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

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.

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.]

4a THU. AM

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 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.

4a THU. AM

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.