher reverberation in the Equalized condition, and for further targets in the Normal condition. Data indicate that level generally provided more accurate discrimination information than direct-to-reverberant ratio. Direct-to-reverberant ratio provided better information for sounds further from the listener than for nearer sounds, and listeners were able to use direct-to-reverberant ratio as effectively as level in highly reverberant rooms when discriminating far sound sources.

2:45

4pPP4. Infants' vowel discrimination in modulated noise. Lynne A. Werner (Dept. Speech & Hearing Sci., Univ. Washington, 1417 NE 42nd St., Seattle WA 98105-6246)

Adult listeners detect and discriminate target sounds better in amplitude modulated noise than in unmodulated noise. This study examined infants' ability to take advantage of masker modulation to improve sensitivity to a target. Listeners were 7–9-month-old infants and 18–30-year-old adults. Vowel discrimination in noise was tested. Listeners learned to respond when a repeated vowel changed from /a/ to /i/ or from /i/ to /a/. An observer-based method was used to assess sensitivity to the vowel change. The maskers were speech-spectrum noise either unmodulated, amplitude modulated with the envelope of single-talker speech or sinusoidally amplitude modulated at 8 Hz with a 75% modulation depth. The overall level of the maskers was 60 dB SPL. The level of the vowels, chosen to yield an average d' of 1 in the unmodulated masker, was 46 dB SPL for adults and 58 dB SPL for infants. Adults' d' was substantially and significantly higher in both modulated maskers than in the unmodulated masker. Infants' d' was also significantly higher in the two modulated maskers than in the unmodulated masker, but the improvement due to modulation was significantly smaller for infants than for adults. [Work supported by NIDCD R01DC00396 and P30DC04661.]

3:00

4pPP5. Effects of silent interval on human frequency-following responses to voice pitch. Fuh-Cherng Jeng and Ronny P. Warrington (Commun. Sci. and Disord., Ohio Univ., 1 Ohio Univ. Dr. Athens, OH 45701, jeng@ohio.edu)

Human frequency-following responses (FFRs) to voice pitch have provided valuable information on how the human brain processes speech information. Recordings of the FFR to voice pitch, however, may overlap when insufficient silent intervals are used. To determine the shortest silent interval that can be used with no overlap between adjacent response waveforms, FFRs were recorded from 12 Chinese adults using a wide range of silent intervals. The stimulus token was a Chinese monosyllable with a rising pitch of 117–166 Hz and a duration of 250 ms. A high stimulus intensity at 70 dB SPL was used to maximize overlaps in the response waveforms. A total of seven silent intervals, ranging from the full length of the stimulus duration down to approximately half period of the fundamental frequency of the stimulus token, were administered at a random order across participants. Two distinct methods (Hilbert transform and root-mean-square amplitudes) were used to delineate the envelopes and overlaps of the response waveforms. A one-way repeated measures analysis of variance was significant ($p = 0.038$) in defining the magnitude of overlaps for the 10 ms pre-stimulus interval. The results indicated the shortest silent interval that could be used without compromising the response is between 35 and 45 ms.

3:15

4pPP6. Psychoacoustics of chalkboard squeaking. Christoph Reuter (Musicological Inst., Univ. of Vienna, Vienna, Austria) and Michael Oehler (Univ. of Cologne, Cologne, Germany)

At least since 1975 the "pleasantness" of a sound is discussed from many different angles (Ely 1975; Aures 1984; Halpern *et al*. 1986; Vaschillo 2003; Neumann & Waters 2006; Cox 2008), but often chalkboard squeaking or scratching a chalkboard with finger nails tops the list of unpleasant sounds. The aim of the presented study is to detect specific parts of the sounds that make chalkboard squeaking particularly unpleasant. With a combination of perception experiments and electro-physiological measurements, it was analyzed to what extent the knowledge about the sounds influenced the subjects' judgments and/or the physiological reactions. Basically the study is a replication of Halpern *et al*. (1986), whose methods were extended by several sophisticated sound analysis and re-synthesis techniques and the measurement of some electro-physiological parameters (heart rate and skin resistance) during listening. First results show that especially the modification of the tonal parts as well as applying a filter between 2000 and 4000 Hz led to a more pleasant sound perception. Almost all stimuli were rated more unpleasant if the subjects knew about the nature of the sounds.

3:30–3:45 Break

3:45

4pPP7. The shopping sound experience. Ralf Jung, Gerrit Kahl, and Lbomira Spassova (DFKI, Campus D 3 4, 66123 Saarbrücken 66123 Germany, Ralf.Jung@dfki.de)

The attempt to influence the shopping behavior of customers in a supermarket through music often fails due to the different music preferences. In this work, a method to bring personalized music in the supermarket is presented by providing a location-aware playback and notification service. By using a web interface, customers are able to create an electronic shopping list, associate list items with music tracks, and thus create an individual shopping playlist. These product-associated music tracks start playing at the customer's instrumented shopping cart when he enters specific product departments where an item from the shopping list is located. Additionally, a service to provide product-awareness through non-speech audio cues is presented. This location-aware service notifies customers when they come closer to products that are listed on their shopping list. A two-stage notification approach is used to create user-centric notification zones. Depending on the customer's distance to the product, he gets notified by an ambient or arousal noise that is mixed into the background music. In addition, further information about the detected product is displayed on the screen that is mounted at our instrumented shopping cart.

4:00

4pPP8. Psychological factors influencing the evaluation of electric vehicle interior noise. Jochen Steffens (Inst. of Sound and Vib. Eng., Duesseldorf Univ. of Appl. Sci., Josef-Gockeln-Strasse 9, 40474 Dsseldorf, Germany, jochen.steffens@fh-duesseldorf.de), Thomas Kueppers (Daimler AG, 70372 Stuttgart, Germany), and Sabrina Skoda (Duesseldorf Univ. of Appl. Sci., 40474 Dsseldorf, Germany)

Increasing global awareness of the benefits of electromobility has brought about the need for new concepts in terms of the acoustic design of future vehicle generations. This includes both the creative design process and the development of suitable methods for the subjective evaluation of target sounds. The main difference between e-car sound surveys and those carried out on familiar sound categories is the potential consumer's lack of experience with electric vehicles. Thus, the consumer has no, or very unspecific, expectations in this regard. Several studies have shown that many subjects have to construct their personal frame of reference for evaluation within the listening experiment. However, this is possibly at odds with experience-based expectations relating to sounds of conventional combustion engines. The result is a conflict of objectives between the traditional and the modern, familiarity and strangeness, and not least between driving freedom and ecological awareness. In this context, the authenticity of the sound and the subjective interpretability of the sound information also appear as moderator variables. Moreover, associations with other vehicle categories, for example, streetcars, also influence the perceived sound quality. Within this contribution, these factors will be expounded and their influence on the evaluation of interior noise discussed.

4:15

4pPP9. Auditory-proprioceptive interaction—How do acceleration forces influence the evaluation of driving sounds? Sabrina Skoda, Jochen Steffens (Inst. of Sound and Vib. Eng., Duesseldorf Univ. of Appl. Sci., Josef-Gockeln-Strasse 9, 40474 Duesseldorf, Germany, sabrina.skoda@fh-duesseldorf.de), Thomas Kueppers (Daimler AG, Mercedesstrasse 122, 70372 Stuttgart, Germany), and Joerg Becker-Schweitzer (Inst. of Sound and Vib. Eng., Duesseldorf Univ. of Appl. Sci., Josef-Gockeln-Strasse 9, 40474 Duesseldorf, Germany)

One fundamental requirement for a widespread use of electric powered vehicles is a high degree of social acceptance of alternative drive concepts. As part of this, perception and evaluation of comfort and quality in a vehicle become increasingly important. The customers' judgment on these factors is strongly influenced by the noise and vibration behavior of the vehicle and always is passed in the context of multiple sensory impressions which are processed in the human brain consciously and unconsciously. The interaction mechanisms of sensory perception are highly complex and raise several scientific questions. During the past years, valuable insights about the interaction of visual and auditory perception have been obtained and there are also a number of theories about the auditory-tactile interaction. In contrast, the connection between auditory and proprioceptive perception remains largely unexplored. This paper deals with the question how acceleration forces influence human auditory perception. In order to make evident possible crossmodal effects, a listening experiment was conducted in different kinds of driving simulators. The results of this study will be presented and discussed.

4:30

4pPP10. Structurized sound design process of electric vehicle interior sound. Thomas Kueppers (Daimler AG, NVH Electric Powertrain, Stuttgart, D-70372 Germany, thomas.kueppers@daimler.com), Jan-Welm Biermann (IKA, RWTH Aachen Univ.), and Jochen Steffens (SAVE Inst., FH Duesseldorf Univ.)

The sound character of electric vehicles extremely differs from internal combustion engine vehicles. While the electric vehicle development is still in the beginning, there is potential to consider customer expectations very early in the development process, to have an impact on the typical desired electric vehicle sound on market launch and to generate unique selling propositions in competition between vehicle manufacturers. The current electric powertrain noise is perceived as inconvenient and nonaesthetical especially on long journeys. A complete reduction of this powertrain noise emphasizes the perception of wind and rolling noise and avoids feedback of velocity and load dependency, which decreases emotional impression. Futuristic sounds by sound design systems have to be evaluated by customers and revised in their acceptance and authenticity. Besides the improvement of the sound character, sound design systems can additionally be used to influence the informative acoustic feedback of driving and vehicle parameters. This article introduces the upcoming scenarios for interior sound development and focuses the contradiction between a great amount of freedom in design and a structurized workflow for designing electrical powertrain sounds.

THURSDAY AFTERNOON, 3 NOVEMBER 2011 PACIFIC SALON 4/5, 1:30 TO 3:20 P.M.

Session 4pSCa

Speech Communication: Forensic Acoustics—On the Leading Edge of the Tidal Wave of Change About to Hit Forensic Science in the US? II

Geoffrey Stewart Morrison, Chair
School of Electrical Engineering, Univ. of New South Wales, Sydney, NSW, 2052, Australia

Chair's Introduction—1:30

Contributed Papers

1:35

4pSCa1. Selection of speech/voice vectors in forensic voice identification. James Harnsberger (Dept. Linguist., U. Florida, Gainesville, FL 32607) and Harry Hollien (Instit. Adv. Study Comm. Proc., U. Florida, Gainesville, FL 32611)

The case for the use of speech/voice vectors in speaker identification was made by Hollien and Harnsberger (2010). Those vectors found most robust in capturing speaker-specific characteristics were voice quality, vowel quality, speaking fundamental frequency, and temporal features. In this study, the speech cues for each of the four vectors will be compared to each other with respect to their predictive power. In addition, different vector algorithms and/or processing approaches for each will be contrasted in terms of their effects on identification robustness. One example is conversion of the vowel formant frequencies and bandwidth measurements to geometric scaling (semits). Finally, a second dataset of 18 male voices obtained from evidence recordings, paired with exemplars recorded from a speaker pool, were used to test a modified form of this speech/voice vector approach. The data from these subjects will be compared to those from the 1993 study (substantial improvement) and those from the 2010 experiment (confirmation).

1:50

4pSCa2. When to punt on speaker comparison? Reva Schwartz (U.S. Secret Service, Forensic Services Div. 950 H St. NW, Ste. 4200, Washington, DC 20223), Joseph P. Campbell, and Wade Shen (MIT Lincoln Lab., Lexington, MA, 02420)

In forensic speaker comparison, it is crucial to decide when completion of the examination may not possible (punt). We explore the factors that make speaker comparison decisions difficult or impossible. These factors may include duration, noise, speaking style, language/dialect, mental state, number of speakers, type and quality of recording, and deception. The analyst needs criteria to decide to reject case work. We present analysis of some of these factors and their impact on automatic speaker recognition systems. We propose a methodology for setting objective thresholds by which comparison examples can be rejected. This methodology could be used by forensic analysts to decide whether or not to proceed with speaker comparisons involving these factors.

2:05

4pSCa3. Defining the default defense hypothesis in likelihood-ratio forensic voice comparison. Felipe Ochoa and Geoffrey Stewart Morrison (Forensic Voice Comparison Lab., School of Elec. Eng. & Telecom., Univ. of New South Wales, Sydney, New South Wales, Australia)

In forensic DNA comparison, the person submitting samples for evaluation does not know what properties the samples will have when they are analyzed at the laboratory; samples are submitted as a matter of routine. In contrast, in forensic voice comparison the decision to submit samples for evaluation is based on prior screening: Typically a police officer, a lay person with respect to forensic voice comparison, has listened to the questioned-speaker recording and the known-speaker recording and decided that they sound sufficiently similar that they could be the same speaker and merit evaluation by a forensic scientist. If they do not sound sufficiently similar they are not submitted for evaluation. Unless the defense proposes a more restrictive hypothesis, the forensic scientist should therefore adopt the following as the default defense hypothesis and select a background database accordingly: The known speaker is not the same person as the questioned speaker but is one member of a population of speakers whom to a lay person sound sufficiently similar to the voice on the questioned-voice recording that they would submit recordings of these speakers for forensic comparison with the questioned-voice recording. Examples of how this theory might be applied are discussed.

2:20–2:35 Break

2:35

4pSCa4. Human error rates for speaker recognition. Wade Shen, Joseph P. Campbell (MIT Lincoln Lab., 244 Wood St, C-290A, Lexington, MA, 02420), and Reva Schwartz (U.S. Secret Service, Forensic Services Div. Ste. 4200, Washington, DC 20223)

It is commonly assumed that speaker identification by human listeners is an innate skill under certain conditions. As such, human listening tests have served as the benchmark for automatic recognition systems. In recent evaluations comparing human and machine performance on a speaker comparison task, error rates of naïve human listeners far exceed those of machines [special session on Human Assisted Speaker Recognition, IEEE ICASSP, Prague, 2011]. In this presentation, we quantify the performance of naïve listeners in a variety of challenging channel conditions and we compare these results against automatic systems and trained human listeners. The results of these experiments impact the admissibility of both forensic voice analysis and courtroom testimony by human listeners.

2:50

4pSCa5. Investigating the acoustic and phonetic correlates of deceptive speech. C. Kirchhuebel (Dept. of Electronics, Audio Lab., Univ. of York, Heslington, York, UK YO10 5DD, ck531@york.ac.uk)

The following study describes an initial investigation into the acoustic and phonetic correlates of deceptive speech using auditory and acoustic analysis. Due to the lack of extant data suitable for acoustic analysis, a laboratory-based experiment was designed which employed a mock-theft paradigm in conjunction with a "security interview" to elicit truthful and deceptive speech as well as control data from a total of ten male native British English speakers. Using PRAAT, the control, truthful, and deceptive speech samples were analyzed on a range of speech parameters including f0 mean and variability, intensity, vowel formant frequencies, and speaking/articulation rate. Preliminary analysis suggests that truth-tellers and liars cannot be differentiated based on these speech parameters. Not only was there a lack of significant changes for the majority of parameters investigated but also, if change was present it failed to reveal consistencies within and between speakers. The remarkable amount of inter and intra-speaker variability underlines the fact that deceptive behavior is individualized and very multifaceted. As well as providing a basis for future research programs, the present study should encourage researchers and practitioners to evaluate critically what is (im)possible using auditory and machine based analyses with respect to detecting deception from speech.

3:05

4pSCa6. Progress toward a forensic voice data format standard. James L. Wayman (Office of Graduate Studies and Res., San Jose State Univ., San Jose, CA 95192-0025), Joseph P. Campbell, Pedro Torres-Carrasquillo (MIT Lincoln Lab., Lexington, MA 02420-9108), Peter T. Higgins (Higgins and Assoc., Int., 3900-A, #7E, Watson Pl. NW, Washington, DC 20016), Alvin Martin (Information Access Div. Natl. Inst. of Standards and Technol., Gaithersburg, MD 20899-8940), Hirotaka Nakasone (Operational Technol. Div. Federal Bureau of Investigation, Quantico, VA 222135), Craig Greenberg, and Mark Pryzbocki (Information Access Div. Natl. Inst. of Standards and Technol., Gaithersburg, MD 20899-8940)

The de facto international standard for the forensic exchange of data for biometric recognition is ANSI/NIST ITL-1/2, "Data Format for the Interchange of Fingerprint, Facial, and Other Biometric Information." This format is used by law enforcement, intelligence, military, and homeland security organizations thoughout the world to exchange fingerprint, face, scar/mark/tatoo, iris, and palmprint data. To date, however, there is no provision within the standard for the exchange of audio data for the purpose of forensic speaker recognition. During the recent 5-year update process for ANSI/NIST ITL-1/2, a consensus decision was made to advance a voice data format type under the name "Type 11 record." Creating such an exchange format type, however, is far from straight forward—the problem being not the encoding of the autio data, for which many accepted standards exist, but rather in reaching a consensus on the metadata needed to support the varied mission requirements across the stakeholder communities. In this talk, we'll discuss the progress that has been made to date, the questions that remain, and the requirements for additional input from the broader stakeholder communities.

Session 4pSCb

Speech Communication: Prosody, Articulation, and Articulatory Modeling (Poster Session)

Meghan Sumner, Chair
Dept. of Linguistics, Stanford Univ., Stanford, CA 94305-2150

Contributed Papers

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

4pSCb1. Reduction of consonants and vowels in the course of discourse. Michael McAuliffe and Molly Babel (Dept. of Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, mcauliff@interchange.ubc.ca)

There is a clear link between the discourse status of a word and the degree of reduction. For instance, Gregory [Dissertation (2002)] provided evidence that hearer knowledge affected reduction in production for discourse-old items. Lexical information, such as word frequency, also plays a crucial role in the degree of reduction [Fosler-Lussier and Morgan, Speech Commun. (1999)]. These previous studies either looked at short discourses or words isolated from context. Therefore, the current study investigates longer discourses, using the VIC Corpus [Pitt *et al*., Corpus (2007)]. The primary question is to what degree do repeated uses cause further reductions, and if the reductions are syllabically and segmentally uniform across a word. Many studies on reduction rely on the intuition that reductions occur when information load is light, such as when the word was repeated recently or has a high probability of occurrence, so the prediction is that unstressed syllables would show more reduction than stressed syllables, due to their lighter load of information. Likewise, as vowels vary more than consonants across dialects, the informational load of vowel quality may be less than that of consonant quality, so the prediction is that vowels would show greater reductions than consonants.

4pSCb2. Automatic analysis of constriction location in singleton and geminate consonant articulation using real-time magnetic resonance imaging. Christina Hagedorn, Michael Proctor, Louis Goldstein, and Shrikanth Narayanan (Dept. of Eng., USC, Los Angeles, CA 90089, shri@sipi.usc.edu)

Research on geminate consonants has attempted to establish whether the control of their articulation differs from that of corresponding singletons in temporal parameters, spatial parameters, or both. One piece of evidence supporting spatial control in Italian geminates is EPG results revealing that the location of maximal constriction (CL) of coronal geminates along the palate exhibits small differences from the CL of singletons (Payne 2005). However, our recent work investigating Italian using real-time magnetic resonance imaging (MRI) has shown that when measuring CL dynamically, the CL of singletons and geminates are identical. Dynamic CL is defined as the region of the image that exhibits maximum intensity change during constriction formation and release. These results can be reconciled with the EPG findings if we hypothesize that differences in CL at the moment of maximal constriction are due to compression effects (in the longer geminates) involving the tongue tip sliding along the palate during the closure duration. We evaluate this hypothesis by testing whether CL differences between singletons and geminates can also be found in MRI data, when CL is measured statically (region on the palate contacted in the most constricted frame) in the same utterances in which dynamic CL is invariant. [Work supported by NIH.]

4pSCb3. Development of speech motor control: Lip movement variability. Anders Lfqvist (Dept. Logopedics, Phoniatrics and Audiol., Lund Univ., Lund, Sweden), Johan Frid, and Susanne Schötz (Humanities Lab., Lund Univ., Lund, Sweden)

This study examined variability of lip movements across repetitions of the same utterance as a function of age in Swedish speakers. Subjects were 37 typically developed Swedish children and adults (19 females, 18 males, aged 5–31 yr). Lip movements were recorded during 15–20 repetitions of a short Swedish phrase using articulography, with a sampling rate of 200 Hz. After correction for head movements, the kinematic records were expressed in a maxilla-based coordinate system. Movement onset and offset of the utterance were identified using kinematic landmarks. The Euclidean distance between receivers on the upper and lower lips was calculated and subjected to functional data analysis [Ramsay *et al*., J. Acoust. Soc. Am. **99**, 3718–3727 (1996)] to assess both temporal and spatial variabilities. Results show a decrease in both indices as a function of age, with a greater reduction of amplitude variability. [Work supported by grant 349-2007-8695 from the Swedish Research Council.]

4pSCb4. Imaging and quantification of glottal kinematics with ultrasound during speech. Benjamin Parrell (Dept. of Linguist., Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089), Adam Lammert (Univ. of Southern California, Dept. of Comput. Sci. 3740 McClintock Ave., Rm. 400, Los Angeles, CA 90089), Louis Goldstein, Dani Byrd (Dept. of Linguist., Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089), and Shrikanth Narayanan (Dept. of Elec. Eng., Univ. of Southern California, Los Angeles, CA 90089)

Most examinations of glottal abduction and adduction during speech have employed laryngoscopic video or transillumination (Hoole, 2006). While these provide accurate information about timing of glottal movements, they are invasive and cannot provide absolute measurements about glottal width. At the same time, recent medical studies have used ultrasonic imaging to accurately capture glottal movements and laryngeal anatomy [Hu, J. Ultrasound Med. (2010); Jadcherla *et al*., Dysphagia (2006)]. We demonstrate novel methods of using ultrasound to measure both temporal and spatial aspects of glottal movements during speech. While previous work on glottal ultrasound has been limited by the need to manually analyze each acquired frame, we present methods to automatically quantify glottal aperture in ultrasound images. Finally, since glottal ultrasound does not interfere with the acquisition of supra-laryngeal articulatory data, we present the results of preliminary experiments that record laryngeal and supra-laryngeal speech movements simultaneously using ultrasound concurrently with electromagnetic articulometry. This allows the analysis of relative timing between movements in the two systems as well as examination of changes in timing or magnitude due to variables such as prosodic structure or speech rate. [Work supported by NIH.]

4pSCb5. Detailed study of articulatory kinematics of critical articulators and dependent articulators of emotional speech. Jangwon Kim (Univ. of Southern California, 3740 McClintock Ave., EEB 400, Los Angeles, CA 90089, jangwon@usc.edu), Sungbok Lee (Univ. of Southern California Los Angeles, CA 90089, sungbokl@usc.edu), and Shrikanth Narayanan (Univ. of Southern California, Los Angeles, CA 90089)

This study investigates the articulatory kinematics of critical articulators and dependent articulators as a function of emotion. Our hypothesis is that critical articulators and dependent articulators are utilized differently for achieving distinctive emotion goals that overlay linguistic goals. For example, speakers may use variability of dependent articulators distinctively to that of critical articulators in achieving emotion goals. Distinctive articulatory movements for different emotions have been observed (Lee *et al*. 2005). This study uses a database of three speakers (2 female and 1 male) collected with electromagnetic articulography to collect kinematic information. Articulatory trajectories are aligned by dynamic time warping. Linguistically identical syllable-level segments are analyzed based on detailed aspects of articulatory movements (e.g., position, velocity and phase), after sampling with a 20-ms window and 10-ms shifting. The emotion-specific patterns that emerge for critical (i.e., goal-directed) articulators are compared to those for articulators that are not under overt control for a particular phone. [Work supported by NIH and NSF Grants.]

4pSCb6. Statistical estimation of speech kinematics from real-time MRI data. Adam Lammert, Vikram Ramanarayanan (Univ. of Southern California, 3740 McClintock Ave., Rm. 400, Los Angeles, CA 90089), Louis Goldstein, Khalil Iskarous (Univ. of Southern California, Los Angeles, CA 90089), Elliot Saltzman (Boston Univ., Boston, MA 02215), Hosung Nam (Haskins Labs., New Haven, CT 06511), and Shrikanth Narayanan (Univ. of Southern California, Los Angeles, CA 90089)

The human speech production system can be fundamentally characterized by the kinematic relationships between low-level articulator variables and relatively high-level tasks. Such kinematics can be illuminating about many system aspects from degrees of freedom and redundancy to dynamics and even control. Since these relationships are generally complex and infeasible to express in closed form, recent work has focused on statistical methods for estimating the relevant relationships from data (Saltzman, 2006, in Dynamics of Speech Production and Perception; Lammert, 2010, Proc. INTERSPEECH). Such methods have been applied to synthetic speech data in order to evaluate their effectiveness, but they have not yet been demonstrated on real data. Here, we apply these methods to real speech data acquired from real-time magnetic resonance imaging. We extract articulator variables that are consistent with those chosen by various articulatory models, and we relate them to high-level task variables such as constriction locations and degrees and formant frequencies. This is done to facilitate an analysis of articulatory motor control during speech production. [Work supported by NIH.]

4pSCb7. Semi-automatic modeling of tongue surfaces using volumetric structural MRI. Daniel K. Bone, Michael I. Proctor, Yoon Kim, and Shrikanth S. Narayanan (Univ. of Southern California, Signal Analysis and Interpretation Lab., Los Angeles, CA 90089)

Although volumetric magnetic resonance imaging has proven to be a valuable tool in the study of consonant production (Narayanan *et al*., 1995; Kröger *et al*., 2000), its utility is limited by the difficulty and laboriousness of reliably extracting tissue boundaries from imaging data. Current methods typically involve manual segmentation of air-tissue boundaries (e.g., Birkholz, 2006). Conventional automated (Atkins, 1998) and semi-automated (Ashton *et al*., 1995) methods used for the segmentation of brain MRI datasets may not be directly applicable to lingual segmentation because they are designed to work with different anatomical features. We present a method for extracting tongue surfaces from hi-resolution volumetric MRI data with limited user intervention. For each vocal tract volume to be analyzed, a lingual bounding box and search seed was first specified by an expert user, and voxel intensity was normalized across the region of interest. Lingual surfaces were automatically identified using a multi-pass region-growing algorithm operating over coronal planes. Thresholding was performed asymmetrically to allow for differential detection of air, teeth, and palatal boundaries, in opposition to adjacent lingual tissue. Smoothed tongue surfaces were fit to the resulting volumes by incorporating prior knowledge of intrinsic lingual musculature. [Work supported by NIH.]

4pSCb8. Enhancement of laryngeal features under segmental and prosodic conditioning. Indranil Dutta (Dept. of Computational Linguisitics, EFL Univ., Tarnaka, Hyderabad, Andhra Pradesh, 500007 India)

Evidence in support of enhancement features (Keyser and Stevens, 2006) is presented from an acoustic study of laryngeal contrasts in Hindi. Segmental contrasts where defining features and their corresponding acoustic outcomes are attenuated are said to be made acoustically salient by enhancing gestural features (Stevens and Keyser, 2010). In this study, the four-way laryngeal contrasts in Hindi are examined under varying prosodic and segmental contexts. In segmental contexts where a defining distinctive feature (slack vocal folds) is attenuated in its acoustic manifestation, namely, closure duration (CD), an enhancing gesture (spread glottis) (Avery and Idsardi, 2001) is added to increase the saliency of contrasts between voiced aspirated stops (VAS) and voiced stops (VS). The acoustic consequence of this enhancement is f_0 lowering in the following vowel. Similar results are obtained under weak prosodic conditions where both f_0 lowering and increase in spectral tilt result from the enhancing gesture (spread glottis). In addition, in segmental contexts where gestural overlap compromises the feature (slack vocal folds), (spread glottis) by way of f_0 lowering in the acoustic dimension enhances the laryngeal contrast between VAS and VS. These results lend support to the theory of enhancement as proposed by Stevens and Keyser (2010).

4pSCb9. Statistical analysis of constriction task and articulatory posture variables during speech and pausing intervals using real-time magnetic resonance imaging. Vikram Ramanarayanan, Louis Goldstein, Dani Byrd, and Shrikanth Narayanan (Univ. of Southern California, Electrical Eng., Los Angeles 90089)

We have shown previously using real-time magnetic resonance imaging data (Ramanarayanan *et al*., 2011, ISSP Montreal) that it is more likely that articulatory posture variables (such as jaw angle) are controlled to achieve articulatory settings during pausing intervals in read speech than constriction task variables (such as lip aperture, tongue tip constriction degree, etc.). In this study, we extend this work to examine correlations between constriction task variables and articulatory posture variables during both speech and pause intervals. This work serves to deepen our understanding of the differences in postural motor control of the vocal tract observed during speaking and pausing. [Work supported by NIH.]

4pSCb10. The scope of phrasal lengthening: Articulatory and acoustic evidence. Daylen Riggs and Dani Byrd (Dept. of Linguist., USC, 3601 Watt Way, GFS 301, Los Angeles, CA 90089)

The temporal lengthening that occurs at phrase edges is known as phrasal lengthening. The scope of phrasal lengthening refers to the distance both before and after a phrase edge that phrasal lengthening can occur. This paper examines the influence of prosodic prominence on the scope of phrasal lengthening in articulation. Pitch accent was placed immediately adjacent to the phrase edge and at varying distances before and after the phrase edge. Articulatory durations of gestures were measured in the pitch-accented syllables and in the syllables intervening between the phrase edge and the pitch-accented syllables. Acoustic measurements of consonant, syllable, and vowel duration were also examined. These constriction durations were compared to those for gestures in phonologically parallel control sentences that lacked a phrase boundary. The results indicate that phrasal lengthening is most systematic immediately adjacent to the phrase edge. However, pitch accent can attract phrasal lengthening. One subject showed phrasal lengthening in a pitch-accented syllable, three syllables away from the boundary. Finally, when the remote pitch-accented syllable showed phrasal lengthening, gestures intervening between the phrase edge and the pitch-accented syllable also showed phrasal lengthening. These patterns are evaluated in the context of the prosodic gestural model of Byrd & Saltzman (2003).

4pSCb11. An ultrasound study of Canadian French rhotic vowels. Jeff Mielke (Arts 401, 70 Laurier Ave East, Ottawa, ON K1N6N5, Canada, jmielke@uottawa.ca)

Some speakers of Canadian French produce the vowels /ø/ and /œ/ with a rhotic perceptual quality, leading *pneu*, *docteur*, and *brun* to sound like /pnr/, /dOktarR/, and /bRr/. English /r/ can be produced with a variety of tongue shapes (including bunched and retroflex variants; Delattre and

Freeman 1968, etc.), raising the question of whether French rhotic vowels are also produced with categorically different tongue shapes. Mid-sagittal ultrasound video was recorded for three native speakers of Canadian French producing words containing /œ/, /œR/, and /ø/ in a carrier phrase. Acoustic analysis of rhotic- and non-rhotic-sounding vowels reveals that the rhotic perceptual quality is associated with a low third formant, which is an important acoustic cue for English /r/. Two of the subjects produced rhotic vowels with a tip-down bunched tongue shape closely resembling Delattre and Freeman's type 4 and one subject produced them with a retroflex tongue shape. The study provides articulatory data on retroflex vowels, which are typologically rare, and on bunched-retroflex variation in a language other than English. The parallel cases of articulatory variability in French and English raise opportunities for investigating articulatory-acoustic mapping in bilinguals.

4pSCb12. Can you say [ṽ] or [x̃]? Aerodynamics of nasalized fricatives in Scots Gaelic. Daniel Brenner, Andrea Davis, Natasha Warner, Andrew Carnie, Muriel Fisher, Jessamyn Schertz, Michael Hammond, and Diana Archangeli (Dept. Ling., Univ. of Arizona, Box 210028, Tucson, AZ 85721-0028, dbrenner@u.arizona.edu)

Scots Gaelic is said to have nasalized fricatives ([ṽ, x̃, h̃] etc.). Nasalized fricatives may be aerodynamically impossible (except [h̃]), because opening the velum would vent the high oral pressure required for frication (Ohala 1975). However, phonologically nasalized fricatives which are realized some other way phonetically are possible, and [h̃] is possible because the frication occurs behind the velic opening. The current work presents oral and nasal airflow data from 14 native speakers of Scots Gaelic, primarily from the Skye dialect, to investigate the nasalized fricative distinction. Results indicate that the most common solution to the aerodynamic problem is to neutralize the distinction: most phonological nasalized fricatives, from most speakers, are simply not nasalized at all ([v] for [ṽ]. Some tokens show nasalization during the preceding vowel ([ãv] for [aṽ]). Some tokens in which the expected fricative is pronounced as an approximant (common in Scots Gaelic) show nasalization ([w̃] for [ṽ]). Furthermore, [h̃] occurs. Very rarely, there may be slight nasalization overlapping part of an oral fricative. Thus, the data show that conflicting aerodynamic demands are resolved variably: in this dialect, the distinction is usually neutralized; when it is not, several alternatives that avoid the aerodynamic conflict are produced.

4pSCb13. An articulatory and acoustic investigation of Mandarin apical vowels. SANGIM LEE (Dept. of Linguist., New York Univ., 10 Washington Pl. NY 10003, sangim@nyu.edu)

The present study investigates the articulatory and acoustic properties of apical vowels in Mandarin Chinese using ultrasound and acoustic measures. Instead of the high front vowel, dental and retroflex sibilants are known to be followed by apical vowels made with tongue tip/blade, rather than the tongue body. The ultrasound images of one Beijing speaker confirmed that the tongue tip/blade gesture indeed remains unreleased during the production of these vowels. However, it also revealed that the tongue body for apical vowels is significantly retracted with a relative order of the vowel following dentals being more retracted than the vowel following retroflexes. This gave rise to a consistent acoustic consequence; the second formant of the vowel after dentals of six Mandarin speakers was significantly lower than that of the vowel after retroflexes. In addition, the third formant of the vowel after a retroflex was significantly lowered and nearly merged with the second formant, indicating that the vowel is also strongly retroflexed. The correct analysis is thus claimed to include the apical gestures as secondary features added to the main articulation of the tongue body as in the following notation: a dentalized back vowel and a retroflexed mid vowel.

4pSCb14. Diphthong centralization and reduction in constriction degree. Fang-Ying Hsieh, Louis Goldstein, and Khalil Iskarous (Dept. of Linguist., Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, fangyinh@usc.edu)

Diphthongs in Mexian Hakka Chinese have been divided into two categories (Cheung 2007): Endpoints of the falling (in vowel space) diphthongs have formant values equivalent to those of the monophthongal vowels, while the endpoints of the rising diphthongs generally have more centralized formants. This categorization provides a chance to test the hypothesis of Iskarous *et al*. (2010) that in speech production the location of constriction (CL) changes discretely in transitions between successive targets, while the degree of constriction (CD) changes continuously. We therefore hypothesize that the centralization in these diphthongs results from a reduction in CD (possibly due to undershoot) rather than CL, thus making it parallel to common consonant reduction changes in languages, such as spirantization of stops. The current study tests this hypothesis using TADA (TAsk-Dynamic) modeling, in which CD and CL in speech production can be manipulated. Diphthongs [ia], [ua], [ai], and [au] were modeled, and the preliminary results indicate that a change in target CD alone yields formant patterns that are more similar to those reported compared with a change in CL. Results of additional studies will be presented that analytically derive CD and CL in these diphthongs from new data collected from speakers of Hakka.

4pSCb15. When inferring lingual gestures from acoustic data goes wrong: The case of high vowels in Canadian French. Will Dalton (Dept. of Linguist., Univ. of Ottawa, 70 Laurier East, Ottawa, ON K1N6N5, wdalt017@uottawa.ca)

Canadian French (CF) is distinguished from other dialects partly by the presence of lax high vowel allophones in closed syllables (Walker, 1984). Acoustically, tense high vowels are characterized by a lower F1 than their lax counterparts, which could be the result of tongue root advancement, tongue body raising, or both (Ladefoged & Maddieson, 1996). High vowel allophony in CF therefore represents a case in which articulatory gestures cannot reliably be inferred from acoustic data alone. Nevertheless, the literature discussing the phonetic properties of high vowels in CF commonly assumes tongue root position to be the parameter that distinguishes between tense and lax vowels, despite an absence of empirical evidence. The purpose of this experiment is to test this assumption and provide articulatory evidence using ultrasound imaging to examine tongue position during speech production by CF speakers. Results indicate that an advanced tongue root gesture is not used to distinguish between high vowels in CF; no significant difference in tongue root position was found between tense and lax allophones. Rather, tongue body height was found to be the distinguishing feature. These findings contribute to our knowledge of the typology of articulatory gestures used to distinguish between so-called tense and lax vowels.

4pSCb16. Identifying relevant analysis parameters for the classification of vocal fold dynamics. Daniel Voigt (Dept. of Linguist., Max Planck Inst. for Evolutionary Anthropology, Deutscher Platz 6, D-04103 Leipzig, Germany, daniel_voigt@eva.mpg.de) and Ulrich Eysholdt (Univ. Hospital Erlangen, Bohlenplatz 21, D-91054 Erlangen, Germany)

In the previous work, a computer-based analysis framework was proposed, which is capable of objectively and automatically classifying vocal fold vibrations as captured by high-speed videoendoscopy during phonation. The method is based on quantitative feature extraction from Phonovibrograms combined with nonlinear machine learning techniques, allowing for the discrimination of normal and pathological laryngeal movement patterns. The diagnostic reliability and potential of this analysis approach were demonstrated. However, the practically relevant question, whether certain control parameters of the procedure can lead to increased classification accuracy, remained partially unanswered. In this study, the following parameter sets of the analysis framework were investigated in a systematic manner: method of feature extraction, type of feature aggregation and normalization, number of considered oscillation cycles, feature laterality, classification task, and employed machine learning algorithm. For this purpose, more than 150 000 experiments were conducted using a data set of 105 laryngeal high-speed video recordings, comprising various clinical cases with non-organic findings and subjects from a healthy control group. The results of this extensive study show the particular suitability of certain parameter combinations, helping to further improve the practical application of the automated classification framework for vocal fold dynamics.

4pSCb17. A two-dimensional/three-dimensional hybrid structural model of the vocal folds. Douglas Cook, Pradeep George, and Margaret Julias (Div. of Eng., New York Univ. Abu Dhabi, Abu Dhabi, United Arab Emirates)

The spatial dimensionality of the vocal fold vibration is a common challenge in creating parsimonious models of vocal fold vibration. The ideal model is one that provides acceptable accurate with the lowest possible

computational expense. Inclusion of full three-dimensional (3D) flow and structural vibration typically requires massive amounts of computation, whereas reduction of either the flow or the structure to two dimensions eliminates certain aspects of physical reality, thus making the resulting models less accurate. Previous two-dimensional (2D) models of the vocal fold structure have utilized a plane strain formulation, which is shown to be an erroneous modeling approach since it ignores influential stress components. We herein present a 2D/3D hybrid vocal fold model that preserves three-dimensional effects of length and longitudinal shear stresses, while taking advantage of a two-dimensional computational domain. The resulting model exhibits static and dynamic responses comparable to a 3D model and retains the computational advantage of a two-dimensional model.

4pSCb18. Nonlinear viscoelastic properties of human vocal fold tissues under large-amplitude oscillatory shear. Elhum Naseri, Mindy Du, and Roger W. Chan (Otolaryngol.—Head and Neck Surgery, Biomedical Eng., Univ. of Texas Southwestern Medical Ctr., Dallas, TX 75390-9035)

Traditionally, viscoelastic shear properties of human vocal fold tissues have been described by the linear viscoelastic moduli G' and G" at small strain amplitudes. However, once the mechanical behavior becomes nonlinear these moduli are no longer sufficient for viscoelastic characterization. MITlaos, a rheological framework developed for better describing such behavior, was used to characterize the nonlinear viscoelastic properties of the vocal fold lamina propria when subjected to large-amplitude oscillatory shear (LAOS). MITlaos involved Fourier transform processing to convert raw stress–strain signals into a filtered/smoothed total stress signal based on the odd-integer harmonic components, with nonlinear viscoelastic parameters simultaneously derived. Rheometric testing of vocal fold cover specimens was performed with increasing strain amplitudes using a controlled-strain, simple-shear rheometer. Smoothed Lissajous–Bowditch curves generated from MITlaos were plotted in Pipkin space, and the nonlinear analysis was summarized by variations of rheological fingerprints. Pipkin diagrams served as a geometric means for identifying the onset of nonlinear behavior, and any distortions due to noise. Results showed that the human vocal fold cover when subjected to LAOS demonstrated intracycle strain stiffening and intercycle strain softening with increasing strain amplitude, as well as shear thinning with increasing strain-rate amplitude. [Work supported by NIH.]

4pSCb19. Recovery of articulatory information from acoustics using linear mixed effects models. Sean Martin (Dept. of Linguist., New York Univ., 10 Washington Pl. New York, NY 10003)

The current study attempts to establish a lower bound on the degree to which articulatory information can be recovered from the speech signal. The recovery of articulatory information is modeled with comparatively simple linear mixed-effects models using synchronized articulatory and acoustic data from the University of Wisconsin X-Ray Microbeam Speech Production Database (Westbury, 1994). The performance of this model is compared with other speech inversion models. While previous work [Mitra *et al*. (2010) and others] demonstrates that a more sophisticated model can perform speech inversion quite accurately, the results here show that even a strongly constrained linear model performs better than might be expected given that the well-known non-linearity of the acoustic-articulatory mapping. More detailed investigation of the results indicates that while non-linearity in the acoustic-articulatory mapping affects the recovery of articulatory information for all tongue regions, it contributes considerably more to error in recovering tongue-tip position. For the tongue body, the correlation between predicted and observed tongue body position is approximately 0.9 while the correlation for tongue tip position was approximately 0.65. A linear approximation seems to be sufficient to recover tongue body position with average error of approximately 12%, even with limited acoustic information.

4pSCb20. Emotion effects on speech articulation: Local or global? Sungbok Lee, Jangwon Kim, and Dany Byrd (Dept. of EE, Univ. of Southern California, Los Angeles, CA 90089, sungbokl@usc.edu)

Although differences in overall articulatory behaviors in emotional speech production have been well acknowledged in the literature, it is not well known whether such articulatory differences are distributed or localized along the time axis. In this study, we examine the lower lip and tongue tip trajectories of a perceptually evaluated EMA (electromagnetic articulography) emotional speech database of one male and two female subjects. The speech material is a set of sentences repeated at least three times, and the emotion types investigated are neutral, angry, and happy. Timing differences between repetitions as well as across emotions are normalized by the FDA (functional data analysis, Ramsay and Silverman, 1997) time- alignment technique so that important kinematic land-marks (e.g., velocity maxima) are aligned with respect to each other. Therefore, not only differences in timing but also differences in movement range and velocity can be investigated along the time line. The normalized quantities are sampled along the time line every 10 ms, with a 20-ms window, and their differences along the time line are compared as a function of emotion. Final results as well as their relation to pitch patterns and other acoustic measures will be reported. [Work supported by NIH and NSF.]

4pSCb21. Suprasegmental features in Northern Paiute. Joseph D. Brooks (josephdbrooks@umail.ucsb.edu)

The relationship between intonation and word-level stress in Northern Paiute (Western Numic; Uto-Aztecan) is described in order to support the level of phonology as a basis for the lexical, syntactic, and ideational integration of connected speech into intonation units (IUs). Word-level stress in Northern Paiute, intrinsic and in most cases peninitially fixed, is manifested by vowel lengthening and a rise in pitch. Intonation depends largely on the word-level pitch rises, minimalizing the frequency range for non-tonic accents and increasing it for most rising tonic accents of most IUs. The opposite pattern also occurs, whereby non-stressed syllables of individual words have an intonation pattern superimposed upon them; most tonic accents with falling pitch and a few with rising pitch over a non-stressed syllable distinguish this second category as independent from word-level stress. Although these two patterns can explain how suprasegmental features integrate words into larger, cohesive units, the deictic word class is an exception. Deictics in Northern Paiute lack intrinsic pitch, have extra-long vowels, and occur overwhelmingly in IU-final position. It is proposed that suprasegmental features serve as a cohesive force between the areas of linguistic structure by varying degrees of phonological integration.

4pSCb22. The prosodic structure of Koasati. Matthew Gordon (Dept. of Linguist., UC Santa Barbara, Santa Barbara, CA 93106, mgordon@linguistics.ucsb.edu), Linda Langley (McNeese State Univ., Lake Charles, LA 70609), and Jack Martin (College of William & Mary, Williamsburg, VA 23187)

This paper presents results of the first systematic acoustic investigation of the prosodic system of Koasati, an endangered Muskogean language of Louisiana. Koasati is prosodically complex, featuring lexically marked pitch accents in nouns and verbs, as well as phonologically predictable pitch accents and boundary tones assigned by the intonational system. While most nouns are characteristically associated with a final F0 rise attributed to a phrase-level tone, some have an additional lexically marked high pitch accent on a non-final syllable. The pitch accent system in verbs is more intricate and consists of different types of aspectually determined accents (termed 'grades' in the Muskogeanist literature) and concomitant segmental changes including lengthening, nasalization, or aspiration. Koasati also has a rich intonation system featuring at least two levels of prosodic constituents, the accentual phrase and the intonational phrase, which are associated with boundary tones and/or pitch accents that often temporally overlap with lexically specified pitch accents. Our results will be compared with those for Creek (Johnson and Martin 2001, Martin and Johnson 2002) and Chickasaw (Gordon 1999, 2003, 2004, 2005, 2007, Gordon *et al*. 2000, Gordon and Munro 2007), Koasati's prosodically best described relatives. [Work supported by NSF.]

4pSCb23. Effects of morpheme boundaries in /n/ palatalization in Korean. Jae-Hyun Sung (Dept. of Linguist., Univ. of Arizona, 1100 E Univ. Blvd, Tucson, AZ 85721, jhsung@u.arizona.edu)

Korean palatalization is known to be one of the rare palatalization phenomena where the morpheme boundary plays a role. For example, /mat+i/ 'the elderly', where a morpheme boundary exists between /t/ and /i/, becomes [madi] by the palatalization process, while a tautomorphemic word /mati/ 'joint' is realized as [madi] (*[madi]), with no palatalization. The present study investigates whether the effect of morpheme boundaries is

also at play in allophonic /n/ palatalization in Korean as well as in /t/ palatalization using ultrasonic analysis, and shows that morpheme boundaries contribute to allophonic variation. This study uses ultrasound imaging and audio recordings of 6 native speakers of Korean to examine the difference in palatalization between two morphologically different words in Korean—tautomorphemic words (e.g., [koni] 'swan') and heteromorphemic words (e.g., [muni] (/mun+i/) 'door + nominative marker'). Comparison of the degree of palatalization before [i] shows there is stronger /n/ palatalization in tautomorphemic /n/ + [i] words than that in heteromorphemic /n/ + [i] words, in contrast to /t/ palatalization. The results from this study are in line with the reported language-universal tendency for greater palatalization in tautomorphemic environments.

4pSCb24. Louder is longer: Amplitude conditioned lengthening in diasporic Siraiki. Aditi Arora (Dept. of Phonet. and Spoken English, EFL Univ., Tarnaka, Hyderabad, Andhra Pradesh, India 500007) and Indranil Dutta (EFL Univ., Tarnaka, Hyderabad, India 500007)

Based on a longitudinal study of three generations of diasporic Siraiki speakers in India, it is shown that first generation (Gen1) implosives develop into lengthened plosives, intervocalically, in the subsequent two generations (Gen2/Gen3). Historically, Prakrit geminates developed into Siraiki and Sindhi implosives. Ohala (1992a) observes that maintaining voicing during intervocalic geminates is facilitated by increased oral cavity volume, achieved by lowered larynx (LL). The consequence of this cavity enlargement is listener misperception of the accompanying cues that lead to the genesis of implosives in the intervocalic position in Sindhi (Ohala, 1993). Closure voicing durations, root mean square (rms) amplitude during closure, and f_0 following release are measured in this study. It is observed that in contexts where the Gen1 speakers produce increased rms amplitude at the release of implosives, the subsequent Gen2/Gen3 produce lengthened plosives. Increased peak-amplitude has been shown to be a weak cue for initial voiceless long consonants in Pattani Malay (Abramson, 1998). Based on the evidence from this study and Pattani, it is argued that the acoustic consequence of the feature LL, mainly increased amplitude, allows a Gen2/Gen3 mapping of the Gen1 implosives onto lengthened plosives. This mapping suggests that in addition to listener misperception, sound changes result from articulatory-acoustic mismatch.

4pSCb25. Phonetic vowel hiatus in Spanish. Erika Varis (Dept. of Linguist., Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089, varis@usc.edu)

Phonologically, Spanish tolerates sequences of adjacent vowels at a word boundary, but coalescence and deletion have been impressionistically identified at this juncture as phonetic resolution strategies (Alba, 2006). In other languages, phonetic measurements of vowels in hiatus show gradient distinctions between different prosodic environments, supporting a gestural overlap analysis that produces the perception of deletion or coalescence (for example, Igbo, Catalan, and Greek). The current experimental study investigates phonetic manifestations of vowel hiatus in four prosodic environments (IP, PP, Pwd, and clitic) in Spanish to determine whether hiatus resolution is accomplished by vowel deletion or gestural overlap. The study was conducted with nine native speakers of Peninsular Spanish. The duration and formant frequencies of vowel sequences was measured in read sentences that manipulated boundary strength separating the two vowels. Higher-level prosodic boundaries IP and PP produced no significant hiatus effects, but vowels separated by lower-level Pwd and clitic boundaries showed formant influences of one vowel on the other. The durations of the two-vowel sequences were longer than singleton controls across all conditions, indicating no vowel deletion. The results support an analysis of vowel gesture overlap and reject deletion, contrary to auditory impressions but consistent with the findings in other languages.

4pSCb26. Acoustic cues to prominence in children's speech. Irina A. Shport and Melissa A. Redford (Dept. of Linguist., Univ. of Oregon, Eugene OR 97403-1290)

Previous work indicates that phrasal prominence patterns are not yet adult-like in 6-year-old children's speech. Perception of prominence in English is influenced by temporal patterning, amplitude changes, and fundamental frequency variations across the phrase. This study examined the relative weightings of these cues to prominence in adult judgments of 6-year-old children and college-aged adults' speech. Eleven adult judges listened to two—word phrases produced by 25 children and 25 adults in a counting task designed to induce stress shift (*thirtéen banana* versus *thirtéen barbeque*). The judges decided which word was most prominent in the phrase: number, noun, or equal prominence. Although the number word was always judged to be prominent regardless of speakers' ages or context, initial results indicate that agreement between judges varied systematically with different cues in children and adults' speech. More judges agreed that the number word was prominent in children's speech when the first syllable of this word was produced with an especially high F0 and long duration, but only the relative intensity of the initial syllable predicted inter-judge agreement for adult speech. The results have implications for understanding the development of cue integration in the prominence production. [Work supported by NIH/R01HD061458.]

4pSCb27. Prosodic pattern detection based on fuzzy set theory. Nicholas Bacuez (Dept. of French and Italian, Univ. of Texas, 1 Univ. Station B7600 Austin, Texas 78712-0224, nicholasbacuez@gmail.com)

This research utilizes fuzzy logic and fuzzy set theory to experimentally abstract the mental representations of prosodic contours otherwise not accessible to speakers or researchers. I have developed a pattern recognition system to extract approximations of these representations and their range of variation from speech datasets. Following the principles of fuzzy set theory, a prosodic contour is defined as a category and the variation range of its components is expressed in terms of degree of membership to the category, from 0 (excluded) to 1 (included). Subsequently, the system analyzes the prosodic contour of new sentences by assessing through fuzzy matching their degree of membership to previously identified categories: 1 is perfect, 0.5 is borderline, and 0.1 is improbable. In the current study, I have used closed questions obtained experimentally from native French speakers. The prosodic contour of closed questions is non problematic and ensures the controllability of the output. The model output is an array storing the ideal curve and its degrees of variation. Fuzzy matching reveals that most closed questions from the dataset are in medium range (0.5). This materializes the idea that vagueness (Russell, Wittgenstein), the linguistic correlate to fuzzy logic, is a natural property of language.

4pSCb28. Pragmatically determined variation in Greek wh-question intonation. Stella Gryllia (Univ. Potsdam, Inst. fuer Linguistik/Allgemeine Sprachwissenschaft, Komplex Golm, Haus 35, 0.02, Karl-Liebknecht-Str. 24-25, D-14476 Golm, Germany, gryllia@uni-potsdam.de), Mary Baltazani (Univ. of Ioannina, Ioannina 45110, Greece), and Amalia Arvaniti (UC San Diego, La Jolla, CA, 92093-0108)

This paper presents production data testing the analysis of Arvaniti and Baltazani (2005) and Arvaniti and Ladd (2005) according to which the default melody used with Greek wh-questions is L*+H L- !H% (showing a delayed accentual peak on the utterance-initial wh-word, a low stretch, and a final curtailed rise), with !H% sometimes being replaced by L%. Here it was hypothesized that the melodies also differ in pitch accent and are used in different contexts. Four speakers, two male and two female, took part in reading a varied corpus of questions in contexts that lead to the use of a wh-question either in order to seek information or in order to politely register disagreement (a function of wh-questions peculiar to Greek). Our results confirmed that there are two different melodies: L*+H L- !H%, with a delayed accentual peak and a final rise, and L+H* L- L%, with an early peak and no final rise. The former is used for requesting information and the latter when questions function as dissenting statements. In addition to leading to a revision of the existing analysis, these results show that distinctions such as statement versus question are too coarse-grained for the analysis of intonational meaning and function.

4pSCb29. Acoustic correlates of narrow focus in Turkish. Canan Ipek (Dept of Linguist., Univ. of Southern California Grace Ford Salvatori 301 Los Angeles, CA 9008, ipek@usc.edu9-1693)

Acoustic correlates of narrow focus in Turkish focus is known to affect the acoustic properties of the lexical element under focus. Its effect on the pre-focus and post-focus domains has attracted much less attention. This study aims to investigate the acoustic changes as a function of focus in the pre-focus and post-focus domains as well as in the on-focus domain in Turkish and listeners' sensitivity to those changes in retrieving information from

the acoustic signal. For this purpose, a production and a perception experiments have been conducted. For the production study, speakers read sentences in which the location of focus was manipulated via wh-questions preceding those sentences. For the perception study, listeners heard selected sentences recorded for the production study and were asked to judge the prominent word in the sentence. Results showed that focused words had increased duration and intensity, and post-focus words had reduced F0, duration and intensity. Interestingly, pre-focus words were found to have increased F0 and duration. Listeners' identification of focus was positively affected by the presence of acoustic changes especially in the post-focus domain. These findings have implications for speech perception and modelling prosody.

4pSCb30. Modeling imperatives in Spanish. Sergio Robles-Puente (Dept. of Linguist., Univ. of Southern California, 3601 Watt Way GFS 301, Los Angeles, CA 90089-1693, roblespu@usc.edu)

The intonation of imperatives in Spanish has traditionally been considered to not differ systematically from that of declaratives. This study shows that given the appropriate contexts, imperatives can exhibit unique phonetic properties. Nine speakers of Peninsular Spanish produced imperatives in response to instructions that elicited different levels of imperativity, along with control declarative items. Results show that while imperatives may fail to differ from declaratives in some conditions, when the context requires a stronger imperative, speakers use intonational configurations not found in declaratives. These include higher F0 values and changes in the overall pitch contour with higher F0 values toward the end of the sentence, different boundary tones and different F0 peak alignments. A perceptual experiment with 13 speakers confirmed the relevance of these intonational modulations by demonstrating that the strategies that were more commonly used in the production experiment were preferred over others to express imperativity. Results can be modeled within the framework of grammar dynamics (Gafos and Benus, 2006. Cognitive Science, **30**, 837–862). [This research was supported by the University of Southern California Del Amo Foundation.]

4pSCb31. Intonation of Mandarin speakers in their English as a Foreign language. Karen Barto-Sisamout (SLAT Program, 1423 E. Univ. Blvd., P.O. Box 210067, Tucson, AZ 85721, kabarto@email.arizona.edu)

Does the prosody of speakers' first language (L1) influence their prosody in their second language (L2)? The current work investigates this for tone languages (Beijing Mandarin and Taiwanese Mandarin) as L1, and an intonation/stress language, English, as L2. English uses F0 contours in the intonation system, to signal syllabic prominence in a word, word prominence in a phrase, and the difference between questions and statements. In English, there is a pitch peak delay, where the F0 peak occurs after the stressed syllable in two-syllable stress-initial words. Conversely, tone languages use F0 contours lexically, and in the case of Mandarin, the F0 peak is on the stressed syllable. Thus, F0 measurements were taken from three subject groups (Beijing Mandarin L1, Taiwanese Mandarin L1 and Native English) who produced narrow and broad focus statements and questions, to learn if the Mandarin speakers lack pitch peak delay in English like in their L1, or delay some of their pitch peaks beyond the corresponding syllable offset, like English speakers. Further, all groups produced contrastive focus statements and questions, to see to what degree differences in the L1 system impact L2 production, as Beijing and Taiwanese Mandarin differ in these structures. Data analysis is ongoing.

4pSCb32. Prosodic characterization of reading styles using audio book corpora. Michael I Proctor and Athanasios Katsamanis (Signal Anal. and Interpretation Lab., Univ. of Southern California, Los Angeles, CA 90089)

Although native English speakers have strong intuitions about the felicity of different reading styles, it is unclear which properties of read speech contribute to these reactions. Although the prosodic structures of read speech and spontaneous speech have been shown to differ (Howell and Kadi-Hanifi, 1991; Blaauw, 1994), it is not clear whether similar prosodic factors contribute to the perception of different reading styles as more felicitous nor even whether such differences can be systematically quantified. A large-scale corpus analysis of read speech was conducted to shed more light on the prosodic characteristics of those reading styles preferred and dispreferred by native speakers of American English. Audio book recordings of classic works of English literature by male and female American readers were rated by native speakers. The two most and least preferred renditions were transcribed at lexical and phonemic levels using SailAlign (Katsamanis, 2011). A variety of metrics were calculated to characterize prosodic properties of each of the readers, including %V, VarCoV, VarCoC, and nPVI (Grabe, & Low 2002; Stojanovic, 2009). The results suggest that although listeners exhibit a preference for syllabic regularity, the perceived felicity of reading styles results from a combination of factors. [Work supported by NIH].

4pSCb33. Acoustic correlates of spanish speech rhythms. Michael J. Harris (Dept. of Spanish and Portuguese, UCSB, Phelps Hall 4206, Santa Barbara, CA 93106-4150, michaelharris@umail.ucsb.edu) and Stefan Th. Gries (UCSB, CA 93106-3100, stgries@linguistics.ucsb.edu)

This paper describes a study of the acoustic correlates of speech rhythms of Hispanic Bilinguals living in California and Mexican Monolinguals living in Mexico City in order to study the effect of bilingualism on language, especially on rhythm classes, and the reliability of acoustic correlates in distinguishing these classes. This study addresses stress-timing versus syllable-timing as described in Pike's pioneering work (1945). Ten monolingual speakers from Mexico and ten bilingual Spanish–English speakers born and raised in California to Mexican parents were recorded speaking spontaneously. Fifty vowel durations per speaker were collected from phrases in these recordings and explored statistically and graphically with R.2.12.2 (R Development Core Team, 2011) in order to determine the reliability of various acoustical correlates of language rhythms in differentiating speech rhythms between varieties of Spanish. Specifically the Pairwise Variability Index, introduced by Low and Grabe (1995), and interval measures, such as the standard deviation and normalized standard deviation of vowel durations, were explored. The effects of word frequencies, as determined by relevant files from the Corpus del Espa'ol (Davies, 2002), were also considered in the analysis of the data. [Work supported by a grant for the University of California Institute for Mexico and the United States (UC Mexus).]

4pSCb34. Effect of linguistic background on convergence of prosodic rhythm. Gayatri Rao (Dept. of Psych., University of Texas, 1 University Station A8000, Austin, TX 78712, raog@mail.utexas.edu), Rajka Smiljanic (Univ. of Texas, 1 University Station B5100, Austin, TX 78712), and Randy Diehl (Univ. of Texas, 1 University Station A8000, Austin, TX 78712)

Speech patterns of the interlocutors become more similar to each other over the course of an interaction. These spontaneous speech adaptations, or phonetic convergence (PC), have been demonstrated for segmental features, such as vowels and voice onset times (VOT) and for suprasegmental features, such as stress. In this study, speaker adaptations to speech rhythm are examined before and after an interactive map task. Using American English and Indian English speakers, convergence was measured using the centroid of the envelope modulation spectrum (EMS + centroid, Rao & Smiljanic, 2011). This spectral measure of rhythm goes beyond considering consonantal and vocalic duration variability, as used in the traditional rhythm measures, and includes information about syllable prominence, stressed and unstressed syllable variation and distribution, and pauses and disfluencies. This research will allow us to examine whether language background has an effect on convergence of global speech properties, such as linguistic rhythm. The results of this study add to our current knowledge of features that are subject to imitation in the speech of dialogue partners.

4pSCb35. A biologically inspired neural network for modeling phrase-final lengthening. Erin C. Rusaw (Dept. of Linguist., Univ. of Illinois, 4080 Foreign Lang. Bldg., 707 S Mathews Ave. Urbana, IL 61801, erusaw2@illinois.edu)

This work proposes a central-pattern-generator-inspired neural network model for the interaction between phrase-final lengthening and stress. Recent work in the area of speech prosody has been concerned with the mechanisms involved in phrase-final lengthening, and specifically how phrase-final lengthening interacts with stress or prosodic prominence. The current study investigates the interaction of stress and lengthening at the end of English phrases. Adult American English speakers were recorded reading aloud sentences in which phrase boundaries had been manipulated so that the target words were either phrase-final or phrase-medial, and the durations of syllables in the target words were compared between the two conditions.

Results so far support previous findings that phrase-final lengthening in English affects stressed syllables near phrase boundaries and phrase-final syllables while leaving unstressed syllables between the two unaffected [Turk and Shattuck-Hufnagel, 2007]. Domain-based models of prosodic lengthening have so far been unable to provide a unified account of this phenomenon. A biologically plausible artificial neural network is shown which provides a model of the mechanism behind this interaction using three oscillators with differing periods which input to three interconnected thresholded integrate-and-fire artificial neurons, the output of which determines the timing of the syllables, stress feet, and phrase.

THURSDAY AFTERNOON, 3 NOVEMBER 2011 PACIFIC SALON 3, 1:00 TO 5:00 P.M.

Session 4pUWa

Underwater Acoustics: Volume Scattering From Objects, Bubbles, or Internal Waves

R. Lee Culver, Chair
Applied Research Lab., Pennsylvania State Univ., P.O. Box 30, State College, PA 16804-0030

Contributed Papers

1:00

4pUWa1. Boundary enhanced scattering by solid metallic simple geometric shapes: Experiments and modeling. Jon R. La Follett (Naval Surface Warfare Ctr., Panama City Div., Panama City, FL 32407, jon.lafollett@navy.mil) and Philip L. Marston (Washington State Univ., Pullman, WA 99164-2814)

The presence of a flat reflecting boundary can enable target scattering mechanisms that are not possible for objects in the free field. Experimental results demonstrating a strong boundary related backscattering feature for a solid aluminum cylinder near an air–water interface will be presented. This effect has been modeled previously [J. R. La Follett, Ph.D. thesis, WSU (2010)] by treating the volume of water bounded by the top surface of a solid cylinder and the air–water interface as a waveguide; qualitative agreement between the model and experimental results was demonstrated. In the present work that model is extended to incorporate aspects of the scatterer geometry. Monostatic and bistatic experimental measurements were obtained by suspending a solid aluminum cylinder, solid steel sphere, and a rectangular aluminum bar through the air–water interface of a tank. Model predictions for a solid cylinder are in good agreement with the observed dependence of the feature on the distance from the cylinder to the air–water interface. [Research supported by the NSWC PCD In-house Laboratory Independent Research program.]

1:15

4pUWa2. Bistatic scattering by scaled solid metallic objects: Circular line-scan measurements and modeling. Jon R. La Follett, Patrick C. Malvoso, and Raymond Lim (Naval Surface Warfare Ctr., Panama City Div., Panama City, FL 32407, jon.lafollett@navy.mil)

Multistatic sonar systems can be used to obtain target scattering information that cannot be measured monostatically. This information has potential benefits for detection and classification schemes. Bistatic and monostatic scattering measurements have been performed simultaneously on several scaled targets using the circular line-scan system of the small scale test bed facility at NSWC PCD. Water tank measurements were made for targets in the free-field and resting proud on or buried in simulated scaled sediment composed of spherical glass beads. Targets were placed in the center of the circular scan line. Circular synthetic aperture sonar and frequency-aspect target strength (acoustic color) results will be presented for a solid steel sphere, solid aluminum cone, and solid aluminum cylinders. Results for cylinders and the sphere are compared with T-matrix simulations to facilitate interpretation of features observed. Comparisons between back and forward scattering results demonstrate particularly strong features for each target in the forward scattering direction. Forward scattering by the sphere and cylinders also exhibits responses that arrive earlier in time than sound that travels directly from the source to the receiver. In the case of the sphere, this is attributed to elastic target responses involving leaky Rayleigh waves. [Research supported by ONR.]

1:30

4pUWa3. Scattering from highly extended acoustical objects using multiple precision computation. C. Feuillade (Facultad de Física, Pontifícia Universidad Católica de Chile, Av. Vicuña Mackenna 4860, Santiago, Chile)

The extended boundary condition (or "T-matrix") technique of Waterman [J. Acoust. Soc. Am., **45**, 1417-1429 (1969)] has recently been used to study scattering from highly non-spherical axisymmetric air-filled acoustical objects in water. Evaluation of the T-matrix for extended objects is frequently problematical, typically requiring the inversion of very large and highly ill-conditioned matrices. A second issue is computation of the matrix elements themselves, requiring accurate numerical quadrature of the surface integrals around the scattering object. In this work, a computational scheme was implemented using MATLAB®, and a freely downloadable software package "mptoolbox," which enables numerical computations to be performed to arbitrary degrees of precision. Here, it was implemented with 250 bits of precision specified for the mantissa part of the number, leading to computations with about 75 decimal places of accuracy. Using a multiple precision method significantly improved the stability and accuracy of both types of operations and indicates a powerful tool for the successful implementation of many scattering and other acoustical problems demanding high precision computational schemes. [Work supported by ONR.]

1:45

4pUWa4. On Rayleigh and Mie scattering. Jerald W. Caruthers (Dept. of Marine Sci., Univ. of Southern Mississippi, 1020 Balch Blvd., Stennis Space Ctr., MS 39560, jerald.caruthers@usm.edu)

Scattering described today as "Rayleigh scattering" represents something that is far short of what Rayleigh actually contributed to the topic in both optics and acoustics. This limited view seems to lie in a few papers in which he truncates series solutions for practical computations, thus leading to scattering of the form $(ka)^4$ for $ka \ll 1$, where k is the wavenumber and a is the radius of the sphere and for selected limitations on index of refraction. These approximations led optical scientists to equating "Rayleigh scattering" to little more than "the blue sky." In 1908, Gustav Mie developed a theory for plane-wave scattering from a sphere to which the names "Mie theory" and "Mie scattering" have been indelibly attached to many applications in optics. It is virtually unknown, especially in optics, that Rayleigh actually developed the full theory of plane-wave scattering from a sphere in 1878 (primarily Section 334, Vol. **2**, *The Theory of Sound*, Macmillan),

including original contributions in the concurrently developing mathematics of Bessel functions. The motivation of this presentation is to establish a means of treating weak scattering from bubbles based on their contribution as a distribution of spheres by combining Rayleigh and Mie.

2:00

4pUWa5. Sound speed, pulse spreading, and reverberation in muddy bubbly sediments. William M. Carey and Allan D. Pierce (Dept. Mech. Eng., Boston Univ., Boston, MA 02215)

The sound speed characteristic of the high-porosity mud has been found to have sonic speeds lower than expected. Since the presence of bubbles is known to be an important factor in decreasing the sound speed, these low sound speeds are attributed to methane microbubbles that result from biological decay. A theoretical treatment of "muddy sediments," the Card House Theory (Pierce and Carey, POMA (5), 7001, 2009), estimated the slow sound speed and frequency dispersion proportional to mud porosity, $C_{mud} \sim (0.91 - 0.97)C_w$. The presence of microbubbles can lower the sound speed consistent with the Mallock-Wood equation [Carey and Pierce, A 1aUW6, J. Acoust. Soc. Am. 129(4,Pt2) April 2011]. The recent Dodge Pond experiment found low sound speed estimates consistent with bubble volume fractions between 10^{-4} to 10^{-5}. The experiment has also produced estimates of pulse time spreading and reverberation. This paper interprets these results in terms of a three-component mixture with the bubbles distributed in a random Poisson process. Since measurement of the bubble size distribution within the mud is difficult, limits on the distribution may be obtained by the frequency dependent nature of the sound speed, pulse spreading, and reverberation characteristics. [Work sponsored by ONR OA and NSWC PCD.]

2:15

4pUWa6. Acoustic scattering by bubbles in naturally occurring mud sediments. Allan D. Pierce and Willam M. Carey (Dept. Mech. Eng., Boston Univ., Boston, MA 02215, adp@bu.edu)

Naturally occurring sediment mud contains bubbles created by decaying vegetable matter. Work reported by Preston Wilson *et al*. (ca. 2007) has determined via x-ray tomography systems that mud bubbles are not spherical in shape, but resemble oblate spheroids and are "inhomogeneously distributed." These features are explained in terms of the card-house structure of mud with an adaptation of the fracture mechanics ideas of Boudreau *et al*. (ca. 2002). The scattering of sound at low frequencies by such non-spherical bubbles has both monopole and dipole components. The scattered wave associated with the monopole term is proportional to the bubble volume. The dipole term involves an effective entrained mass tensor, which is found by a solution of Laplace's equation. All bubbles, regardless of shape, have a smallest resonance frequency, and the scattered radiation near the resonance frequency is monopole in character. Example solutions for the resonance frequencies and the scattering near resonance are given for oblate spheroidal bubbles, and a suggested interpolation from low frequencies to resonance frequencies is given. A discussion is also given of how one can make use of the range-evolving form of compact-source generated pulses to infer information about the bubbles near the propagation path.

2:30

4pUWa7. Affects of nearby bubbles on underwater array gain. R. Lee Culver and J. Daniel Park (Appl. Res. Lab, Penn State Univ., P.O. Box 30, State College, PA 16804)

Combining multiple sensor signals coherently (i.e., beamforming) improves spatial or angular resolution and increases signal to noise ratio (SNR). When the array is steered, signals arriving from the steering direction add in phase, while signals arriving from other directions do not (proper choice of signal frequency assumed). Array gain (AG) is a measure of how much the SNR at the array output is increased relative to array input SNR. The degradation in underwater acoustic array AG by scattering from nearby bubbles was measured at the AB Wood tank located at the Institute of Sound and Vibration Research (ISVR), University of Southampton, in June 2008. AG degradation is separate from the effects of bubbles in water to attenuate acoustic signals. Measured statistics of signal phase at the individual sensors show that as bubble density increases, phase differences between the elements increase and AG is degraded. We present a theory and numerical simulation that attributes the phase shifts to scattering from nearby bubbles and provides a way to predict AG degradation from the bubble density. Work sponsored by ONR Undersea Signal Processing.

2:45

4pUWa8. Transport equations for cross-frequency transport equations for acoustic intensity moments. Dennis B. Creamer (P.O. Box 660537, Arcadia, CA 91066)

The second and fourth moment mode-amplitude statistics for ocean sound propagation through random sound-speed perturbations are investigated using exact transport theory for the cross-frequency cross-mode coherence matrix. These exact equations are derived using the method of successive approximations, originally developed by Klyatskin and Tartarskii. These equations allow the determination of the validity of the usual transport equation (involving the Markov approximation), which is the first order approximation (in a infinite sequence of approximations to the exact equations). The range scales for the approach to the asymptotic behavior of the intensity moments, and the decay of the cross-modal coherence is easily determined at all frequencies.

3:00

4pUWa9. Transport equation approach for second order mode statistics in an ocean with random sound speed perturbations: Coherence. John Colosi and Tarun Chandrayadula (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943)

A transport equation has been derived to describe the range evolution of the single frequency cross mode coherence matrix so that acoustic field coherence functions with temporal as well as depth and transverse separations can be easily computed. The theory assumes 2-D propagation in the depth range plane, small angle weak multiple forward scattering, and the Markov approximation, and it has been previously shown to accurately predict the observable of mean intensity for both deep and shallow water environments. This talk will address the issues of: the accuracy of the approximations, relative contributions from coupling and adiabatic effects, scaling with range and frequency, and the functional form of the coherence with regards to lag.

3:15–3:30 Break

3:30

4pUWa10. Investigating sources of variability of the range and structure of the low frequency shallow convergence zone. Stephen D. Lynch, Gerald L. D'Spain (Marine Physical Lab.—SIO, 291 Rosecrans, San Diego, CA 92106), Kevin D. Heaney (OASIS, Lexington, MA 02421), Arthur B. Baggeroer (MIT, Cambridge, MA 02139), Peter Worcester (SIO, La Jolla, CA, 92093), James Mercer (APL-UW, Seattle, WA, 98105), and James Murray (OASIS, Lexington, MA 02421)

During an experiment in the northern Philippine Sea in 2009, a ship towing Penn State's Five-Octave Research Array (FORA) at approximately 120 m depth drove counter-clockwise in an arc, maintaining constant range at one convergence zone (CZ) from a second ship holding station with an acoustic source deployed at 15 and 60 m. In addition, the FORA was towed at various depths in a star pattern about the station-keeping source ship, thereby sampling the first CZ in range, depth, and azimuth. Throughout the experiment, sound speed profiles were measured using expendable bathythermographs, expendable sound velocimeters, and conductivity/temperature versus depth sensors, and detailed bathymetric data were collected using the multibeam systems aboard these and other ships. By incorporating this extensive environmental information into numerical models, variability observed in these measurements of the range and structure and asymmetry of the distribution of received levels of the first CZ resulting from a shallow source to shallow receiver are attributed to variability in the sound speed of the upper-most water column and bathymetry (especially bathymetric features mid-way between the source and receiver). It is found that the sensitivity of the CZ to bathymetry is dependent on the sounds speed in the upper water column.

3:45

4pUWa11. Spectral effects of nonlinear internal waves on narrowband shallow-water signal propagation. Chad M. Smith and David L. Bradley (Appl. Res. Lab., Penn State Univ., P. O. Box 30, State College, PA 16804)

Water column, bathymetry, and acoustical sediment properties collected during the transverse acoustic variability eXperiment (TAVEX) of 2008 are used to create a 30 km computational environment for propagation models. These models are used to display internal-wave-induced acoustic variability which is characteristic of the shallow-water northern East China Sea environment where the experimental work took place. Analyses of computational models are then compared to narrowband acoustic recordings made during the experiment using a spectral and time-of-arrival analysis approach. The expected acoustic effects and computational impact of specific internal wave characteristics on acoustic arrivals will be discussed along with recorded data comparison. [Work supported by the Office of Naval Research.]

4:00

4pUWa12. Variability of horizontal interference structure of the sound field in the presence of moving nonlinear internal waves. Mohsen Badiey (Univ. of Delaware, College of Earth, Ocean, and Environment, Newark, DE 19716, badiey@udel.edu), Boris Katsnelson, and Andrey Malikhin (Voronezh State Univ., Voronezh 394006, Russia)

Variations of the low-frequency sound field when a train of nonlinear internal wave (NIW) crosses the acoustic track are considered. Three positions of moving NIW train, during 10:30 to 11:37 GMT on 19 August 2006 are examined. During this time, three periods of sound transmission took place, each lasting about 7.5 min. These periods referred to as starting time (10:30–10:37.5 GMT), mid-time (11:00–11:07.5 GMT), and the end-time (11:30–11:37.5 GMT). Low frequency modulated signals centered at 300 Hz, and bandwidth of 60 Hz with duration of about 2 s were transmitted. During this time, NIW consisting of 6-7 separate solitons was shifted in horizontal position by 2.5 to 2.8 km, moving toward the coast at the velocity of about 0.75 m/s. Fluctuations of the sound field in horizontal plane are described in detail for three periods of the sound pulses radiation: near the forward front, in the middle of the train, and near the back front using the fluctuations of an angle of horizontal refraction. Estimation of this angle using experimental data and those of the corresponding theory show the same value. [Work supported by ONR.]

4:15

4pUWa13. Observations of Philippine Sea sound-speed perturbations, and the contributions from internal waves and tides, and spicy thermohaline structure. John Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943), Brian Dushaw (Univ. of Washington, Seattle Washington, 98105), Lora Van Uffelen (Univ. of Hawaii, Honolulu, HI, 96822), Matt Dzieciuch, Bruce Cornuelle, and Peter Worcester (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

In the PhilSea09 pilot study two moorings equipped with temperature (T), conductivity (C), and pressure sensors, along with upper ocean ADCP, monitored ocean variability for a month in the Spring. The measurements reveal an energetic and nonlinear mixed diurnal-semidiurnal internal tide, a diffuse Garrett-Munk (GM) type internal wave field at or above the reference GM energy level, and a strong eddy field. One mooring, which was equipped with pumped sensors for enhanced salinity (S) resolution, was able to accurately quantify T and S variability along isopycnals (spice). The spice contribution to sound-speed fluctuation is observed to be strong near the mixed layer but significantly weaker than the other contributions in the main thermocline. Frequency spectra as well as vertical covariance functions will be presented to quantify the temporal and vertical spatial scales of the observed fluctuations.

4:30

4pUWa14. Refraction of horizontal rays and vertical modes from receding solitary internal wave front in shallow water. Mohsen Badiey (Univ. of Delaware, College of Earth, Ocean, and Environment, Newark, DE 19716, badiey@udel.edu), Valery Grogorev, and Boris Katsnelson (Voronezh State Univ., Voronezh 394006, Russia)

Previously, it was shown that on the approach of an internal wave to an acoustic source-receiver track in shallow water, interference between a direct and a horizontally refracted acoustic path can occur (J. Acoust. Soc. Am. **129**(4), EL141, 2011). This phenomenon that is dependent on the angular geometry between the solitary internal wave (IW) front and the acoustic track is similar to the well known Lloyd mirror interference phenomenon in optics. In particular, for a specific time during the SW06 experiment, it was shown that the refracted pulse propagating along horizontal ray, corresponding to the fourth vertical mode arrives with temporal delay relative to the direct horizontal ray. In this paper, we show this phenomenon occurring on the back front of the solitary IW. On the receding front, the forth mode of the refracted signal occurs with an increasing delay in the modal arrival time as the IW leaves the acoustic track. This separation between the direct and the refracted path continues while it gets larger until the IW is far enough from the track for refraction not to occur. A theoretical description of this phenomenon in support of the experimental observation is also presented. [Work supported by ONR.]

4:45

4pUWa15. Variations in the active and reactive intensity components of the sound field due to nonlinear internal waves. Robert J. Barton (Naval Undersea Warfare Ctr., Div. Newport, 1176 Howell St., Newport, Rhode Island 02841), Georges A. Dossot (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett Bay Campus, Narragansett, RI 02882), and Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA 93943)

Shallow water acoustic energy propagation influenced by nonlinear internal waves is investigated by examining complex acoustic intensity vector fields. The acoustic field is modeled using the three-dimensional Cartesian version of the Monterey-Miami parabolic equation (MMPE) algorithm, which relies upon a split-step Fourier approach. The modeled internal wave is approximated using environmental mooring data from the Shallow Water ’06 (SW06) field experiment, interpolated into a representative three-dimensional sound speed profile, and incorporated into the PE model. The soliton wavecrests are oriented such that they are parallel to the direction of forward acoustic propagation and variations along their length (such as curvature) are neglected. Both pressure and particle velocity fields are computed in a self-consistent manner, allowing a full description of the three-dimensional acoustic intensity field which describes the flow of energy in the presence of the solitons. The complex intensity field is separated into its active and reactive (real and imaginary) and spatial components and presented in the form of energy plots. Specific modeled examples showing horizontal refraction, focusing, and defocusing effects on the structure of the acoustic intensity field are illustrated.

Session 4pUWb

Underwater Acoustics: Measurement, Characterization, and Mitigation of Underwater Anthropogenic Noise

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Contributed Papers

1:00

4pUWb1. Statistical analysis of northern Philippine Sea underwater sounds. Brianne Moskovitz, Gerald D'Spain (Marine Physical Lab., Scripps Inst. of Oceanogr., 291 Rosecrans St., Bldg. 4, San Diego, CA 92106, bmoskovi@ucsd.edu), Peter Worcester, Matt Dzieciuch (Scripps Inst. of Oceanogr., San Diego, CA, 92037), Kevin D. Heaney (OASIS Inc., Fairfax Station, VA 22039), Jim Mercer (Univ. of Washington, Seattle, WA 98105), and Art Baggeroer (MIT, Cambridge, MA 02139)

The deep ocean experiment, PhilSea09, was conducted April–May, 2009, in the central part of the northern Philippine Sea. The deep 1000-m section of the distributed vertical line array (DVLA), which was composed of 30 elements (upper 10 at 90 m spacing starting at 4285 m depth, and 5 m spacing for the deepest 20 elements over the depths 5185–5280 m), recorded primarily four types of sounds not created as part of the experiment: wind noise, ship noise, airgun signals, and earthquake T-phases. The statistical properties of these sounds are examined quantitatively using non parametric statistical tests operating on the single element and beam level narrowband envelope time series. These statistical tests include the Wald-Wolfowitz runs test for mutual independence of the data samples, the Kolmogorov-Smirnov two-sample test for stationarity, and the Lillifors test for Gaussianity. One result is that these sounds, except for wind-dominated noise, fail the test for Gaussianity. Higher-order spectral analysis is performed to quantify the degree of non-Gaussianity. In addition, analytical probability density functions are fit to the histograms of the envelope values. Physics-based models are developed to predict the statistical characteristics of some of these sounds. [Work supported by the Office of Naval Research.]

1:15

4pUWb2. Ambient noise bathymetric domains. Donald Ross (Wesley Palms Rm 325, 2404 Loring St., Box 101 San Diego, CA 92109, donaldnmiross@mac.com), Megan F. McKenna, Sean M. Wiggins, and John A. Hildbrand (Univ. of California San Diego, La Jolla, CA 92093)

For the purposes of describing and understanding ambient sea noise for frequencies below about 300 Hz, most of the world's seas can be classified as belonging to one of three bathymetric domains. These domains are distinguished by their proximity to shipping lanes and by the degree to which they are exposed to noises originating at long distances. The three domains display different short-term characteristics as well as different historical patterns. In this paper, the three domains are described and typical ambient noise characteristics for each are shown, including changes which are attributable to increased ocean commerce.

1:30

4pUWb3. Unintended consequences of recent changes in ship traffic. Megan F. McKenna, Sean M. Wiggins, John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, Ritter Hall 200E, 8635 Kennel Way, La Jolla, CA 92093-0205, megan.mckenna@gmail.com), and Donald Ross (San Diego, CA 92109, USA)

Underwater ambient noise levels measured off the coast of southern California were correlated with regional changes in commercial shipping trade. Between 2007 and 2010, two events occurred that resulted in a decrease in ship traffic in the Santa Barbara Channel: the economic recession and a coastal air-quality improvement rule. From October 2005 to June 2010, monthly low-frequency ambient noise levels at a site 3 km from a major shipping route were compared to regional traffic levels. Two different metrics of ship traffic showed that on average a 1 dB reduction in low-frequency noise levels resulted from a decrease in traffic by one ship passage per day in a coastal basin.

1:45

4pUWb4. Passive acoustic monitoring of vessel presence at Rose Atoll and the National Park of American Samoa. Pollyanna Fisher-Pool (Joint Inst. for Marine and Atmospheric Res., Univ. of Hawaii, Honolulu, Hawaii, pollyanna.fisher-pool@noaa.gov), Marc O. Lammers, Lisa M. Munger (Univ. of Hawaii, Kaneohe, Hawaii), Kevin Wong, and Russell E. Brainard (Pacific Islands Fisheries Sci. Ctr., Honolulu, Hawaii)

American Samoa is in the process of evaluating the development of a network of marine protected areas (MPAs) to preserve coral reef environments and to prevent the decline of fish populations. Two long-standing MPAs in American Samoa are Rose Atoll Marine National Monument (RAMNM) and coastal marine regions within the National Park of American Samoa (NPSA) system. NPSA includes areas on the populated island of Tutuila, while RAMNM is approximately 130 miles away from the nearest population. Both are protected marine reserves where commercial and public recreational fishing are restricted, although the size and remoteness of the locations create a challenging task for observation and enforcement. A growing management concern over the decline of large fishes and possible illegal fishing has prompted interest in vessel incidence within the two MPAs. We gathered evidence of vessel presence with the use of long-term, autonomous passive acoustic monitoring within the two MPAs. Here we present results of vessel detection within acoustic recordings collected 2009–2010 at RAMNM and 2006–2007 and 2008–2009 at the NPSA Tutuila location. Results from this study highlight the patterns and seasonality of vessel incidence and provide managers with information to assist enforcement.

2:00

4pUWb5. A high-fidelity model for mitigating underwater pile driving noise in a shallow ocean waveguide. Kevin L. Cockrell, Ann Stokes, and Dwight Davis (Appl. Physical Sci. Corp., 2445 Truxtun Rd. #200, San Diego, CA 92106)

Driving large piles into the seafloor, as is done when constructing offshore wind farms, produces high level underwater noise that can have an adverse effect on local marine life. This talk reviews an investigation of methods to mitigate this pile driving noise. A numerical model was used to simulate the structural vibrations in the pile and its coupling to the acoustic field in surrounding air, water and sediment. The simulated acoustic field in the immediate vicinity of the pile was then coupled into an ocean waveguide propagation model using a virtual-source technique to match the boundary conditions. These numerical models were used to assess the relative contribution of the air-borne, water-borne, and sediment-borne acoustic radiation to the noise level in the water-column at ranges up to several hundred meters from the pile. Various noise mitigation methods were simulated and compared. It was determined that a dewatered cofferdam, which places a layer of air between the vibrating pile and the seawater, has the potential to reduce the far-field noise level by approximately 20 dB. A practical method for creating a dewatered cofferdam during construction is the subject of ongoing work.

2:15

4pUWb6. A model for noise radiated by submerged piles and towers in littoral environments. Todd A. Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, Preston S. Wilson, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

Pile driving in shallow water during the construction of bridges and other structures can produce transient broadband noise of sufficient intensity to kill fish and disturb marine mammals. Sustained tonal noise radiated by towers supporting offshore wind turbines contains energy in frequency bands that may inhibit detection of coastal activities via passive sonar and seismic sensors. Understanding the generation and propagation of underwater noise due to pile driving and wind farms is important for determining the best strategies for mitigating the environmental impact of these noise sources. An analytic model, based on a Green's function approach, is presented for the sound radiated in the water column by a submerged cylindrical structure embedded in horizontally stratified layers of sediment. The sediment layers are modeled as viscoelastic media and the Green's function is derived via angular spectrum decomposition. Noise radiation due to both vibration of the structure and impulses delivered to the sediment is considered. Contributions to the pressure field in the water column due to radiation directly into the water, radiation from the sediment into the water, and Scholte waves propagating along the sediment-water interface will be discussed. [Work supported by the ARL:UT IR&D program.]

2:30

4pUWb7. Numerical analysis of sound radiation underwater from a fully submerged pile. Shima Shahab, Katherine F. Woolfe, and Mardi C. Hastings (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The study of acoustic radiation from pile driving is essential because the impact between the hammer and pile causes extremely high underwater sound pressure levels that are potentially harmful to the marine environment. When the hammer strikes a cylindrical steel pile, movement of the pile wall in circumferential, longitudinal, and radial directions can excite many modes of vibration. The radial expansions of the pile that propagate along the pile after impact create sound waves in the surrounding water. A finite-difference time-domain (FDTD) formulation is presented to analyze the sound produced from a fully submerged pile with simply supported ends. The pile considered is a cylindrical shell of finite length, bounded by sediment at the bottom end, and surrounded by and filled with water. Three coupled partial differential equations govern the vibration of the shell, and the effects of acoustic media including water and sediment are added in the radial direction. Results of a correlation between the radiated sound field predicted by the FDTD model and acoustic data from a scaled physical model of a fully submerged pile hit with an impact hammer are presented. [Work supported by the Oregon Department of Transportation and Georgia Institute of Technology.]

2:45

4pUWb8. A scaled physical model to study underwater noise from impact pile driving. Katherine F. Woolfe, Shima Shahab, Juan Morales, and Mardi C. Hastings (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

Development of computational models to predict underwater noise from impact pile driving is limited because of the difficulty and cost involved in collecting acoustic field data for model verification during construction activities. To alleviate this situation, a scaled physical model for marine pile driving was designed and implemented in a 500-gallon shallow water tank, 3.5 m long and 0.85 m wide. The scaled piles are steel pipes having lengths up to 1 m, and radius-to-wall thickness and length-to-radius ratios similar to large cast-in-shell steel (CISS) piles. The wavelength-to-depth ratio for the primary breathing mode of the fully submerged scaled piles and a fully submerged CISS pile of diameter 2.4 m and length 30 m is between 2.0 and 2.5. The impact force is generated and measured with an impulse hammer, and sound field data are collected using a small 2-D hydrophone array. Data are correlated with the results of numerical and analytical models developed to predict sound radiation from CISS piles to verify that the scaled physical model accurately represents their structural acoustics behavior. [Work supported by the Georgia Institute of Technology and the Oregon Department of Transportation.]

3:00

4pUWb9. Underwater structure-borne noise analysis using a finite element/boundary element coupled approach. Dooho Lee (Dept. of Mech. Eng., Dongeui Univ., Busan, 614-714, South Korea, dooho@deu.ac.kr), Hyun-Sil Kim, Bong-Ki Kim, and Seong-Hyun Lee (Acoust. Team, KIMM, Daejeon, South Korea)

Radiated noise analysis from ship structure is a challenging topic due to difficulties in accurate calculation of fluid-structure interaction as well as massive degree of freedoms of the problem even in the case of finite element/boundary element coupled formulations. To reduce the severity of the problem, a new added mass and damping approach is proposed in this paper. The complex frequency-dependent added mass and damping matrices are calculated by using the high-order Burton-Miller boundary element formulation in order to obtain accurate values over all frequency bands. The calculated fluid-structure interaction effects are added to the structural matrix calculated by commercial finite element software, MSC/NASTRAN. An iterative solver is introduced to solve the eigenvalue problem because the combined system matrices are complex and frequency-dependent. The calculated eigenfrequencies for a submerged cylindrical shell show good agreement on reference data. The accuracy and efficiency of the present formulation is compared with those from conventional finite element/boundary element formulation. The comparison results show that the present formulation has better accuracy than the conventional one because of accurate added mass and damping calculation.

3:15

4pUWb10. Non-cavitating propeller noise source inversion. Woojae Seong, Jaehyuk Lee, Ara Hyun (Dept. of Ocean Eng., Seoul Natl. Univ., Seoul 151-742, Korea, wseong@snu.ac.kr), and Junghoon Lee (Samsung Heavy Industry, Taejon, Korea)

Ship's propeller cavitation generates major inboard noise and vibration over the aft body surface of a ship. During the last decade, cavitation-induced hull pressure forces have been reduced considerably leading to reduced noise and vibration levels owing to cavitation, in which case the non-cavitating components of propeller excitations have to be considered: pressure fluctuations due to blade loading and blade thickness. In this work, an algorithm to invert for the non-cavitating propeller noise source parameters based on the experimental data in a cavitation tunnel is shown. The fluctuating hull pressure is estimated by forward modeling based on acoustic boundary element method (BEM). The propeller blade loading and thickness noise sources are modeled using rings of dipoles and quadrupoles, respectively. The inversion for these pseudo sources are carried out by matching the data obtained from a cavitation tunnel experiment, where several hydrophone are flush mounted on the ship's surface near the propeller. Proper inversion results are obtained when six or more dipole and quadrupole rings are placed at each blade. Using the inverted source parameters, hull pressures were estimated, which show good agreement with the experiment data.

THURSDAY EVENING, 3 NOVEMBER 2011 7:30 TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

Animal Bioacoustics	Pacific Salon 1
Noise	Royal Palm 1/2
Speech Communication	Royal Palm 5/6
Underwater Acoustics	Pacific Salon 3