Session 2aAA

Architectural Acoustics, Psychological and Physiological Acoustics, Engineering Acoustics, and Noise: Binaural and Spatial Evaluation of Acoustics in Performance Venues

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Chair's Introduction—7:55

Invited Papers

8:00

2aAA1. Modeling binaural processing: What next? Jens Blauert (Ruhr-University Bochum, Communication Acoustics, Bochum 44780, Germany, jens.blauert@rub.de)

Models of binaural processing are traditionally based on signal-processing in a bottom-up (signal-driven) architecture. Such models are sufficient for a number of technologically important applications, such as perceptual coding, sound-source identification and localization, dereverberation and decoloration, but fail when applications require cognition. Future models will thus include symbol processing in addition to signals processing, will have inherent knowledge bases and employ top-down (hypothesis-driven) strategies in addition to bottom-up ones. Some of these features are known from automatic speech recognition and may be generalized for broader application, e.g., blackboard structures. With the new models more sophisticated applications may be approached, for instance, quality evaluation and assessment on the basis of internal references, such as needed to determine estimates of the quality of performance spaces and/or audio systems. Further, to enable autonomous learning, future models will employ feed-back loops to realize active exploratory actions. Some of these features can be imported from recent research in robot audition. In our contribution, we shall, among other things, report on ideas and concepts as currently discussed in AABBA, an international grouping of 14 laboratories in Europe and the US that are dealing with auditory assessment by means of binaural algorithms.

8:20

2aAA2. Sound quality from binaural and multidirectional measurements. David H. Griesinger (Research, David Griesinger Acoustics, 221 Mt Auburn St #504, Cambridge, MA 02138, dgriesinger@verizon.net)

There has been rapid progress in methods to gather binaural and multi-directional point-to-point impulse responses from unoccupied venues. With individual matching of headphones to a listener this data can sometimes be auralized to obtain a glimpse of the sound of a venue. But numerical methods to analyze such data in order to quantify the precise sound of a particular seat remain elusive. This paper will discuss some of the limitations of point measurements, recent work on binaural technology, and impulse response analysis techniques that are not based on sound energy, but on the methods used by the ear and brain to aurally perceive music, speech, and sonic environments. The goal - nearly in sight - is developing methods for obtaining objective quality assessments from binaural or multidirectional recordings of live music and speech.

8:40

2aAA3. Modeling binaural suppression processes for predicting speech intelligibility in enclosed spaces. Vanessa Li (Graduate Program in Architectural Acoustics, School of Architecture, Rensselaer Polytechnic Institute, Troy, NY 12180, vanessa.li@gmail.com), Ning Xiang, and Jonas Braasch (Graduate Program in Architectural Acoustics, School of Architecture, Rensselaer Polytechnic Institute, Troy, NY)

Speech from a target speaker reaches a listener via multiple paths in a room due to room reflections. As sound waves from the target speaker approach the listener, degradation to the signal is caused by ambient noise and reverberant energy. The speech transmission index (STI) is a commonly used metric for predicting speech intelligibility accounting for both noise and reverberation. This metric, however, is a monophonic measure that does not take into consideration binaural cues used for unmasking undesired effects. As a result, using the STI on its own tends to under-predict intelligibility under binaural listening conditions. The proposed research aims to improve speech intelligibility predictions with the presence of room effects by implementing a psychophysical binaural model as a front-end to the STI calculation. The equalization-cancellation (EC) theory is applied to spatially unmask noise, while late incoherent reverberant energy is suppressed by applying a weighting function based on interaural coherence. Preliminary comparisons between listening tests and model predictions reveal promising results, indicating a useful tool in acoustical planning in addition to further study into binaural suppression processes.

9:00

2aAA4. Utilizing head movements in the binaural assessment of room acoustics and analysis of complex sound source scenarios. Jonas Braasch, Anthony Parks, Torben Pastore, and Samuel W. Clapp (School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180, braasj@rpi.edu)

The measurement of transfer functions is currently standard practice for the acoustic evaluation of performance venues. The pathway between a measurement loudspeaker and a microphone or binaural manikin in a room can be treated as a linear time-invariant system, and meaningful acoustical parameters can be derived from measured impulse responses. Unfortunately, this method neglects that human listeners typically move their heads when exploring an acoustic venue. This paper addresses these implications when designing systems to take head movement into account. A number of approaches will be discussed based on existing research and technology at Rensselaer, including a binaural manikin with a motorized head, a technique to simulate head movements from impulse responses recorded with a higher-order spherical microphone, and a binaural model that can process head movements. The model distinguishes between a room coordinate system and a head-related coordinate system. Its binaural activity map is rotated with head movements in order to separate front-back images, resolve the reverberation-reduced angle of lateral sound sources, assess different surround loudspeaker configurations for immersive sound systems, and separate acoustic sources. The research presented here has received support from the National Science Foundation under Grant No. 1002851.

9:20

2aAA5. Using a higher-order spherical microphone array to assess spatial and temporal distribution of sound in rooms. Gary W. Elko (mh Acoustics LLC, 25A Summit Ave, Summit, NJ 07901, gwe@mhacoustics.com) and Jens M. Meyer (mh Acoustics LLC, Fairfax, NJ)

We have developed a spherical microphone called the Eigenmike® microphone array that is capable of achieving up to third-order spherical harmonics decomposition of the sound field. One potential use of the spherical array is to investigate the spatial nature of sound fields in rooms. In this talk, we will show some measurement results where the processed data from an Eigenmike® array is used to compute various energy ratios between the direct, lateral, rear, and floor and ceiling directions. We will also show some other simple measures that might be useful in the spatial analysis and characterization of room acoustics.

9:40

2aAA6. Reciprocal binaural room impulse response measurements. Johannes Klein, Martin Pollow, Janina Fels, and Michael Vorlaender (ITA, RWTH Aachen University, Neustr. 50, Aachen 52066, Germany, mvo@akustik.rwth-aachen.de)

Multi-channel spherical loudspeakers have been introduced in shapes of cubes, dodecahedra, or higher-order discrete representations of spheres. In this contribution a spherical source with a partial Gaussian distribution of 28 channels is presented. With sequential measurements and rotation of the sphere a radiation of effectively 23rd order of spherical harmonics can be obtained. Accordingly directional patterns of not only sound sources but also of receivers such as HRTF can be modeled in detail up to quite high frequencies. The high order of spherical harmonics allows investigation of individual differences of pinna cues. When applied in a reciprocal measurement of room impulse responses in performance venues, an almost perfect omnidirectional microphone on the stage and an HRTF source in the audience can be used to study spatial room acoustic parameters such as early lateral energy fractions, late lateral strength and IACC of dummy heads and individuals. This is obtained by post processing of just one set of multi-channel impulse responses in the venue. Opportunities and challenges of this approach will be discussed.

10:00–10:20 Break

10:20

2aAA7. Measuring and inferring the directional properties of the early room response. Jonathan Botts, Samuel Clapp, Ning Xiang, and Jonas Braasch (Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th St., Greene Building, Troy, NY 12180, botts.jonathan@gmail.com)

In an effort to understand the response of a room more completely, spherical microphone arrays have been used to produce a three-dimensional map of a room impulse response. To locate reflections from various room surfaces, the most common approach is to search for peaks in the beamformed response. Particularly when using this approach with a low-order array there is no way to distinguish a side-lobe from an additional arrival. Furthermore, overlapping arrivals skew the maxima of beampatterns, resulting in incorrect inferences. This talk seeks to demonstrate that a Bayesian, model-based analysis of the data addresses the complete problem of image source estimation, with mechanisms to determine the number and locations of simultaneous arrivals. Particularly with low-order arrays, substantially more accurate estimates can be made, which both increases the overall quality of the analysis and extends the portion of the impulse response that may be reliably analyzed.

10:40

2aAA8. A measurement technique achieving high spatial resolution for sound sources within a performance venue. Alex Case (Sound Recording Technology, University of Massachusetts, Lowell, MA 01854, alex@fermata.biz), Agnieszka Roginska, and Jim Anderson (New York University, New York, NY)

A proof of concept for gathering high spatial resolution sound radiation, from near field to far field, in 3 dimensions around an electric guitar amplifier is presented, with an eye and ear toward applying a similar technique to other essential sound sources. A high density microphone array is used to gather many thousands of impulse response in a hemi-anechoic space. The resulting data serves as a useful input to room models and auralizers, but finds added purpose as an educational tool in musical acoustics and sound recording.

11:00

2aAA9. Comparison of headphone- and loudspeaker-based concert hall auralizations. Samuel Clapp (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Greene Building, Troy, NY 12180, clapps@rpi.edu), Anne Guthrie (Arup Acoustics, New York, NY), Jonas Braasch, Ning Xiang, and Terence Caulkins (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, Troy, NY)

In this research, a spherical microphone array and a dummy head were used to measure room impulse responses in a wide variety of concert and recital halls throughout New York State. Auralizations were created for both headphone playback and second-order ambisonic playback via a loudspeaker array. These two systems were first evaluated objectively to determine the level of accuracy with which they could reproduce the measured soundfields, particularly with respect to important binaural cues. Subjects were then recruited for listening tests conducted with both reproduction methods and asked to evaluate the different spaces based on specific parameters and overall subjective preference, and the results of the two playback methods were compared.

11:15

2aAA10. A spatial encoding method for measured room impulse responses. Sakari Tervo, Jukka Pätynen, and Tapio Lokki (Department of Media Technology, Aalto University School of Science, P.O. Box 15500, Espoo FI00076 Aalto, Finland, sakari.tervo@aalto.fi)

The spatial information contained in measured room impulse responses can be used to explain some of the acoustical properties of performance spaces. This paper presents a spatial encoding method, which can extract accurate spatial information from impulse responses that are measured with at least four microphones in an open 3-D array. The method is based on decomposing a spatial room impulse response into a set of image-sources, i.e., every single sample in the impulse response is considered as an image source. Each of the image-sources is localized with an acoustic source localization method, which depends on the applied microphone array and the acoustic conditions. Due to the image-source presentation, the presented method can be applied to any compact array and used in conjunction with variety of current spatial loudspeaker reproduction systems to create convolution reverb-type spatial sound reproduction. The method allows static and interactive binaural reproduction via virtual loudspeaker arrays. The presentation includes demonstrations with a binaural reproduction system.

11:30

2aAA11. Source locations, listener locations, and measurement devices when making acoustical measurements in performance spaces. Elizabeth Lamour (University of Kansas, Lawrence, KS 66045, lizlamour@gmail.com)

Does the location of the source affect the results of an acoustical measurement? This is the question that sparked the author's Master's Project which explores the differences between measurements taken with a source located on the stage of a performance space and a source located higher above the stage using the space's existing sound system. Impulse responses were gathered from four different performance halls with respect to source location, microphone location, and measurement devices used. Comparisons were made between trends in reverberation time, early decay time, and interaural cross-correlation coefficient. The results are not only interesting, but they also question the typical measurement practices of acousticians and confirm assumptions made regarding important acoustical characteristics of performance spaces.

11:45

2aAA12. Applying direct algebraic sound source localization method for time-domain reflectometry of conference room. Tsukassa Levy and Shigeru Ando (Information Physics and Computing, Tokyo University, Tokyo-to Bunkyou-ku Hongou, Tokyo 113-0033, Japan, levy.t@alab.t.u-tokyo.ac.jp)

A novel localization method has been previously exposed, using an explicit formula of direction and distance of a monopole source and a circular array [S. Ando, ASA Seattle Meeting, 2011]. However, this localization method has little been applied in real environments, such as concert halls or conference rooms. It has also been shown that the algorithm has a high temporal resolution [T.Levy, S. Ando, Hong Kong Acoustics 2012, 2012] that enables to localize sound sources using reflected sound waves. Thus, the aim of this study is to study conference room's reflection using the proposed algorithm and to test its robustness and its efficiency in such conditions. In the experiments, the main reflectors in the conference room will be identified and a comparison between the previously proposed method and the traditional sound source localization algorithms is done, in terms of rapidness and precision.

Session 2aAB

Animal Bioacoustics and Acoustical Oceanography: Acoustics as Part of Ocean Observing Systems

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Lora J. Van Uffelen, Cochair
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Chair's Introduction—7:55

Invited Papers

8:00

2aAB1. Acoustic tomography as a component of the Fram Strait observing system. Brian D. Dushaw, Hanne Sagen, Stein Sandven (Nansen Environ. and Remote Sensing Ctr., N-5006, Norway, brian.dushaw@nersc.no), and Peter Worcester (Scripps Institution of Oceanography, UCSD, La Jolla, CA)

The Fram Strait, a deep constriction between Svalbard and Greenland, is the primary location for the exchange of heat, mass and freshwater between the Arctic and Atlantic Oceans. With existing data and ocean modeling, current estimates of these exchanges, critical for understanding the Arctic Ocean climate, are inaccurate. To try to improve these estimates, during 2008-9 the DAMOCLES project deployed a test tomography path spanning the deep, ice-free part of the northward-flowing West Spitzbergen Current (WSC). Small-scale scintillations of sound speed due to eddies, fronts, and internal waves, are an important aspect of acoustic propagation of the region. Variability within Fram Strait, and the WSC in particular, is characterized by ubiquitous mesoscale eddies with 20-km scale. These eddies extend to depths of several hundred meters. Understanding the forward problem is essential for the inversion of acoustic data. The sound speed environment of Fram Strait generally prevents individual ray arrivals from being resolved in O(100-km) acoustic paths. An accurate inversion of these data for path-averaged sound speed (temperature) can be still be obtained, however. An objective mapping study, combining acoustic and existing data types, demonstrates that tomography will be a valuable and effective addition to the Fram Strait observing system.

8:20

2aAB2. Long-term acoustic monitoring off Central California. Curtis A. Collins, John E. Joseph, Ching Sang Chiu, John Colosi, and Christopher W. Miller (Oceanography, Naval Postgraduate School, 833 Dyer Rd, Monterey, CA 93943-5122, collins@nps.edu)

The use of moored ocean arrays for environmental acoustic measurements off Central California is discussed. A cabled array off Point Sur, CA, which was designed for long-range, low-frequency listening was used by NPS and collaborators from late 1997 through mid-2000 and provides examples of a wide range of activities including use for student laboratories, faculty and student research, as well as monitoring, e.g. ambient acoustic noise, test ban treaty activities. From mid-2006 to present, passive acoustic data have continued to be collected off Pt. Sur using single hydrophone moored autonomous listening stations which record data intermittently at sampling rates of 200 kHz. We have recently considered re-establishment of cabled passive acoustic measurements using MARS, an example of an observatory which was designed and located for more traditional oceanographic studies. The utility of MARS for acoustic measurements depends both on how well it can characterize the regional acoustic environment as well as local oceanographic processes that can be resolved acoustically (canyon effects, geography, sound speed variability, sediments and local vessel traffic). These can be contrasted with existing cabled and autonomous data from Point Sur.

8:40

2aAB3. Acoustics at the ALOHA Cabled Observatory. Bruce M. Howe (Ocean and Resources Engineering, University of Hawaii at Manoa, 2540 Dole St, Holmes Hall 402, Honolulu, HI, bhowe@hawaii.edu), Fred Duennebier (Geology and Geophysics, University of Hawaii at Manoa, Honolulu, HI), Roger Lukas (Oceanography, University of Hawaii at Manoa, Honolulu, HI), and Ethan Roth (Ocean and Resources Engineering, University of Hawaii at Manoa, Honolulu, HI)

Since 6 June 2011, the ALOHA Cabled Observatory (ACO) has been collecting ocean acoustic data, continuing an earlier data set covering February 2007 - October 2008. The ACO is at Station ALOHA 100 km north of Oahu, the field site of the Hawaii Ocean Time-series (HOT) program that has collected biological, physical, and chemical oceanographic data since 1988. At 4728 m water depth, it is the world's deepest operating cabled observatory. ACO provides power and communications to user instrumentation. Among the instrumentation there are two hydrophones 1 m off the bottom separated by 1 m. One is an OAS Model E-2PD meant for low frequencies (0.014 Hz to 8 kHz). A second (uncalibrated) hydrophone is meant for higher frequencies. Current sampling rates for both hydrophones are 96 kHz; subsampled 24 kHz data are streamed to the Web in real-time. The system will be described and examples of acoustic events and signals presented, including local and distant earthquakes, marine mammals, surface waves, wind, rain, ships, sonars, and implosions. Plans for future acoustics research will be discussed. [Work supported by the National Science Foundation.]

9:00

2aAB4. On the region of feasibility of interference alignment for underwater acoustic communication. Dario Pompili (Electrical and Computer Engineering, Rutgers University, 64 Brett Road, Piscataway, NJ 08854, pompili@cac.rutgers.edu)

To enable underwater applications such as coastal and tactical surveillance, undersea explorations, and picture/video acquisition, there is a need to achieve high data-rate underwater acoustic communications, which translates into attaining high acoustic channel spectral efficiencies. Interference Alignment (IA) technique, which has recently been proposed for radio-frequency MIMO terrestrial communication systems, aims at improving the spectral efficiency by enabling nodes to transmit data simultaneously at a rate equal to half of the interference-free channel capacity. The core of IA lies in designing transmit precoding matrices for each transmitter such that all the interfering signals align at the receiver along a direction different from the desired signal. To decode, the receiver projects the received signal onto a decoding vector that is orthogonal to the data vector of interfering signal. While promising, there are challenges to be solved for the use of IA underwater, i.e., 1) imperfect acoustic channel knowledge, 2) high computational complexity, and 3) high communication delay. We study the feasibility of IA in underwater environment under these challenges; we also propose a distributed computational framework to parallelize the iterative IA algorithm and determine to what extent we can parallelize it among neighboring nodes under different channel coherence times.

9:20

2aAB5. Recent development and application of active acoustic techniques for studies of zooplankton ecology and implications for ocean observatories. Gareth L. Lawson, Andone C. Lavery, Peter H. Wiebe, Jonathan R. Fincke, and Nancy J. Copley (Woods Hole Oceanographic Institution, 266 Woods Hole Rd, Woods Hole, MA 02543, glawson@whoi.edu)

High-frequency active acoustic techniques enjoy a long history in the study of zooplankton ecology and increasingly are being incorporated into ocean observing systems, addressing a pressing need for zooplankton-sampling capabilities. Discriminating among sources of scattering remains a key problem in ecological applications of active acoustics, however, especially when deploying on autonomous platforms, where independent sampling with nets or optics to verify acoustic observations is often not feasible. Here we consider the ecological insights that can be afforded by active acoustic methods and implications to the design of ocean observing systems by reporting on (1) a series of recent field studies of krill ecology employing both a traditional multi-frequency system (43, 120, 200, 420 kHz) and a recently-developed broadband system (30-600 kHz) designed to provide enhanced capabilities for discrimination of scattering sources, and (2) test deployments on autonomous platforms of a low-power active acoustic system capable of broadband or narrowband transmission. Comparisons to concurrent sampling with a depth-stratified net system, when available, allow an assessment of the abilities of these acoustic systems for remotely discriminating among sources of scattering and for estimating the abundance and size of animals.

9:40

2aAB6. Passive acoustics monitoring as part of integrated ocean observing systems. Joseph J. Luczkovich (Biology/Institute for Coastal Science and Policy, East Carolina University, 383 Flanagan Building, Greenville, NC 27858, luczkovichj@ecu.edu), Mark W. Sprague (Physics, East Carolina University, Greenville, NC), Cecilia S. Krahforst (Coastal Resources Doctoral Program, East Carolina University, Greenville, NC), D. Reide Corbett, and John P. Walsh (Geological Sciences & Institute for Coastal Science and Policy, East Carolina University, Greenville, NC)

Passive acoustic monitoring can be a useful tool to include on Ocean Observing Systems. As an example, we describe the monitoring the acoustic environment in the coastal waters of North Carolina (USA) using an instrumented platform. The ECU Itpod (instrumented tripod) has been deployed in several locations in Pamlico Sound and river estuaries since 2006 to study fishes in the Family Sciaenidae (drums and croakers). We will present data recorded with hydrophones deployed on the Itpod with remote data loggers, acoustic Doppler current profilers, turbidity meters and water quality instruments. We have used passive acoustic recordings to study the correlations of fish sounds and environmental parameters (temperature, salinity, turbidity, dissolved oxygen, wave action, river discharge, tropical storms). The long-term data suggest that spring temperature increases are associated with increased activity of acoustically mediated courtship and spawning behavior of sciaenid fishes; these sounds decline in the fall as water temperature declines. In addition, we have observed acoustic interactions between marine mammal predators and their fish prey and the effects of noise from tugs and small boats on fish sound production. Itpods must be recovered periodically to recover data and replenish batteries; solar-powered platforms and automated fish detection algorithms are under development.

10:00–10:30 Break

10:30

2aAB7. The power of acoustics in ocean observing systems: A case study in the Bering Sea. Jennifer L. Miksis-Olds and Laura E. Madden (Applied Research Laboratory, The Pennsylvania State University, PO Box 30, State College, PA 16804, jlm91@psu.edu)

Acoustic time series are incredibly powerful as independent data sets. Passive acoustic recordings provide information on environmental sound levels, the presence of vocalizing animals, surface conditions, marine precipitation, and anthropogenic activities within the area of acoustic coverage. Active acoustic systems provide a time series of acoustic backscatter from which biological scatter can be measured and quantified to provide estimates of relative abundance and numerical density. The combination of acoustic technology with other hydrographic sensors within an ocean observing system now affords the opportunity to develop an understanding of ecosystem dynamics ranging from the physical oceanographic conditions to the distribution and behavior patterns of top predators. This is especially critical in sub-Arctic regions like the Bering Sea where rapid changes associated with climate change are having impacts at multiple levels. Here we discuss the environmental parameters that are the best predictors of different marine mammal species as determined through generalized linear and general additive mixed models. Predictor variables considered were percent ice cover, ice thickness, sound level at five frequencies, and percent composition of 4 biologic scattering groups.

10:50

2aAB8. A low cost, open source autonomous passive acoustic recording unit for recording marine animals. Robert D. Valtierra (Dept. Mechanical Engineering, Boston University, 110 Cummington St., Boston, MA 02215, rvaltier@bu.edu), Sofie M. VanParijs (Northeast Fisheries Science Center, National Oceanic and Atmospheric Administration, Woods Hole, MA), R. G. Holt, Connor Mace, Kara Silver, and Chris Bernard (Dept. Mechanical Engineering, Boston University, Boston, MA)

An autonomous passive acoustic recording unit (ARU) was developed through a collaboration between the Boston University Department of Mechanical Engineering and the NOAA Northeast Fisheries Science Center. The ARU consists of two main sections, an electronic data logger and a mechanical pressure case and release. The datalogger makes use of widely adopted commercial hardware such as SD card memory and USB connectivity. In addition, WAV file formats and open-source compiler software allow flexibility and programmability at minimal expense. The pressure case was designed for shallow water (100 m) applications with few machined parts and several "off the shelf" parts. The overall system can be constructed at a minimal cost and has been successfully tested during both laboratory and at-sea trials.

11:05

2aAB9. Autonomous detection of neotropical sciaenid fishes. Sebastian Ruiz-Blais, Mario R. Rivera-Chavarria (Centro de Investigaciones en Tecnologías de la Información y Comunicación, Universidad de Costa Rica, Sede "Rodrigo Facio Brenes" Montes de Oca, San José 2060, Costa Rica, mariorrivera@gmail.com), and Arturo Camacho (Escuela de Ciencias de la Computación en Informática, Universidad de Costa Rica, San Jose, Costa Rica)

Sciaenid passive acoustics are a demonstrated valuable tool for fisheries management. In spite of this, an efficient software tool to detect and identify fish sounds is not currently available. Such tool would be useful for autonomous recognition and array methodologies. For Neotropical environments this lack is even more conspicuous since the availability of corroborated sciaenid sounds is limited. We are developing such tools using corroborated Cynoscion squamipinnis (Pisces: Sciaenidae) sounds. Our approach is based on timbre statistics, short and long-term partial loudness, and the 30 Hz typical pattern found on the signal's stridulations. Relevant fish drums are detected through empirically found fix thresholds for the timbre statistics and the 30 Hz pattern, and a dynamic threshold established by an unsupervised algorithm based on the long-term loudness. Current results show a recognition rate of 80%. Despite these promising numbers, there are still challenges ahead. In the future, we plan to incorporate other variables that affect underwater sound characteristics such as depth, source level distance, and physical chemical properties, which may be crucial to make a user friendly, accurate, and practical tool, for neotropical marine environmental managers. We also plan to extend this method to other soniferous coastal fish.

11:20

2aAB10. Fish recordings from NEPTUNE Canada. Ana Sirovic, Sophie Brandstatter, and John A. Hildebrand (Scripps Institution of Oceanography, 9500 Gilman Drive, La Jolla, CA 92093-0205, asirovic@ucsd.edu)

NEPTUNE Canada is a regional-scale ocean observing system deployed off the west coast of Vancouver Island, Canada. Among the data streams broadcast live over the internet are video collected using black and white low-light camera and audio collected with Naxys hydrophone (5 - 3,000 Hz). These data allow for description of sound production by fishes in the vicinity of the system. Concurrent video and hydrophone data are available from the Barkley Canyon node (~900 m depth). While the hydrophone recordings were continuous, strobes for video are only turned on during short, irregular (~10 min) intervals. Approximately 30 h of concurrent video and audio recordings were analyzed. The most commonly seen fish was sablefish (*Anoplopoma fimbria*), and the most common fish-like sound was a broadband, short pulse that occurred on nearly half of the recordings. On approximately one-fifth of concurrent video and audio recordings both sablefish and fish-like pulsed sounds were detected. It may be possible to use these sounds to monitor sablefish abundance across the northeastern Pacific Ocean. *NEPTUNE Canada Data Archive*, http://www.neptunecanada.ca, hydrophone and video data from May, June, August, and December 2010 and January and February 2011, Oceans Networks Canada, University of Victoria, Canada. Downloaded 2012.

11:35

2aAB11. Tracking and source level estimation of multiple sperm whales in the Gulf of Alaska using a two-element vertical array. Delphine Mathias, Aaron M. Thode (Marine Physical Lab, Scripps Institution of Oceanography, 9500 Gilman Drive, La Jolla, CA 92037-0238, delphine.mathias@gmail.com), Jan Straley (University of Alaska Southeast, Sitka, CA), and Russel D. Andrews (School of Fisheries and Ocean Sciences, University of Alaska Fairbanks, Fairbanks, AK)

Between 15-17 August 2010 a two-element vertical array (VA) was deployed in 1200 m deep water off the continental slope of Southeast Alaska. The array was attached to a longline fishing buoyline at 300 m depth, close to the sound-speed minimum of the deep-water profile. The line also attracted seven depredating sperm whales to the area, each generating impulsive 'clicks' that arrived on the VA via multiple ray paths. The propagation model BELLHOP was used to model relative arrival times and vertical elevation angles of click ray paths as a function of depth and range from the VA. The resulting tracking system yielded range-depth tracks of multiple animals out to at least 35 km range. These locations, along with the transmission loss estimates of the model, permitted the sound source levels to be recovered. Here we present the consistency of source levels from individuals over time, the degree of source level variation between individuals, and possible correlations between inter-click interval and source level. This analysis suggests how a relatively simple ocean observing acoustic system could localize bioacoustic signals over large ranges, given the appropriate deployment configuration.

11:50

2aAB12. Acoustic thermometry as a component of the global ocean observing system. Brian D. Dushaw (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105-6698, dushaw@apl.washington.edu)

Acoustic data acquired during the 1995-2006 Acoustic Thermometry of Ocean Climate (ATOC) program were used to test the accuracy of ocean state estimates of the North Pacific obtained by various means: simple forward integration of a model, objective analysis of hydrographic and altimeter data, and data assimilation using general circulation models. The comparisons of computed and measured time series stringently tested the accuracy of the state estimates. The differences were substantial, indicating that acoustic thermometry provides unique information about the large-scale temperature. On some acoustic paths, changes in temperature occurring over time scales of weeks with magnitudes comparable to the seasonal cycle were observed. Acoustic thermometry offers valuable constraints on the large-scale thermal variability for the ocean observing system. Acoustic tomography was accepted as part of the Ocean Observing System during the OceanObs'99 and '09 international workshops. Sources and receivers of acoustic thermometry can serve multiple purposes. Hydrophone arrays are used to study a wide range of human, biological, and geological activity. Acoustic sources can transmit signals that can be used to track drifting instrumentation. A modest number of active and passive acoustic instruments deployed worldwide can form a general purpose global acoustic observing network.

Session 2aBA

Biomedical Acoustics and Physical Acoustics: Modeling of Nonlinear Medical Ultrasound

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Chair's Introduction—8:00

Invited Papers

8:05

2aBA1. Full-wave nonlinear ultrasound simulation on distributed clusters using the k-space pseudospectral method. Bradley E. Treeby, Jiri Jaros (Research School of Engineering, Australian National University, Canberra, ACT 0200, Australia, bradley.treeby@anu.edu.au), Ben T. Cox (Department of Medical Physics and Bioengineering, University College London, London, United Kingdom), and Alistair P. Rendell (Research School of Computer Science, Australian National University, Canberra, ACT, Australia)

Performing realistic simulations of the propagation of nonlinear ultrasound waves through biological tissue is a computationally difficult task. This is because the domain size of interest is often very large compared to the wavelengths of the high-frequency harmonics generated as the ultrasound waves progress. Recently, the k-space pseudospectral method has been applied to this problem to reduce the number of grid points required per wavelength compared to finite difference methods. However, the global nature of the spectral gradient calculations used in this method introduces new challenges for tackling large-scale problems. Here, we discuss three important issues for pseudospectral methods in the context of distributed computing. (1) Decomposing the domain to allow distribution across multiple nodes while still retaining the global accuracy of the spectral gradient calculations. (2) Using non-uniform grids to allow grid points to be clustered around highly nonlinear regions. (3) Avoiding aliasing errors due to modeling nonlinear wave propagation on a fixed grid. For each issue, solutions that retain the efficiency advantages of the pseudospectral method are discussed. We then present recent results of large-scale 3D nonlinear ultrasound simulations in heterogeneous and absorbing media running on both shared memory computers and distributed computer clusters.

8:25

2aBA2. Numerical simulations of three-dimensional nonlinear acoustical waves: Application to the modeling of helicoïdal beams. Régis Marchiano (Institut Jean le Rond d'Alembert (UMR CNRS 7190), University Pierre and Marie Curie, 4 place Jussieu, Paris 75005, France, regis.marchiano@upmc.fr), Jean-Louis Thomas, and Diego Baresch (Institut des NanoSciences de Paris (UMR CNRS 7588), University Pierre and Marie Curie, Paris, France)

A numerical method for the simulation of three-dimensional nonlinear acoustical wave propagation through a homogeneous or weakly heterogeneous medium is presented. This method is based on the resolution a nonlinear wave equation taking into account diffraction, nonlinearities and weak heterogeneities exact up to second order. It is numerically solved in a one-way manner by using a classical splitting method in three steps: angular spectrum method for diffraction, implicit finite differences method for heterogeneities, and semi-analytical Burgers-Hayes method for nonlinearities. Simulations of propagation of nonlinear helicoïdal beams will illustrate the capacities of the method. This kind of acoustical beams featuring radial, azimuthal and axial variations of the field is intrinsically three-dimensional. Furthermore, they have properties of potential interest in biomedical acoustics either for imaging purposes (dynamics of the so-called topological charge) or for therapy purposes (generation of helical shock front or acoustical tweezers with radiation force).

8:45

2aBA3. Rationale behind the Iterative Nonlinear Contrast Source method. Martin D. Verweij, Libertario Demi, and Koen van Dongen (Laboratory of Acoustical Wavefield Imaging, Faculty of Applied Sciences, Delft University of Technology, Delft 2628CJ, Netherlands, m.d.verweij@tudelft.nl)

Modern medical echoscopy increasingly relies on imaging modalities that exploit the features of nonlinear ultrasound. The development of these modalities and corresponding dedicated transducers requires accurate simulations of pulsed nonlinear acoustic wave fields in realistic biomedical situations. This involves nonlinear media with frequency power law attenuation and spatially dependent acoustic properties. Simulations frequently concern strongly steered beams, hundreds of wavelengths long, and their grating lobes. The Iterative Nonlinear Contrast Source (INCS) method is a full-wave method that has been developed for this purpose. It treats the nonlinear term in the Westervelt equation as a contrast source that operates, alongside other source terms, in a homogeneous linear 'background' medium. The background Green's function is known analytically, and convolution with the source terms yields an integral equation. This is solved iteratively to obtain the nonlinear pressure field. The convolution over the four-dimensional computational domain is performed with FFT's, and a grid with only two points per wavelength suffices due to prior filtering of the involved quantities. The present talk elaborates on the characteristic steps of the INCS method, i.e. the contrast source formulation and the filtered convolution. Comparisons with other methods will be made, and recent developments will be presented.

9:05

2aBA4. Numerical schemes for the Iterative Nonlinear Contrast Source method. Koen W.A. van Dongen, Libertario Demi, and Martin D. Verweij (Laboratory of Acoustical Wavefield Imaging, Faculty of Applied Sciences, Delft University of Technology, P.O. Box 5046, Delft 2600 GA, Netherlands, k.w.a.vandongen@tudelft.nl)

Nonlinear acoustics is gaining importance for medical acoustical imaging and high intensity focused ultrasound. With the latter one, high-amplitude acoustic wave fields are used to damage or kill cancer cells. For accurate treatment planning, a full-wave method which can model the propagation and scattering of the nonlinear field in attenuative, heterogeneous media is required. The Iterative Nonlinear Contrast Source (INCS) method is a full-wave method originally developed for homogeneous medium. It recasts the Westervelt equation into an integral equation which can be solved using a Neumann scheme. For heterogeneous media, the same approach results in additional contrast source terms. When these additional contrast sources become strong, convergence of the Neumann scheme may become an issue. To overcome this problem, the Westervelt equation may be linearized and the resulting linear integral equation may be solved with more advanced schemes such as Bi-CGSTAB. Restart strategies may be applied to eliminate systematic errors in the higher harmonics caused by the linearization. However, for realistic wave speed contrasts convergence remains problematic. To overcome these limitations, schemes such as steepest descent may be applied on the original nonlinear integral equation. In the present talk, the different schemes and their pros and cons will be discussed.

9:25

2aBA5. Medical application of nonlinear wave vector frequency domain modeling. Gregory T. Clement (Harvard Medical School, Boston, MA 02115, gclement@hms.harvard.edu) and Yun Jing (Mechanical and Aerospace Engineering, North Carolina State University, Rayleigh, NC)

While the nonlinear properties of tissues have been well documented, the computational demands required to solve nonlinear partial differential equations have generally restricted the dimensionality of numeric studies. These restrictions have been significantly reduced over time, thanks to increased memory and processing availability. Combined with more efficient computational approaches, PC-based three-dimensional modeling of nonlinear fields in tissues is now becoming feasible. Both diagnostic and therapeutic ultrasound could benefit from such accessible modeling, providing a tool for studying customized energy deposition, harmonic signal buildup, parametric methods, transducer characterization, thermometry methods, etc. We will present one such approach, which calculates diffraction through the solution to the homogenous frequency-domain Westervelt equation, while nonlinearity is calculated by the particular solution through a Green's function. The validity and efficiency of the method will be demonstrated by comparison with other well-established methods. This approach also permits backward projection of waves from an initial measurement plane toward the source, allowing a single plane to characterize an entire field, including nonlinear induction of both harmonic and sub-harmonic wave components. This ability will be shown using experimental measurements acquired with a focused source designed for HIFU.

Contributed Papers

9:45

2aBA6. Modeling acousto-optic sensing of high-intensity focused ultrasound lesion formation. Matthew T. Adams, David S. Giraud (Mechanical Engineering, Boston University, 110 Cummington St, Boston, MA 02215, adamsm2@bu.edu), Robin O. Cleveland (Institute of Biomedical Engineering, University of Oxford, Oxford, Oxfordshire, United Kingdom), and Ronald A. Roy (Mechanical Engineering, Boston University, Boston, MA)

Real-time acousto-optic (AO) sensing has been shown to non-invasively detect changes in tissue optical properties, a direct indicator of thermal damage, during high-intensity focused ultrasound (HIFU) therapy. In this work, a comprehensive model is developed to describe the AO sensing of lesion formation during HIFU therapy. The angular spectrum method is used to model ultrasound propagation, and the temperature field due to the absorption of ultrasound is modeled using a finite-difference time-domain (FDTD) solution to the Pennes bioheat equation. Thermal damage dependent optical properties are calculated based on a probabilistic and calibrated thermal dose model. To simulate light propagation inside of insonified and optically heterogeneous tissue, an open-source graphics processing unit (GPU) accelerated Monte Carlo algorithm is used. The Monte Carlo algorithm is modified to account for light-sound interactions, using input from the angular spectrum method, and to account for AO signal detection. Results will show how wavelength and illumination/detection configurations affect the detectability of HIFU lesions using AO sensing.

10:00

2aBA7. Validation of time-domain and frequency-domain calculations of acoustic propagation from breast models derived from magnetic resonance images. Andrew J. Hesford, Jason C. Tillett, Jeffrey P. Astheimer, and Robert C. Waag (Electrical and Computer Engineering, University of Rochester, UR Med Ctr Box 648, Rochester, NY 14642-8648, andrew.hesford@rochester.edu)

Magnetic resonance images with an isotropic resolution of 200 microns were collected for two human breast specimens. The images were interpolated to achieve a resolution of 50 microns and segmented to produce images of skin, fat, muscle, ductal structures, and connective tissues in consultation with a breast pathologist. The images were then mapped to acoustic parameters of sound speed, absorption and density. Calculations of acoustic propagation of fields radiated by point sources outside of the specimens were performed using the k-space finite-difference time-domain method and the frequency-domain fast multipole method. Time-domain k-space results were Fourier transformed and the 3-MHz component was compared to 3-MHz frequency-domain calculations. For the first model, measuring $1180 \times 1190 \times 290$ voxels, the two methods were found to agree in a representative coronal slice to within 5.0% (RMS). The second specimen, comprising $1350 \times 1170 \times 790$ voxels, yielded temporal and frequency-domain results that agreed to within 5.8% (RMS) in a representative coronal slice. Comparable results were obtained in other planes orthogonal to the representative slices. The close agreement establishes confidence in the accuracy of the methods when simulating propagation through large, complicated, realistic models of human tissue.

10:15–10:30 Break

Invited Papers

10:30

2aBA8. High-order numerical methods for nonlinear acoustics: A Fourier Continuation approach. Nathan Albin (Mathematics, Kansas State University, 138 Cardwell Hall, Manhattan, KS 66503, albin@math.ksu.edu)

Dispersion errors, which result from the use of low-order numerical methods in wave-propagation and transport problems, can have a devastating impact on the accuracy of acoustic simulations. These errors are especially problematic in settings containing nonlinear acoustic waves that propagate many times their fundamental wavelength. In these cases, the use of high-order numerical schemes is vital for the accurate and efficient evaluation of the acoustic field. We present a class of high-order time-domain solvers for the treatment of nonlinear acoustic propagation problems. These solvers are based on the Fourier Continuation method, which produces rapidly-converging Fourier series expansions of non-periodic functions (thereby avoiding the Gibbs phenomenon), and are capable of accurately and efficiently simulating nonlinear acoustic fields in large, complex domains.

10:50

2aBA9. Nonlinear modeling as a metrology tool to characterize high intensity focused ultrasound fields. Vera Khokhlova (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, vera@apl.washington.edu), Petr Yuldashev (Physics Faculty, Moscow State University, Moscow, Russian Federation), Wayne Kreider (Applied Physics Laboratory, University of Washington, Seattle, WA), Oleg Sapozhnikov (Physics Faculty, Moscow State University, Moscow, Russian Federation), Michael Bailey, and Lawrence Crum (Applied Physics Laboratory, University of Washington, Seattle, WA)

High intensity focused ultrasound (HIFU) is a rapidly growing medical technology with many clinical applications. The safety and efficacy of these applications require accurate characterization of ultrasound fields produced by HIFU systems. Current nonlinear numerical models based on the KZK and Westervelt wave equations have been shown to serve as quantitatively accurate tools for HIFU metrology. One of the critical parts of the modeling is to set a boundary condition at the source. In previous studies we proposed using measurements of low-amplitude fields to determine the source parameters. In this paper, two approaches of setting the boundary condition are reviewed: The acoustic holography method utilizes two-dimensional scanning of pressure amplitude and phase and numerical back-propagation to the transducer surface. An equivalent source method utilizes one-dimensional pressure measurements on the beam axis and in the focal plane. The dimensions and surface velocity of a uniformly vibrating transducer then are determined to match the one-dimensional measurements in the focal region. Nonlinear simulations are performed for increasing pressure levels at the source for both approaches. Several examples showing the accuracy and capabilities of the proposed methods are presented for typical HIFU transducers with different geometries. [Work supported by NIH EB007643.]

Contributed Papers

11:10

2aBA10. Dual-time-scale method for modeling of the nonlinear amplitude-modulated ultrasound fields. Egor Dontsov and Bojan Guzina (University of Minnesota, 500 Pillsbury Drive SE, Minneapolis, MN 55455, guzina@wave.ce.umn.edu)

This study focuses on modeling of the nonlinear acoustic wave propagation in situations when the amplitude of the focused ultrasound field is modulated by a low-frequency signal. This problem is relevant to both ultrasound imaging applications entailing the use of the acoustic radiation force, and treatment applications such as histotripsy. The difficulty of predicting the pressure wavefield lies in a fact that the excessive length of the low-frequency modulated signal may significantly increase the computational effort. To tackle the problem, this study utilizes the dual-time-scale approach, where two temporal variables are introduced to distinguish between ultrasound-scale and modulation-scale variations. In this case, the Westervelt-type equation can be effectively solved using hybrid time-frequency algorithm for any transient (sufficiently smooth) modulation envelope. To validate the proposed approach, the Khokhlov-Zabolotskaya-Kuznetsov equation was solved in the time domain for an example pressure profile on the boundary. A comparison between the time-domain and hybrid calculations demonstrates that the latter are notably faster, require significantly less memory, and have satisfactory accuracy for the ratios between the modulation and carrier ultrasound frequencies below 0.1.

11:25

2aBA11. Modeling translation of a pulsating spherical bubble between viscoelastic layers. Daniel R. Tengelsen, Todd A. Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Applied Research Laboratories, University of Texas, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

A model was developed previously that enables calculation of the translational force exerted on a pulsating bubble between parallel viscoelastic layers [Hay et al., J. Acoust. Soc. Am. **128**, 2441 (2010)]. Here the translational motion of the bubble is taken into account. Force on the bubble is calculated with a Green's function for the reverberant acoustic field in the channel. The Green's function takes into account not only elastic waves in the channel walls but also viscous boundary layers at the interfaces with the liquid. The dynamical response of the bubble is modeled by an equation of Rayleigh-Plesset form for pulsation, coupled to a momentum equation for translation. The dynamical equations are coupled to the Green's function providing the reverberant pressure field and its gradient acting on the bubble. Calculation of the time-dependent Green's function requires integration over both wavenumber and frequency space at each location along the trajectory of the bubble. Different numerical implementations were considered based on accuracy and efficiency. Simulations will be presented for several combinations of bubble radius, standoff distance, and viscous boundary layer thickness. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and NIH DK070618.]

11:40

2aBA12. Liquid compressibility effects in the dynamics of acoustically coupled bubbles. Derek C. Thomas (Dept. of Physics and Astronomy, Brigham Young University, N283 ESC, Provo, UT 84097, dthomas@byu.edu), Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Applied Research Laboratories, University of Texas at Austin, Austin, TX)

Accurate models for clusters of interacting bubbles are sought for both biomedical and underwater applications. Multiple bubble models have been developed by treating the bubbles as a system of interacting oscillators. The models are obtained initially for bubbles in an incompressible, irrotational, and inviscid liquid; additional effects are included in an *ad hoc* fashion. The existing oscillator models for the dynamics of interacting bubbles are improved by including the effect of liquid compressibility. In particular, while existing models have been improved by including propagation delays in the bubble interactions, the effect of bubble interaction on radiation damping has not been considered. The current work develops corrections for the radiation damping of coupled bubbles in both linear and nonlinear

models of bubble dynamics. These corrections eliminate certain instabilities that have been observed in delay differential equation models of coupled-bubble dynamics. Additionally, an increase in the coupling strength between bubbles undergoing high-amplitude radial motion is predicted when coupled radiation damping is included; this increase in coupling strength strongly affects the predicted motion of the system and the resultant pressure in the surrounding medium. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and NIH DK070618.]

TUESDAY MORNING, 23 OCTOBER 2012 SALON 7 ROOSEVELT, 8:15 A.M. TO 10:00 A.M.

Session 2aEA

Engineering Acoustics: Wideband Transducers and Their Impact on Total Acoustic System Design

Stephen C. Thompson, Chair
Applied Research Lab., Pennsylvania State Univ., State College, PA 16804

Chair's Introduction—8:15

Invited Papers

8:20

2aEA1. Motional current velocity control of piezoelectric loads. Robert C. Randall (Electroacoustic Research Laboratory, Advanced Technology and Manufacturing Center, Fall River, MA 02720, bobrandall81@gmail.com)

It is well known that acoustic interactions affect the transmit radiation pattern of a SONAR array, particularly when the element spacing is small relative to the acoustic wavelength. A negative feedback system with a velocity sense signal fed back to the power amplifier is one method of mitigating the array interactions, and has significant advantages for wideband use compared to either open loop compensation or passive electrical tuning. A velocity control loop flattens the transducer's frequency response, and also reduces the effects of the array interactions proportional to the loop gain. The velocity feedback signal for piezoelectric loads may be obtained by the motional current method, which is equivalent to using an ideal massless accelerometer if the transducer's electrical branch admittance is estimated correctly. The transducer's coupling coefficient, mechanical Q, and a priori estimate of the blocked capacitance fundamentally limits both the maximum stable loop gain, and the output velocity gain and phase tracking relative to the amplifier's input voltage. The array equations governing the acoustical outputs are presented, both with and without motional current velocity control.

8:40

2aEA2. Evaluating transducer bandwidth and effectiveness on overall acoustic system performance. Corey L. Bachand, David A. Brown, and Boris Aronov (BTech Acoustics LLC and UMass Dartmouth, 151 Martine Street, Fall River, MA 02723, corey.bachand@cox.net)

Piezoelectric ceramic cylindrical transducers are used extensively for underwater acoustic communication applications on mobile platforms (UUVs). Employing piezoelectric single crystals, used in either fully-active or active-passive segmented cylinders, in the transducer design has the potential to increase the usable bandwidth while reducing the overall size and weight of the device. In some instances, one single crystal transducer may replace several ceramic transducers and reduce the number of hardware channels. The impact on acoustic system design (power amplifier, matching transformer, and tuning network) for several piezoelectric ceramic and single crystal cylindrical transducer technologies is modeled and supported with measured transducer data.

9:00

2aEA3. A wideband moving coil electrodynamic transducer system for autonomous underwater vehicle-based geoacoustic inversion. Donald P. Massa (Massa Products Corporation, 280 Lincoln St., Hingham, MA 02043, Massa@massa.com) and Juan I. Arvelo (Applied Physics Laboratory, Johns Hopkins University, Laurel, MD)

A small expendable wideband low-frequency sound source that will be deployed on the seafloor is being developed to be used for geoacoustic inversion surveys in conjunction with a terrain-hugging AUV. This low-cost deployable source contains a transducer that produces a relatively flat transmit response over the broad frequency band of 100 to 4000 Hz. In operation, a seafloor interface wave will be excited and exploited for geoacoustic inversion by the deployed sound source and a receiving array on the bottom-hugging AUV. A feasibility study is also being performed that includes physics-based sonar simulations to infer the performance of geoacoustic inversion in a number of AUV scenarios and environmental conditions. Based on this study, design trade-offs will be determined to finalize key factors of the transducer, such as its physical size, weight, and production cost. Battery technology is also being developed to optimize the source level, the duty cycle, and the operating life of the signals that will be transmitted during data collection. This effort is being supported by ONR.

9:20

2aEA4. Array synthesis and wide band system. Dehua Huang (NAVSEANPT, Howell St., Newport, RI 02841, DHHuang@cox.net)

Acoustic arrays play important role as the key devices in wide band system designs. The Laguerre polynomials are successfully applied to the array synthesis first time. Conventional uniform array is the simplest design, because each element is excited at the same weight, which leads to high side lobe levels, artifacts and noise to advanced systems. Dolph utilized the first kind Chebyshev polynomials to synthesize the array beam pattern for side lobe control. However, it comes with the equal levels of the side lobes, due to the mathematical nature of the first kind Chebyshev polynomials. Taylor introduced a modified version Dolph-Chebyshev synthesis technique, which displayed tapered down side lobe levels in the region away from the main lobe. The key characteristics from the sample numerically simulated arrays by the Laguerre polynomials synthesis, i. e. radiation patterns, half-power beam width, directivity and the beam efficiency are compared with those from the synthesis of the Dolph-Chebyshev of the first kind or second kind, Taylor shading, Legendre and Hermite polynomials techniques. Work supported by the U.S. Navy.

9:40

2aEA5. High intensity air ultrasound source for determining ultrasound microphone sensitivity up to 400 kHz. Angelo J. Campanella (Acculab, Campanella Associates, 3201 Ridgewood Drive, Ohio, Hilliard, OH 43026, a.campanella@att.net)

2a TUE. AM

A broadband air jet ultrasound source, RSS101U-H, for animal bioacoustics research produces broadband ultrasound to at least 400 kHz. Free field reciprocity calibration of 1/2" condenser microphones on-axis, grid caps removed, using sine wave excitation by the method of Rudnick and Stein [JASA 20, pp 818-825, (1948)] was made (previously used to 280 kHz [JASA 67, p 7, (1980)]). Measurement from 5 kHz to 100 kHz was made in 1 kHz bins via an FFT analyzer. A communications receiver was used from 40 kHz to 400 kHz. The sensitivity of a 1/4" microphone was determined from the free field of the reciprocity 1/2" microphone source. Air jet ultrasound level at 80 mm distance was then determined with the 1/4" microphone. Communications receiver 2.5 kHz bandwidth data was reduced to 1 kHz bin values. Air humidity sound absorption was determined via ANSI 1.26. The 1/4" microphone sensitivity and broadband source sound level results in 1 kHz bands to 400 kHz are presented. Air jet spectral level was 97 dB re 20 uPa @ 75 kHz to 57 dB @ 400 kHz. This can be used to rapidly determine the sensitivity of any air ultrasound microphone over this frequency range.

TUESDAY MORNING, 23 OCTOBER 2012 ANDY KIRK A/B, 7:55 A.M. TO 12:00 NOON

Session 2aED

Education in Acoustics: Engaging and Effective Teaching Methods in Acoustics

Wendy K. Adams, Cochair
Physics, University of Northern Colorado, Greeley, CO 80631

Preston S. Wilson, Cochair
Applied Research Lab., Univ. of Texas at Austin, Austin, TX 78712-0292

Chair's Introduction—7:55

Invited Papers

8:00

2aED1. Collaborating to improve science teaching and learning through the ComPADRE digital library I. Bruce Mason (Physics & Astronomy, University of Oklahoma, Norman, OK 73019, bmason@ou.edu) and Lyle Barbato (AAPT, College Park, MD)

Most educators have found that improving their classes is best done as a collaborative process, by sharing best practices and resources with others. The ComPADRE digital library has been supporting these collaborations for the past decade through a vetted, online database of teaching and learning materials, personalization services, and tools for groups to interact. This talk will explore some examples of the resources available through the ComPADRE database that can be used to engage students in learning and can help instructors improve the outcomes of their courses. It will cover the organization of materials and how ComPADRE members can meet their personal needs. The talk will also explore examples of ComPADRE collections built by and for communities of teachers interested in specific topics or courses. Of course, examples of fun and engaging learning materials will also be demonstrated. ComPADRE is a collaboration of the American Association of Physics Teachers, the American Institute of Physics, the American Physical Society, and the Society of Physics Students and is part of the National STEM Digital Library. It is supported, in part, by funding of the National Science Foundation.

8:40

2aED2. Teaching musical acoustics with clickers. William Hartmann (Physics-Astronomy, Michigan State University, 4208 BPS Bldg., East Lansing, MI 48824, hartman2@msu.edu)

Musical acoustics is a well-proved avenue for teaching scientific concepts to students whose fields of study and interests are far removed from any science. In recent years the Michigan State course in musical acoustics has benefited greatly from using clickers. Frequent clicker questions (1 point for any answer; 2 points for the correct answer) promote attendance and help maintain a lively, interactive classroom environment, even for a large lecture class. Nobody sleeps when clicker points are on the line. Students are encouraged to discuss responses to clicker questions among themselves before answering, and the response protocol allows students to change their responses at any time before the polling is closed. Musical acoustics lectures include many demonstrations that can be presented as experiments requiring students to predict the result in advance using their clickers. "No-count" or "all-good" clicker questions can be used to determine student responses to perceptual experiments, and the feedback from the scoring algorithm gives the answer and the inevitable variability. Most important, responses to clicker questions give an instructor instant feedback about whether new lecture material has been understood. To use clickers in this way requires flexible instruction and spontaneous generation of new clicker questions.

9:00

2aED3. Providing interactive engagement in introductory acoustics through design-intensive laboratories. Andrew Morrison (Natural Science Department, Joliet Junior College, Joliet, IL 60431, amorrison@jjc.edu)

More than three decades worth of education research has shown with overwhelming evidence that the best way for students to learn is to be actively engaged in the classroom rather than passively taking in material delivered from an instructor. Although the reform of introductory classes has been widely adopted by many instructors, the implementation of reformed introductory laboratory curricula has not been as widely adopted. In our introductory acoustics course, students complete design-intensive labs where much of the instruction has been stripped away. The emphasis on student-driven experiment design and analysis is intended to provide a more scientifically authentic experience for students. The course is taught using an integrated lecture and laboratory approach. An overview of the laboratory framework and example laboratory activities used in our introductory acoustics class will be presented.

9:20

2aED4. Techniques for teaching building acoustics and noise control to university architecture students. Robert C. Coffeen (School of Architecture, Design & Planning, Universtiy of Kansas, 1465 Jayhawk Blvd, Lawrence, KS 66045, coffeen@ku.edu)

Architects are visual people. And, we cannot see sound in an architectural venue. Perhaps this has something to do with their historically poor record in dealing with acoustic and noise control issues in building spaces. Experience in teaching architecture students indicates useful teaching techniques include visits to venues with both suitable and unsuitable acoustic conditions, using modeling and auralization so that students can hear simulations of acoustical conditions produced by various interior surface shapes and architectural materials, relating their actual listening experiences in venues of various types to interior surface shapes and finish materials, and discussing the acoustical characteristics of interior materials so that a visual inspection of a space can lead to a general determination of the room acoustic conditions to be anticipated. Also discussed will be techniques for teaching architecture students the basics of architectural noise control and the basics of mechanical system noise control.

9:40

2aED5. Acoustic tweets and blogs: Using social media in an undergraduate acoustics course. Lily M. Wang (Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu)

Each fall, the author teaches an undergraduate architectural acoustics course to around 40 third-year architectural engineering students at the University of Nebraska. Beginning in 2011, a social media component was introduced to explore the use of this technology and how it may supplement the students' learning experience. Students were given an opportunity to receive extra credit by using twitter and/or blogging about course material using a set hashtag (#AE3300) or through the course website. Results were positive, and the author will discuss pros and cons that she has experienced in adding this social media component. Suggestions for future implementations and examples of student participation will be presented.

10:00–10:15 Break

10:15

2aED6. Use of pre-class quizzes to promote active learning in acoustics. Kent L. Gee and Tracianne B. Neilsen (Dept. of Physics and Astronomy, Brigham Young University, Provo, UT 84601, kentgee@physics.byu.edu)

Classroom instruction can be inefficient or ineffective when students do not come to class prepared. One strategy to engage students prior to class is the use of pre-class quizzes. One pedagogical method developed for introductory courses by physics education researchers is pre-class "just-in-time-teaching" quizzes. As a variation on that idea, pre-class learning activities have been used with great success in the general education acoustics course at Brigham Young University (BYU). However, such methods are not often applied at the advanced undergraduate and graduate levels. This paper reviews some of the findings from the introductory course efforts and then describes the implementation of pre-class quizzes for two advanced acoustics courses at BYU. Two lessons learned thus far are 1) the questions, which have a free-response format, must be carefully constructed so that the instructor can gauge student understanding, and 2) when successfully implemented, the quizzes can provide an effective framework for a class discussion of a topic, rather than a lecture with little to no participation.

10:35

2aED7. Active-learning techniques in an introductory acoustics class. Tracianne B. Neilsen and Kent L. Gee (Dept. of Physics and Astronomy, Brigham Young University, Provo, UT 84602, tbn@byu.edu)

The goal of active-learning techniques is to encourage the students to become involved with the material and take ownership for their learning, which fosters long-term knowledge and enjoyment of the subject. In this era of student-based learning outcomes, an active-learning approach is important because it focuses on what the students are doing to facilitate learning instead of what the instructor is trying to teach. To benefit most from class time, the students need to have the opportunity to actively engage with the material beforehand. If meaningful pre-class activities are required, it is easier to interact with the students during class. Some key methods for encouraging active learning during class include incorporating their pre-class experiences, conducting discussions, encouraging student participation, and evaluating student understanding with a response system, such as i-clickers. After the class time, students need apply what they have learned in answering additional questions on homework assignments and in hands-on laboratory experiences. Lessons learned after several years' worth of step-by-step efforts to approach these goals in an introductory acoustics class, which serves a wide range of majors as a general science elective, are presented.

10:55

2aED8. 25 years of distance education in acoustics. Daniel A. Russell and Victor W. Sparrow (Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, PA 16802, drussell@engr.psu.edu)

Online education is quickly becoming a popular means of delivering course content to the masses. Twenty-five years ago this Fall, the Graduate Program in Acoustics at Penn State began offering graduate level instruction in acoustics to students at a distance. In 1987 courses were offered via satellite links to Navy and industry labs, with PictureTel video conferencing to two centralized locations added in 1992. By 1994 videotapes allowed for broader content distribution to students at more varied locations. In 2002 videostreaming lectures over the internet expanded the delivery even further. Currently courses are taught to a blended audience of both resident and distance students, with lectures being live streamed over the internet and archived digitally for offline access. This talk will briefly summarize the history and development of online graduate education in acoustics at Penn State. We will discuss the use of technology both as a tool for delivering course content as well as the impact that technology has on the quality and means of instruction and the interaction between teacher and students. The necessary adaptation of teaching styles and the adjustments required to meet the varied needs of a blended student audience will also discussed.

Contributed Papers

11:15

2aED9. Problem solving assessment. Wendy K. Adams (Physics, University of Northern Colorado, CB 127, Greeley, CO 80631, wendy.adams@colorado.edu)

Although educators and employers highly value problem solving and have put extensive effort into understanding successful problem solving, there is currently no efficient way to evaluate it. Science educators regularly make use of concept inventories and perceptions surveys (aka: attitudes and beliefs) to evaluate instruction. However, these only touch on a fraction of what is learned in a course. Students apply a range of processes, expectations and bits of knowledge when solving a physics problem and some of these are impacted by the course. The question is how can we identify what these processes, expectations and knowledge are, how can we teach them and then how can we measure them? While developing the CAPS (Colorado Assessment of Problem Solving), I identified 44 processes, expectations and bits of knowledge used to solve an in depth real world problem. In this presentation CAPS and some of what was learned during the development will be presented.

11:30

2aED10. Teaching graduate level acoustics courses to a blended enrollment of resident and distance education students. Daniel A. Russell (Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, PA 16802, drussell@engr.psu.edu)

The Graduate Program in Acoustics offers courses leading to the M.Eng. in Acoustics online through Penn State's World Campus. The current method of course content delivery is to live-stream (with digital video archive) lectures to a blended student audience consisting of about 10-15 resident students physically present in the classroom and another 15-20 students at a distance who may be watching the class live, or who may be viewing the archived recording afterward. This paper will explore several issues involving the engagement of this blended audience of students. How does one encourage and enable students with a broad range of backgrounds, interests, and physical locations to engage with the topic material? How does one foster collaboration and interaction between distance students and the teacher and between resident and distance students? How does one manage office hours, help sessions, group projects, experiments, and student presentations for a blended student audience? Current practice and personal experiences from our faculty will be shared, and ideas from the audience will be welcomed.

11:45

2aED11. Real-time audio signal capture and processing using MATLAB object oriented programming. Samarth Hosakere Shivaswamy, Xiang Zhou, Stephen Roessner, Gang Ren (Dept. of Electrical and Computer Engineering, Univ. of Rochester, Rochester, NY 14627, g.ren@rochester.edu), Dave Headlam, and Mark Bocko (Dept. of Electrical and Computer Engineering; Dept. of Music Theory, Eastman Sch. of Music, Univ. of Rochester, Rochester, NY)

In MATLAB programming language the real-time audio processing functions are usually simulated in non-real-time due to a lack of real-time audio programming support. As a result the real-time audio signal capture and processing functionalities are usually implemented in other programming languages and cannot utilize the extensive signal processing functionalities provided by MATLAB. In this paper we introduce a MATLAB real-time signal processing framework based on MATLAB timer object and audiorecorder object. The proposed processing framework serves as an alternative solution for real-time programming implementation and demonstration. In our proposed processing framework the timer object is implemented to handle the looping of processing cycle, schedule the signal processing tasks, and handle the error processing. The audio capturing/processing functionality is implemented in the timer cycle by using two audiorecorder objects that read the audio streaming data and feed a segment of data to signal processing alternatively. The proposed framework achieves satisfactory real-time performance with no missing audio frames when a short audio delay setting of 10ms is applied. Several application examples of our proposed framework are also demonstrated.

Session 2aNS

Noise and Architectural Acoustics: Sound Quality, Sound Design, and Soundscape

Brigitte Schulte-Fortkamp, Cochair
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Klaus Genuit, Cochair
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Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Sq., Vernon, CT 06066

Chair's Introduction—7:45

Invited Papers

7:50

2aNS1. Approaching environmental resources through soundscape. Brigitte Schulte-Fortkamp (TU Berlin, Einsteinufer 25 TA 7, Berlin 10587, Germany, bschulte_f@web.de)

The Soundscape concept is introduced as a scope to rethink the evaluation of "noise" and its effects and to focus on a diverse field of experts and expertise in order to fulfill the requirements for a "good environment" or a "sensitive environment" with respect to quality of life. Moreover, Soundscape is defined as an environment of sound with emphasis on the way it is perceived and understood by the individual, or by a society. Therefore it is suggested to explore noise in its complexity and its ambivalence and its approach towards sound to consider the conditions and purposes of its production, perception, and evaluation, to understand evaluation of noise/ sound as a holistic approach. Qualitative methods referring to a heterogeneous 'field of research' and among them are different forms of observation, interviewing techniques and the collection of documents or archival data as well as binaural measurements will be presented and proven regarding their effects of explanations. The intention of scientific research here is to learn about the meaning of the noise with respect to people's living situation and to implement the adequate procedure to open the "black box" of people's mind regarding their needs for a supportive environment.

8:10

2aNS2. Relationship between environmental noise, sound quality, soundscape. Klaus Genuit, André Fiebig (HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de), and Brigitte Schulte-Fortkamp (Technical University Berlin, Berlin, Germany)

The term "Environmental Noise" is well known for many years. Its characteristic is often described by parameters like A-weighted SPL, Lden, Lday, Levening, Lnight. These parameters can be measured and calculated. In the field of "Sound Quality" psychoacoustic parameters are additionally used like loudness, sharpness, roughness and others, which can be measured but not calculated for a complex sound field. On an international level a standard is available only for the loudness of stationary sounds so far. The "relatively" young term "soundscape" will be standardized in ISO 12913-1. Moreover, as it considers human perception including cognitive aspects, context and interaction it goes beyond physics and psychoacoustics. It involves a concept, where environmental noise is not reduced to an averaged quantity evoking only unpleasantness feelings estimated by statistical probabilities, but understanding noise as a valuable resource, which can be purposefully utilized. In spite of recent progresses in the standardization process lots of misinterpretations occur in practical use, where the terms are heavily mixed up. Environmental noise and soundscape are no synonyms, for example low noise level does not directly mean a good sound quality. The paper will clarify options and limitation of both terms.

8:30

2aNS3. Soundscape and sound quality—Similar and powerful design techniques. Bennett M. Brooks (Brooks Acoustics Corporation, 30 Lafayette Square - Suite 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com)

Two powerful analysis techniques available to acoustical researchers and designers include the sound quality method and the very similar soundscape method. In each of these techniques physical acoustical measurements are combined with in-depth interviews and opinion juries to determine the cause and effect relationship that a particular sound, or set of sounds, has on a population. The sound quality technique has been in use for many years, and focuses on product development. An example is the sound of an automobile door closing - is the car door closing sound perceived as "solid and expensive" or "cheap and tinny". Another example is a vacuum cleaner - does it sound "powerful and effective" or "weak and ineffective"? The soundscape technique focuses on environmental sound, often in public spaces like a park or in a residential neighborhood. For example, is a certain transportation vehicle sound or outdoor entertainment facility sound acceptable or unacceptable to the wider community? This paper will explore the similarities between these two related fields and the opportunities they offer to sound designers.

8:50

2aNS4. The sound-absorbing city—New ideas for living environments around airports. Juergen Bauer (Department of Architecture, Waterford Institute of Technology, Granary, Hanover St, Co Waterford, Ireland, jbauer@wit.ie)

Great efforts and progress have been made in terms of noise protection measures in both urban, suburban and rural environments. Local or regional urban planning guidelines and anti-noise-manuals provide experienced and practical advice to reduce noise, in order to provide a better quality of life. Most of the anticipated solutions such as noise-protection-walls, fences, planted mounds etc. will address issues caused by land traffic. However, due to their nature, they fail to respond to "airborne" noise immission. In addition, there is a common public misconception that sound should be interpreted as noise ie. as a waste to get rid of, instead of critically identifying sound and sound clusters as a potential and as a resource to be integrated. The concept of the sound-absorbing city applies the same principles of sound reflection and sound absorption, as applied to an architectural space, in an urban space. It also investigates the possibility of combining unwanted sound, such as air traffic noise with wanted sound, such as nature and community sound. The paper discusses the concept of the sound-absorbing city, its potentials and its apparent limits, with regard to new settlements and existing agglomerations around airports.

9:10

2aNS5. I hear what you mean—Source intensity versus receiver level. Alex Case (Sound Recording Technology, University of Massachusetts Lowell, Lowell, MA 01854, alex@fermata.biz)

A receiver's assessment of the quality of any single element of a soundscape is not limited to the objective values of sound attributes at the receiver location, but includes an intuitive or instinctive compensation for the source-to-receiver path. Borrowing from the time-proven connection between performance intensity versus presented level in sound recordings, emotion and meaning are found to include what a listener infers about the local environment and context at the sound source, discounting whatever may have happened on its way to the listener.

9:30

2aNS6. Relevance and applicability of the soundscape concept to physiological or behavioral effects caused by noise at very low frequencies which may not be audible. Wade Bray (HEAD acoustics, Inc., 6964 Kensington Road, Brighton, MI 48116, wbray@head-acoustics.com)

A central tenet of the Soundscape concept is that humans immersed in sonic environments are objective measuring instruments (New Experts), whose reports and descriptions must be taken seriously and quantified by technical measurements. A topic category in acoustics meetings of recent years is "Perception and Effects of Noise." There is growing evidence from the field, and from medical research, that the ear's two-part transducer activity involving inner hair cells (IHC, hearing, velocity-sensitive) and outer hair cells (OHC, displacement-sensitive) may, through demonstrated OHC activation and neural signals at up to 40 dB below the audibility threshold, produce behavioral and physiological effects as reported by a growing number of people. The Soundscape concept centering on human responses, New Experts, is as important and applicable to responses to effects from sound as it is to responses to directly audible sound. In a wider sense, this is a new sound quality and psychoacoustic issue.

9:50

2aNS7. Soundwalks in urban areas—Triangulation of perceptive and acoustical data. Kay S. Voigt and Brigitte Schulte-Fortkamp (Institute of Fluid Mechanics and Engineering Acoustic, Technische Universitaet Berlin, Einsteinufer 25, Berlin 10587, Germany, kay.s.voigt@gmx.de)

Current work takes a comparative view on the analysis of several soundwalks in urban areas, investigating the benefit of additional effort in enhancing the attendees' description-level for data triangulation. Soundwalk is a tool in the soundscape approach for an all-embracing analysis of the unique sonic environment. The triangulation of data has to combine acoustical measurements of the same procedures and perceptive appraisals differing in its quality of description. The focus of each research varies from perception of places to questions of the overall feeling of safety at the different locations. Local, acoustical and safety experts are involved. The qualitative analysis considers several variables like profession of soundwalkers, knowledge about places, kind of places, chosen or given locations, and the used native language of questioning. Different levels of description narrated by the participators will be identified, as well as its possible emphasis by discourse on attendees' scaled ratings and written notes. This analysis progress contributes to an appropriate assignment of subjective descriptions to values of psychoacoustical parameters and the elucidation of predominant aspects in the soundscape. Furthermore the soundwalk's contrasting capacity with regards to the content of previous interviews detected multiple layers on the issue of safety in municipal locations.

10:10–10:25 Break

10:25

2aNS8. Sound preference prediction in a design stage—A case study in the Shenzhen Dongmen Open Space. Lei Yu (HIT Shenzhen Graduate School, E425 HIT Campus, Shenzhen University Town, Xi Li, Nan Shan, Shenzhen 518055, China, Leilayu@hitsz.edu.cn)

Attention on visual effects is insufficient in urban open spaces, while soundscape is complementation and sound design is crucial. In this paper, sound effects in the Shenzhen Dongmen Open Space have been studied. It shows that exist sounds are not delightful to satisfy acoustic comfort but to cause annoy perceptions. Therefore, this study is focused on examining various sound effects in the Shenzhen Dongmen Open Space concerning on sound preference evaluations. Based on various sounds influencing the subjective preference evaluations, Artificial Neural Network (ANN) models have been developed to predict how delightful of the sonic environment in terms of different sound design schemes to the space. Furthermore, the sound preference predictions of ANN models' output will be compared with the preference evaluations from lab experiments, and then validated by the lab results.

10:45

2aNS9. Soundscape analysis of two parks in Berlin. Natalia Manrique-Ortiz and Brigitte Schulte-Fortkamp (Technische Universität Berlin, Einsteinufer 25, Berlin 10587, Germany, natalia.manrique.ortiz@gmail.com)

Nowadays the protection of quiet areas is an issue of increasing importance. This importance is reflected in the European directives and policy intentions of many countries around the world. In order to protect this areas, is important to characterize their soundscapes and analyze the areas, paying special attention to the geography, aesthetics, social, psychological and cultural aspects, since these aspects play a significant role in the noise perception. The aim of this research is to analyze two of the most important Berliner parks. Victoria-Park and Schlosspark are located in very different areas in Berlin. Schlosspark is in a more quiet neighborhood with families and Victoria-Park is in a lively, young and multicultural neighborhood. The protection of these parks begins understanding the community who make use of them. The research will be based on interviews and soundwalks according to the soundscape approach. The results will be presented.

11:00

2aNS10. Sound and noise in urban parks. Antonio P. Carvalho (Laboratory of Acoustics, University of Porto, FEUP - Fac. Eng. Univ. Porto, DEC (NIF501413197), Porto P-4200-465, Portugal, carvalho@fe.up.pt) and Ricardo C. Dias (Laboratory of Acoustics, University of Porto, Porto, Portugal)

The main goal of this work is to study the soundscape of urban gardens and parks using a sample of ten sites in Porto, Portugal to characterize their noise levels through the acoustic characterization of the park's exterior and interior noise levels (LAeq, LA10, LA50 and LA90) and by a socio-acoustic survey to the visitors to check their perception of acoustic quality. The measurements showed gardens/parks with interior noise levels from 47 to 61 dBA (with maximum exterior noise levels up to 67 dBA). The difference between exterior and interior LAeq was between 3 and 19 dBA. The gardens with lower noise levels are the larger and out of downtown. An "acoustic" classification for gardens/urban parks is proposed regarding their noise "isolation" capacity and acoustic ambience. Measurements done in 1990 allow for the comparison of the evolution in the last 21 years. The socio-acoustic survey concludes that Porto's city parks are visited mostly by an elderly male population that regards these places as sites of gathering and to practice some physical activity rather than as an acoustic retreat. The population seems accustomed to the dominant noise, classifying these spaces as pleasant and quiet, even when noise is over acceptable limits.

11:15

2aNS11. Investigation of tranquility in urban religious places. Inhwan Hwang, Jooyoung Hong, and Jin Yong Jeon (Architectural Engineering, Hanyang University, Seoul, Seongdong-gu 133791, Republic of Korea, jyjeon@hanyang.ac.kr)

In the present study, tranquility in urban religious places including a cathedral and a Buddhist temple has been assessed by soundwalks. Both Myung-dong Cathedral and Bongeun Temple located in the center of Seoul were selected as measurement sites. During the soundwalks, audio-visual recordings were conducted at selected positions. From the field measurements, the temporal and frequency characteristics of the sound environment in two religious places were explored. Participants evaluated their perceived soundscape using a soundwalk questionnaire along the soundwalk routes in the church and temple gardens in order to investigate the value of tranquility as urban stress relievers. From the results, indicators representing tranquility difference in particular soundscapes were examined.

11:30

2aNS12. Psychoacoustic assessment of a new aircraft engine fan noise synthesis method. Selen Okcu (National Institute of Aerospace, Hampton, VA 23666, selen.okcu@nasa.gov), Matthew P. Allen (Department of Mechanical Engineering, Virginia Tech, Blacksburg, VA), and Stephen A. Rizzi (Structural Acoustics Branch, NASA Langley Research Center, Hampton, VA)

Simulation of aircraft flyover events can facilitate psychoacoustic studies exploring the effects of noise generated by future aircraft designs. The perceived realism of a simulated flyover event may be impacted by the perceived realism of the synthesized fan noise of the aircraft engine. Short-term fluctuations in tonal amplitude and frequency are important cues contributing to that perception of realism, but are not accounted for by predictions based on long-term averages. A new synthesis method has been developed at NASA Langley Research Center to generate realistic aircraft engine fan noise using predicted source noise directivities in combination with short-term fluctuations. In the new method, fluctuations in amplitude and frequency are included based upon analysis of static engine test data. Through psychoacoustic testing, this study assessed perceived effectiveness of the new synthesis method in generating realistic fan noise source. Realism was indirectly assessed by judging the similarity of synthesized sounds (with and without fluctuations) with recordings of fan noise. Results of ANOVA analyses indicated that subjects judged synthesized fan noise with fluctuations as being more similar to recordings than synthesized fan noise without fluctuations.

11:45

2aNS13. A geospatial model of ambient sound pressure levels in the continental United States. Dan Mennitt, Kurt M. Fristrup (Natural Sounds and Night Skies Division, National Park Service, Fort Collins, CO 80525, daniel_mennitt@partner.nps.gov), and Kirk Sherrill (Inventory and Monitoring, National Park Service, Fort Collins, CO)

There has been much effort in the US and worldwide to measure, understand and manage natural soundscapes which are often complex due to a multitude of biological, geophysical, and anthropogenic influences. The sound pressure level is a time and space varying quantity that represents the aggregate of present sources. This work presents a predictive model relating seasonal sound pressure levels to geospatial features such as topography, climate, hydrology and anthropogenic activity. The model utilizes random forest, a tree based machine learning algorithm, which does not explicitly incorporate any apriori knowledge of acoustic propagation mechanics. The response data encompasses 271,979 hours of acoustical measurements from 192 unique sites located in National Parks across the contiguous United States. Cross validation procedures were used to evaluate model performance and identify GIS explanatory variables with predictive power. Using the model, the effect of individual explanatory variables on sound pressure level can be isolated and quantified revealing trends across environmental gradients. An example application of projecting predicted sound pressure levels across the Olympic peninsula is discussed. Because many wildlife habitats, geological processes, and anthropogenic impacts occur on a regional scale, the extent of acoustical analyses must be on similar scales.

Session 2aPA

Physical Acoustics: Waves in Heterogeneous Solids I

Joseph A. Turner, Cochair
Dept. of Mechanical and Materials Engineering, Univ. of Nebraska-Lincoln, Lincoln, NE 68588-0526

Goutam Ghoshal, Cochair
Electrical and Computer Engineering, University of Illinois at Urbana-Champaign, Urbana, IL 61801

Chair's Introduction—7:55

Invited Papers

8:00

2aPA1. Ultrasound therapy delivery and monitoring through intact skull. Kullervo Hynynen (Medical Biophysics, University of Toronto, Sunnybrook Health Sciences Centre, Toronto, ON M4N 3M5, Canada, khynynen@sri.utoronto.ca), Yuexi Huang, Meaghan O'Reilly (Physic Sciences Platform, Sunnybrook Research Institute, Toronto, ON, Canada), Ryan Jones, Dan Pajek (Medical Biophysics, University of Toronto, Toronto, ON, Canada), and Aki Pulkkinen (Physic Sciences Platform, Sunnybrook Research Institute, Toronto, ON, Canada)

Magnetic Resonance imaging guided and monitored focused ultrasound is now tested for deep focal thermal ablation of brain tissue in clinical setting. The major barrier for this treatment is the propagation of ultrasound through an intact skull that strongly attenuates and scatters the ultrasound wave. The distortion can be corrected by using CT-derived bone density information and computer simulations to derive phase and amplitude information such that the driving signals can be adjusted to reduce the distortions. In this paper the current results on ultrasound propagation through skull will be discussed and the clinical applications of noninvasive ultrasound treatments will be reviewed. The advance in online acoustic monitoring of cavitation based treatments methods will also be shown

8:20

2aPA2. Modeling of transcranial ultrasound for therapeutic and diagnostic applications. Gregory T. Clement (Harvard Medical School, Boston, MA 02115, gclement@hms.harvard.edu)

Ultrasound's use in the brain has conventionally been limited by its inability to penetrate the skull. To overcome these limits, we have been investigating techniques to maximize energy transfer and minimize distortion through the skull bone. These model-based aberration correction approaches - now in the early stages of clinical testing - rely on both practical and accurate numeric methods. Efforts to improve these methods necessitate an increasingly detailed consideration of skull heterogeneity. To facilitate this numerically-intensive problem, we are utilizing an inhomogeneous pressure simulation code, based on a pseudo-spectral solution of the linearized wave equation. Forward and scattered waves are determined over a pre-specified volume with scattering determined by the impedance mismatch between a given voxel and regional points in the projection plane. The total forward-scattered pressure is recorded over the relevant k-space, while reflected energy is processed in a separate backward projection. This process is repeated iteratively along the forward projection plane until the volume of interest has been traversed. This procedure can be repeated an arbitrary number of times N, representing N-1 order scattering. Abilities and limitations of the method will be demonstrated by comparison with FDTD simulation.

8:40

2aPA3. Validation of a finite-difference acoustic propagation model of transcranial ultrasound. Guillaume Bouchoux, Kenneth B. Bader (Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, bouchoge@ucmail.uc.edu), Joseph J. Korfhagen (Neuroscience Graduate Program, University of Cincinnati, Cincinnati, OH), Jason L. Raymond (Biomedical Engineering Program, University of Cincinnati, Cincinnati, OH), Shivashankar Ravishankar, Todd A. Abruzzo (Radiology, University of Cincinnati, Cincinnati, OH), and Christy K. Holland (Internal Medicine, University of Cincinnati, Cincinnati, OH)

Adjuvant ultrasound exposure improves rtPA thrombolysis in stroke patients. Transmission of 120-kHz ultrasound through the temporal bone is efficient but exhibits skull-dependent distortion and reflection. Numerical models of acoustic propagation through human skull based on computed tomography (CT) data have been developed. The objective of our study was to validate a finite-difference model of transcranial ultrasound quantitatively. The acoustic fields from a two-element annular array (120 kHz and 60 kHz) were acquired in four ex-vivo human skulls with a calibrated hydrophone (10 kHz-800 kHz frequency range). The spatial distributions of the acoustomechanical properties of each skull were obtained from CT scans and used for simulations. Predicted acoustic fields and waveform shapes were compared with corresponding hydrophone measurements and were in good agreement. Transmitted wave amplitudes were systematically underestimated (14%) and reflected wave amplitudes were overestimated (30%). The acoustic impedance of each skull was likely underestimated from the CT scans. However, high correlation between predictions and measurements (R<+>2<+>=0.93 and R<+>2<+>=0.88 for transmitted and reflected wave amplitudes, respectively) demonstrates that this model can be used quantitatively for evaluation of 120-kHz ultrasound-enhanced thrombolysis. This work was supported by NIH-RO1-NS047603.

9:00

2aPA4. Ultrasound assessment of bone with ultrasound: Present and future. Pascal Laugier (Laboratoire d'Imagerie Parametrique, CNRS/University Pierre et Marie Curie, 15 rue de l'ecole de medecine, Paris 75017, France, pascal.laugier@upmc.fr)

Bone is a composite, porous and anisotropic material whose complex hierarchical structure extends over several levels of organization from the nanoscale to the macroscopic scale. One of the striking features of this tissue is its ability to adapt to variable loading conditions. This results in spatially, temporally and directionally variable elastic properties leading to a perfect adaptation to locally varying functional demands. Elastic properties of bone are nowadays widely used in fundamental studies, in conjunction with numerical models, to investigate the structure-function relationships and in clinical applications to predict fracture risk or to monitor fracture healing. However, the problem of multiscale assessment of bone elastic properties, spanning the full range of applications from in vitro to in vivo applications, remains a challenge. Novel emerging quantitative ultrasound technologies, taking benefit of the scalability of ultrasound, have emerged to noninvasively investigate elastic properties at multiple organization level. These include scanning acoustic microscopy, ultrasonic resonant spectroscopy and guided waves propagation. These techniques will be presented to show how they can help in characterizing the anisotropic stiffness tensor in vitro or determine bone properties in vivo. Relationships of quantitative ultrasound variables with structural and elastic alterations will be illustrated through multiple examples.

Contributed Papers

9:20

2aPA5. Plumbing the depths of Ligeia: Considerations for acoustic depth sounding in Titan's hydrocarbon seas. Juan I. Arvelo and Ralph Lorenz (Applied Physics Laboratory, Johns Hopkins University, 11100 Johns Hopkins Rd., Laurel, MD 20723, juan.arvelo@jhuapl.edu)

Saturn's moon Titan is the only satellite in our solar system with a dense atmosphere and hydrocarbon seas. The proposed Titan Mare Explorer (TiME) mission would splashdown a capsule to float for 3 months on Ligeia Mare, a several-hundred-kilometer wide sea near Titan's north pole. Among TiME's scientific goals is the determination of the depth of Ligeia, to be achieved with an acoustic depth-sounder. Since Titan's surface temperature is known to vary around 92 K, all instruments must be ruggedized to operate at cryogenic temperatures. This paper's contributions include an approach to infer key acoustic properties of this remote environment, their influence on the development of a cryogenic depth sounder, and on an approach to infer the transducer's response, sensitivity and performance when unable to perform in-situ calibration measurements or to replicate key environmental conditions. This effort was conducted under the auspices of the Civilian Space Independent Research and Development program from the Johns Hopkins University Applied Physics Laboratory.

9:35

2aPA6. Acoustophoresis in gases: Effect of turbulence and geometrical parameters on separation efficiency. Etienne Robert, Ramin Imani Jajarmi (Mechanics, Kungliga Tekniska Högskolan (KTH), Osquars Backe 18, Stockholm 100 44, Sweden, etienne@mech.kth.se), Markus Steibel, and Klas Engvall (Chemical Technology, Kungliga Tekniska Högskolan (KTH), Stockholm, Stockholm, Sweden)

Advanced particle manipulation techniques based on acoustophoresis have been developed in recent years, driven by biomedical applications in liquid phase microfluidics systems. The same underlying physical phenomena are also encountered in gases and hold great potential for novel particle separation and sorting techniques aimed at industrial and scientific applications. However, considering the physical properties of gases, optimizing the performance of flow-through separators unavoidably requires an understanding of the re-mixing effect of turbulence. In the work presented here we have investigated the effect of turbulence intensity on the separation efficiency of a variable frequency acoustic particle separator featuring a rectangular cross-section with adjustable height. This allows the creation of a standing wave with a variable frequency and number of nodes. The air flow is seeded with alumina particles, 300 nm nominal diameter, and the excitation source is an electrostatic transducer operated in the 50-100 kHz range. In addition to flow and acoustic parameters, the separation efficiency is investigation as a function of geometric parameters such as the parallelism of the resonator walls and the matching between the channel height and the excitation frequency. The measurements made using laser doppler anemometry and light scattering provide guidance for the design of separator configurations capable of advanced separation and sorting tasks with submicron particles.

9:50

2aPA7. Ultrasonic measurements of clays and silts suspended in water. Wayne O. Carpenter (National Center for Physical Acoustics, University of Mississippi, 1 Coliseum Drive, University, MS 38677, wocarpen@olemiss.edu), Daniel G. Wren, Roger A. Kuhnle (Agriculture Research Service - National Sedimentation Laboratory, U.S. Department of Agriculture, Oxford, MS), James P. Chambers, and Bradley T. Goodwiller (National Center for Physical Acoustics, University of Mississippi, University, MS)

Ongoing work at the National Center for Physical Acoustics is aimed at using acoustics to provide monitoring for fine sediments suspended in water. The ultimate goal of the work is to field an acoustic instrument that can monitor fine particle concentration in rivers and streams. Such an instrument would have several advantages over currently available technologies. Expanding upon work from Carpenter et. al (2009), two immersion transducers were placed at a fixed distance to measure attenuation and backscatter from acoustic signals at 10 MHz and 20 MHz propagated through clays (bentonite, illite, and kaolinite) and silt. The resulting data set encompasses a wide range of concentrations (0.01 - 14 g/L) and particle sizes (0.1 - 64 micron diameter particles). Backscatter and attenuation curves for each material across the range of concentrations will be shown and compared to the theoretical attenuation curves developed by Urick (1948). This work has produced a data set for model development using a combination of backscatter and attenuation to allow for single-frequency discrimination between clay and silt particles suspended in water.

10:05–10:20 Break

10:20

2aPA8. Acoustic wave propagation in a channel bifurcated by an elastic partition. Katherine Aho and Charles Thompson (University of Massachusetts Lowell, 1 University Ave, Lowell, MA 01854, katherine_aho@student.uml.edu)

Linear wave propagation in a narrow channel that is axially partitioned by a flexible membrane is examined. The enclosed fluid is excited by the time harmonic displacement in the channel cross-section. The axial variation in the acoustic impedance of the partition gives rise to the generation of evanescent modes in the channel. The effect of these evanescent modes on the vibration of the membrane is of particular interest. It is shown that in the limit of high channel aspect ratio one can model these modes by an effective source distribution along the surface of the membrane. The asymptotic analysis of the source distribution is presented. (NSF Grant 0841392)

10:35

2aPA9. Selected theoretical and numerical aspects of fast volume and surface integral equation solvers for simulation of elasto-acoustic waves in complex inhomogeneous media. Elizabeth Bleszynski, Marek Bleszynski, and Thomas Jaroszewicz (Monopole Research, 739 Calle Sequoia, Thousand Oaks, CA 91360, marek@monopoleresearch.com)

Comparative analysis is considered of two fast FFT-matrix-compression based elasto-acoustic integral equation solvers, employing volumetric and surface formulations, and designed to analyze sound propagation inside a human head; in particular to examine mechanisms of energy transfer to the inner ear through airborne as well as non-airborn path-ways, and to assess effectiveness of noise-protection devices. Verification tests involving the fast surface and volume integral equation solvers are carried out comparing their predictions with those following from an analytical solution of field distribution in an elasto-acoustic layered sphere.Resultsiare presented of representative numerical simulations of acoustic energy transfer to the cochlea for a human head model containing a detailed geometry representation of the outer, middle, and inner ear. The geometry model consists of: (1) the outer surface of the skin surrounding the skull and containing (2) the outer ear represented by its exterior surface, the surface of the auditory canal, and the tympanic membrane modeled as a finite-thickness surface, (3) the middle ear consisting of the system of ossicles and supporting structures, (4) the skull described by its external surfaces and including (5) a set of surfaces representing the inner ear (boundaries of the cochlea, the vestibule, and the semi-circular canals). *This work is supported by a grant from US Air Force Office of Scientific Research.

10:50

2aPA10. A modeling and simulation suite for design of buried object scanning sonars. Hunki Lee, Eunghwy Noh (Mechanical Engineering, Yonsei University, Seoul, Republic of Korea), Kyounghun Been, Hongmin Ahn, Wonkyu Moon (Mechanical Engineering, Pohang University of Science and Technology, Pohang, Republic of Korea), and Won-Suk Ohm (Mechanical Engineering, Yonsei University, 50 Yonsei-ro, Seodaemun-gu, Seoul, Seoul 120-749, Republic of Korea, ohm@yonsei.ac.kr)

In this talk we highlight a work in progress, concerning the development of a comprehensive modeling and simulation (M&S) suite for design of buried object scanning sonars. The M&S suite is expected to cover almost all aspects of physical and engineering acoustics involved in the design process, ranging from transducers, sound propagation, sediment acoustics, backscattering by buried objects, to sonar image processing. The overview of the M&S suite is given along with a preliminary demonstration in the context of a cylindrical object buried in sandy sediment. [This work was conducted in the Unmanned Technology Research Center (UTRC) sponsored by the Defense Acquisition Program Administration (DAPA) and the Agency for Defense Development (ADD) in the Republic of Korea.]

11:05

2aPA11. Multi-frequency modes in dispersive media. Craig N. Dolder and Preston S. Wilson (Department of Mechanical Engineering & Applied Research Laboratories, University of Texas, 10000 Burnet Road, Austin, TX 78758, dolder@utexas.edu)

A common phenomenon in acoustics is the existence of multiple eigenfunctions (mode shapes) corresponding to the same eigenvalue (frequency), which is known as degeneracy. In highly dispersive media the opposite can occur, whereby a single eigenfunction corresponds to multiple eigenvalues. Several ways to visualize the source of, and interpret the physical meaning of, this phenomenon are presented. Instances of this phenomenon occurring in analytical models and experiments are used as examples.

11:20

2aPA12. Estimating the acoustic impedance of the ground using signals recorded by a 3D microphone array. W. C. Kirkpatrick Alberts (RDRL-SES-P, US Army Reserach Laboratory, 2800 Powder Mill, Adelphi, MD 20783, kirkalberts@verizon.net)

In applications where there is a need to accurately classify impulsive acoustic events, the impedance of the ground can significantly alter the reflected wave such that the superimposed direct and reflected waves could lead to erroneous classifications. If the phase and amplitude changes attributable to impedance ground are considered, knowledge of the ground impedance near every array site can mitigate classification errors caused by impedance ground. Under controlled experimental conditions, the ground impedance in the vicinity of an array can be deduced using standard methods. However, when an array is fielded in an uncontrolled environment, alternative ground impedance estimation techniques must be explored. Soh et al. [J. Acoust. Soc. Am, 128(5), 2010] demonstrated that the impedance of the ground could be directly determined using a pair of vertically spaced microphones and an impulsive source 400 m from the microphones. With some increase in complexity, this direct method of determining ground impedance can be applied to recordings from typical three dimensional acoustic arrays. Estimates of the acoustic ground impedance obtained directly from recordings by microphones distributed in a 1 m radius tetrahedral array will be discussed.

11:35

2aPA13. Evaluating duplex microstructures in polycrystalline steel with diffuse ultrasonic backscatter. Hualong Du and Joseph A. Turner (University of Nebraska-Lincoln, Nebraska Hall, Lincoln, NE 68588, hualong.du@huskers.unl.edu)

The performance of metallic components is governed in large part by the microstructure of the base material from which the component is manufactured. In this presentation, diffuse ultrasonic backscatter techniques are discussed with respect to their use for monitoring the microstructure of polycrystalline steel as a result of the manufacturing process. To improve the mechanical properties, the surface of polycrystalline steel is quenched, a process which transforms the initial phase to a pearlite phase within grains. A diffuse ultrasonic backscatter model is developed that includes the duplex microstructure through the addition of an additional length scale in the two-point spatial correlation function. This function defines the probability that two randomly chosen points will fall into the same grain and/or same crystallite. The model clearly shows the dependence of the diffuse ultrasonic backscatter signal with respect to frequency, average grain size and lamellar spacing of the crystallites. Experimental results are used to show how the two length scales can be extracted from the measurements. The spatial variation of the microstructure with respect to depth from the quench surface is also examined. These diffuse ultrasonic techniques are shown to have the sensitivity to deduce the duplex microstructure throughout the sample.

11:50

2aPA14. Elastic properties of coarse grained lead-free solder alloys. Josh R. Gladden and Sumudu Tennakoon (Physics & NCPA, University of Mississippi, University, MS 38677, jgladden@olemiss.edu)

Because of health and environmental concerns about lead, lead-free solder alloys in most consumer electronics have been required in the European Union since 2006. Many of these alloys are prone to mechanical failure over time, leading to less reliable circuitry. The source of these failures is not well known and many have conjectured that the coarse grained alloys become more brittle over time when exposed to elevated temperatures (~100 °C). Our group, in collaboration with Cisco Systems, has recently studied the effects of aging on the mechanical properties of Sn-Ag-Cu (SAC) solder alloys using using both resonant ultrasound spectroscopy (RUS) and conventional pulse-echo methods. With grain sizes on the order of 100's of microns, the heterogeneity of these alloys present a particular problem for RUS and interpretation of pulse-echo data. Resonance data exhibiting the effect of the heterogeneity will be presented and discussed. Elastic moduli derived from pulse-echo methods as a function of temperature and isothermal aging time will also be shown.

Session 2aSAa

Structural Acoustics and Vibration: Session in Honor of Preston W. Smith, Jr.

Allan D. Pierce, Cochair
P.O. Box 339, East Sandwich, MA 02537

J. Gregory McDaniel, Cochair
Mechanical Engineering, Boston Univ., Boston, MA 02215

Chair's Introduction—8:30

Invited Papers

8:35

2aSAa1. Preston Smith and waves in a cylindrical shell. James G. McDaniel (Mechanical Engineering, Boston University, 110 Cummington Street, Boston, MA 02215, jgm@bu.edu)

In 1955, Preston Smith wrote a landmark paper on free waves in a cylindrical shell. That paper described the displacement components and dispersions of the flexural, shear, and longitudinal waves that propagate in helical directions in the wall of the cylinder. The present author first met Preston over forty years later, when the hand calculations of 1955 had been transferred to a computer. After that meeting in 1996, the group at BBN was interested in how these waves reflect from terminations. Preston developed an approach for solving this problem and worked with the group to implement it using finite element analysis of a cylindrical shell with terminations. The essence of the approach was to mechanically excite different mixtures of waves using different excitations. The amplitudes of incident waves were related to the amplitudes of reflected waves by a reflection matrix. This matrix quantified the wave conversion that occurs at shell terminations. In addition, Preston formulated the reciprocity conditions that must be satisfied by the reflection matrix. His work revealed the most important physics of a very complex system. The present lecture will describe his approach and will highlight his profound style of analysis.

8:55

2aSAa2. Vibrational response features of a locally excited fluid-loaded plate with attached mass-spring oscillator systems. David Feit (Treasurer, Acoustical Society of America, 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502, feit.d@att.net)

Preston Smith and others have examined the radiation features of locally excited plates that are periodically supported by inertial masses. This presentation looks at the vibrational response and on-surface pressure field of a locally excited fluid-loaded plate that has one or more attached mass-spring oscillators. The analysis makes use of the "rational function approximation" (RFA) representation of the fluid loading effect first introduced by Diperna and Feit (J. Acoust. Soc. Am. **114**(1), July 2003, pp. 194-199).

9:15

2aSAa3. Wave number filtering on a finite periodically supported plate: Implications on the vibration field, radiated power, and validity of SEA. Robert Haberman (Raytheon IDS, 11 Main Street, Mystic, CT 06355, Robert_C_Haberman@Raytheon.com)

Preston Smith published and presented a number of papers on wave propagation and sound radiation from periodically supported plates. An excellent reference is, "Radiation from Periodically Supported Fluid-Loaded Plates", BBN Report No. 3999, January 1979. In these papers he identifies the physics of Bloch wave radiation, coherent scattering from ribs and spatial attenuation. As an extension to Preston's work, the problem of propagation of a local isotropic wave field on a finite periodically supported plate is considered. The specific question to be addressed is: do the rib supports provide wave number filtering of the isotropic field as it propagates throughout the plate-stiffener system? This question is answered via a series of analytical and finite element models. The implications on radiated power and validity of the SEA isotropic vibration field assumption will be discussed.

9:35

2aSAa4. A review of recent advances in vibro-acoustic system response variance determination in statistical energy analysis: A tribute to Preston Smith, Jr. Robert M. Koch (Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell Street, Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708, Robert.M.Koch@navy.mil)

Since the pioneering work of Preston Smith, Jr. and Richard Lyon in 1959 in the development of the theory of Statistical Energy Analysis (SEA), followed by many others in the 1960's on through today, the US Navy has utilized the important SEA vibro-acoustic simulation approach for high frequency self- and radiated-noise predictions of a multitude of undersea vehicles and systems. As a tribute to Preston Smith, this talk will review the current state of research in the determination of the variance/probability distribution about the mean response of a system modeled in SEA. While the subject of system response variance (or confidence interval) has obviously been of interest since the inception of this energy-based statistical method, there has been significant recent research in the literature advancing this area that is worth reviewing. As an additional acknowledgement of Preston Smith's later important work in the area of underwater cylindrical shell acoustics, the current presentation will also revisit the canonical structural acoustic problem of a point-excited, finite cylindrical shell with fluid loading and compare SEA-derived radiated noise level predictions with a variety of different classical analytical and modern day numerical approach solutions (e.g., FEA, EFEA/EBEA, closed form plate and shell theory solutions).

9:55

2aSAa5. Macro ocean acoustics. Henry Cox (Information Systems and Global Solutions, Lockheed Martin, 4350 N. Fairfax Dr., Suite 470, Arlington, VA 22203, harry.cox@lmco.com)

Today, physical insight based on fundamental principles and invariants frequently gives way to precise computations based on sophisticated models with less than perfect inputs. In the spirit of Preston W. Smith it is appropriate to revisit what can inferred from the sound speed profile and simple power flux reasoning without extensive computations. For example, based on the sound speed profile, the angle at the axis, the turning depths, the grazing angles at the surface and bottom, ray angle diagrams, the cycle length, cycle time, the group velocity, the adiabatic invariant and the so-called shallow water invariant all can be parameterized in terms of the phase velocity of each mode or ray. Applications of Preston's ideas to ambient noise near the bottom on the deep ocean, coherence and interference patterns will be discussed.

10:15

2aSAa6. Beam broadening for planar transmit arrays with maximal transmit power constraint. Evan F. Berkman (Applied Physical Sciences Inc., 49 Waltham St, Lexington, MA 02421, fberkman@aphysci.com)

Broad active sonar transmit beamwidth enables wide sector search. Many active search sonars attain wide transmit beamwidth by virtue of a cylindrical or spherical geometry which provides naturally wide beamwidth when all elements are driven in phase. However, planar array geometry is desired for some applications. Arrays often utilize many elements with resulting aperture large compared to an acoustic wavelength in order to provide high power output. However, planar arrays with aperture large compared to an acoustic wavelength will have naturally narrow beams which cannot be appreciably broadened by conventional amplitude shading without sacrifice of total power output and inefficient use of transducer and power amplifier channels. Maintenance of broad sector coverage over large fractional bandwidth with all element channels fully driven for maximum power output over the entire frequency band is also challenging. Thus, in order to maintain efficiency of element usage a scheme has been developed for obtaining a specified broad transmit array beamwidth invariant over a wide frequency range by frequency dependent phasing of array elements with all elements constrained to uniform amplitude gain. The intuitive physical basis for the phasing scheme is described as well as the mathematical results. A numerical example is provided.

10:35

2aSAa7. Statistical characterization of multipath sound channels. Peter Cable (Applied Physical Sciences Corp., 135 Four Mile River Road, Old Lyme, CT 06371, petercable@att.net)

Averaged transmission characteristics for underwater sound channels, using energy flux descriptions, were independently introduced and developed by Leonid Brekhovskikh [Sov. Phys.-Acoust. 11, 126-134 (1965)], David Weston [J. Sound Vib. 18, 271-287(1971)], and Preston Smith [J. Acoust. Soc. Am. 50, 332-336 (1971)]. While most attention in these studies was focused on elucidating averaged transmission loss in range dependent environments, Preston, in particular, also examined the application of energy flux techniques to other statistical characterizations of multipath environments, including channel impulse response and spatial coherence. Preston's incisive physical insight resulted in channel statistical characterizations that were simple to apply, but notably effective for signal processing and sonar studies. My purpose in this talk will be to trace the development of statistical characterizations of multipath channels based on energy flux notions, and to sketch some specific applications of these ideas, such as to source or receiver motion induced acoustic fluctuations.

10:55

2aSAa8. A range recursive algorithm for random sum acoustic propagation loss prediction. Cathy Ann Clark (Sensors & Sonar Systems, NUWCDIVNPT, 1176 Howell Street, B1320, R457, Newport, RI 02841, cathy.clark@navy.mil)

The work of P. W. Smith, Jr. includes prediction of the averaged impulse response for sound transmission in a shallow-water channel. His predictions are most applicable in situations for which cycle mixing with range results in randomization of phase interference between modes. In this talk, a calculation of random summation propagation loss for these types of conditions is presented. An integral expression approximating the normal mode sum is reformulated through a change of variable to an integral with respect to cycle number. The resultant formulation leads to a recursive relation in the range variable which enables calculations to be simplified significantly. The integral formulation is shown to successfully reproduce propagation loss with range by comparison to measurements in a number of environments.

11:15

2aSAa9. Preston Smith's theory of the effect of heat radiation on sound propagation in gases. Allan D. Pierce (P.O. Box 339, East Sandwich, MA 02537, adp@bu.edu)

The 1996 Trent-Crede Medal encomium by Barger states that Smith's favorite paper was "Effect of heat radiation on sound propagation in gases" (JASA, 1957). Smith stated this also in a private communication some time earlier to the present writer. The paper was prompted by works by Stokes and Rayleigh in which heat effects were modeled by Newton's law of cooling, which presumes that the heat radiation out from a limited region of heated matter is proportional to the difference of the local temperature and that of the surrounding medium. Stokes in 1851 constructed a theory for how this assumption leads to a prediction of the dependence of phase velocity and attenuation on frequency, and this theory was used by Rayleigh in the Theory of Sound to analyze whether acoustical fluctuations were more nearly isothermal or adiabatic. However, as Smith pointed out, apparently for the first time, neither Stokes and Rayleigh fully understood the relevant physics. When the atomic nature of heat radiation within gases is taken into account, the effect of heat radiation (in contrast to the effect of thermal conduction) is negligible at all frequencies. For all frequencies for which the attenuation is small, the acoustic fluctuations are adiabatic.

11:35

2aSAa10. Preston Smith and NASA Contractor Report CR-160. Richard H. Lyon (Consulting, RH Lyon, 60 Prentiss Lane, Belmont, MA 02478, rhlyon@lyoncorp.com)

When I came from a post-doc in England to BBN in 1960, Preston had already been at BBN for a few years. One of his interests was the interaction of structural resonances and reverberant sound fields. He had found that if the damping of the structure were to vanish, the response would not diverge, but reach a limit proportional to the sound pressure alone, independent of structural parameters. It turned out that this limit corresponded to modal energy equality between the sound field and the structure and was consistent with work I had done at Manchester on the energy flow between resonators. The combination of the two approaches was the beginning of SEA, presented to the community in "Sound and Structural Vibration, NASA Contractor Report CR-160 by Preston W. Smith Jr. and Richard H. Lyon", March 1965, the first publication on Statistical Energy Analysis (SEA). Interestingly, the words "Statistical Energy Analysis" did not appear in the report, but the ideas and viewpoint were there.

TUESDAY MORNING, 23 OCTOBER 2012 LIDO, 8:30 A.M. TO 11:45 A.M.

Session 2aSAb

Structural Acoustics and Vibration: Guided Waves for Nondestructive Evaluation and Structural Health Monitoring I

Tribikram Kundu, Cochair
Civil Engineering & Engineering Mechanics, University of Arizona, Tucson, AZ 85721

Wolfgang Grill, Cochair
Institute of Experimental Physics II, University of Leipzig, Leipzig 04312, Germany

Chair's Introduction—8:30

Invited Papers

8:35

2aSAb1. Monitoring of corrosion in pipelines using guided waves and permanently installed transducers. Michael J. Lowe, Peter Cawley, and Andrea Galvagni (Mechanical Engineering, Imperial College London, South Kensington, London SW7 2AZ, United Kingdom, m.lowe@imperial.ac.uk)

Guided Wave Testing (GWT) of pipelines for the detection of corrosion has been developed over about 20 years and is now a well established method worldwide, used mostly in the oil and gas industry. The established approach is as a screening tool: GWT is used to detect the presence of significant reflectors which are then examined locally in detail using conventional methods of NDE. To date most of the equipment has been developed for deployment solely at the time of test. However recent developments include permanently-attached transducers which can be left in place after testing, for example to allow easier access for future testing at difficult locations such as buried pipes. This is enabling a new approach, in which improved sensitivity may be achieved by detecting changes with respect to earlier reference signals, and also continuous monitoring which may follow degradation during service. The presentation will include a summary of the GWT method and discussion of current research for monitoring.

9:00

2aSAb2. Estimation of adhesive bond strength in laminated safety glass using guided mechanical waves. Henrique Reis (Industrial and Enterprise Systems Engineering, University of Illinois at Urbana-Champaign, 117 Transportation Building, 104 South Mathews, Urbana, IL 61801, h-reis@illinois.edu)

Laminated safety glass samples with different levels of adhesive bond strength were manufactured and tested using mechanical guided waves. The adhesive bond strength of the test samples was then also evaluated using the commonly used destructive testing method, i.e., the pummel test method. The interfaces between the plastic interlayer and the two adjacent glass plates are assumed to be imperfect and are modeled using a bed of longitudinal and shear springs. The spring constants were estimated using fracture mechanics concepts in conjunction with surface analysis of the plastic interlayer and of the two adjacent glass plates using atomic force microscopy and profilometer measurements. In addition to mode shape analysis, the phase and energy velocities were calculated and discussed. The guided wave theoretical predictions of adhesion levels using energy velocities were validated using the experimental pummel test results. From the attenuation dispersion curves, it was also observed that the S1 mode exhibits attenuation peaks in specific frequency ranges, and that the attenuation of these peaks is sensitive to the interface adhesion levels. Results show that this guided wave approach is useful in the nondestructive assessment of adhesive bond strength in laminated safety glass.

9:25

2aSAb3. Incorporating expected sparsity of damage into ultrasonic guided wave imaging algorithms. Jennifer E. Michaels and Ross M. Levine (School of Electrical and Computer Engineering, Georgia Institute of Technology, 777 Atlantic Drive, NW, Atlanta, GA 30332-0250, jennifer.michaels@ece.gatech.edu)

Many imaging methods employing ultrasonic guided waves are based upon delay-and-sum algorithms whereby echoes scattered from sites of damage are constructively reinforced after signal addition. Resolution of the resulting images depends upon such factors as the underlying array geometry, spectral content, knowledge of the propagation environment, and incorporation of phase information. For plate-like structures of engineering interest, geometrical features such as edges, cut-outs and fastener holes contribute to signal complexity and can cause significant image artifacts, which hinders detection and localization of actual damage. However, it is reasonable to make the *a priori* assumption that damage is spatially sparse. If this assumption is properly incorporated into imaging algorithms, then the resulting images should also be sparse and thus be easier to interpret. Several algorithms are developed and implemented that are based upon sparse reconstruction methods, and their performance on both numerical and experimental data is evaluated in terms of image quality and computational efficiency.

9:50

2aSAb4. Modeling of nonlinear guided waves and applications to structural health monitoring. Claudio Nucera and Francesco Lanza di Scalea (University of California San Diego, 9500 Gilman Drive, La Jolla, CA 92093-0085, flanza@ucsd.edu)

Research efforts on nonlinear guided wave propagation have increased dramatically in the last few decades because of the large sensitivity of nonlinear waves to structural condition (defects, quasi-static loads, instability conditions, etc...). However, the mathematical framework governing the nonlinear guided wave phenomena becomes extremely challenging in the case of waveguides that are complex in either materials (damping, anisotropy, heterogeneous, etc...) or geometry (multilayers, geometric periodicity, etc…). The present work develops predictions of nonlinear second-harmonic generation in complex waveguides by extending the classical Semi-Analytical Finite Element formulation to the nonlinear regime, and implementing it into a highly flexible, yet very powerful, commercial Finite Element code. Results are presented for the following cases: a railroad track, a viscoelastic plate, a composite quasi-isotropic laminate, and a reinforced concrete slab. In these cases, favorable combinations of primary wave modes and resonant double-harmonic nonlinear wave modes are identified. Knowledge of such combinations is important to the implementation of structural monitoring systems for these structures based on higher-harmonic wave generation. The presentation will also present a specific application of nonlinear guided waves for the monitoring of thermal stresses in rail tracks to prevent buckling.

10:15–10:30 Break

10:30

2aSAb5. Imaging-based quantitative characterization of fatigue crack for structural integrity monitoring using nonlinear acousto-ultrasonics and active sensor networks. Zhongqing Su (The Department of Mechanical Engineering, The Hong Kong Polytechnic University, Office: FG 642, Kinmay W. Tang Building, Hong Kong, MMSU@polyu.edu.hk), Chao Zhou, Li Cheng, and Ming Hong (The Department of Mechanical Engineering, The Hong Kong Polytechnic University, Hong Kong, Kowloon, Hong Kong)

The majority of today's damage detection techniques rely substantially on linear macroscopic changes in either global vibration signatures or local wave scattering phenomena. However, damage in real-world structures often initiates from fatigue cracks at microscopic levels, presenting highly nonlinear characteristics which may not be well evidenced in these linear macroscopic changes. It is of great significance but also a great challenge to quantitatively characterize micro-fatigue cracks without terminating the normal operation of an engineering structure. This is a critical step towards automatic and online structural integrity monitoring (SIM). By exploring the nonlinearities of higher-order acousto-ultrasonic (AU) waves upon interaction with fatigue cracks, a damage characterization approach, in conjunction with use of an active piezoelectric sensor network, was established, with the particular capacity of evaluating multiple fatigue cracks at a quantitative level (including the co-presence of multiple cracks, and their individual locations and severities). Identification results were presented in pixelated images using an imaging algorithm, enabling visualization of fatigue cracks and depiction of overall structural integrity in a quantitative, rapid and automatic manner. The effectiveness of the proposed technique was demonstrated by experimentally characterizing multiple fatigue cracks near rivet holes in aluminium plates.

10:55

2aSAb6. Ultrasonic waves for the inspection of underwater waveguide structures. Elisabetta Pistone and Piervincenzo Rizzo (Civil and Environmental Engineering, University of Pittsburgh, 3700 O'Hara Street, Pittsburgh, PA 15261, pir3@pitt.edu)

The non destructive inspection of immersed structures is popular as it minimizes unexpected and costly failures of important marine structures. In this paper we present a non-contact laser/immersion transducer technique for the inspection of underwater waveguide structures. The technique uses laser pulses to generate leaky guided waves and conventional immersion transducers to detect these waves. To prove the feasibility of the proposed methodology, a laser operating at 532 nm is used to excite leaky guided waves in a plate subjected to different damage scenarios. The plate is immersed in water at constant temperature and damage is first simulated using different weights located in the region of interest, i.e. between the point of the laser illumination and the immersion transducers. Damage is also simulated by engraving a series of notches on the face of the plate exposed to the probing system. The waveforms are then processed using the joint time-frequency analysis of the Gabor wavelet transform, statistical features and advanced signal processing techniques to identify and locate the presence of the defects. The findings show that the probing system and the signal processing algorithm used are able to detect differences between pristine and damaged conditions.

11:20

2aSAb7. Defect visualization in pipes using a longitudinal guided wave mode. Hyeonseok Lee (Korea Advanced Institute of Science and Technology, Daejeon, Republic of Korea), Hyun Woo Park (Department of Civil Engineering, Dong-A University, Busan, Republic of Korea), and Hoon Sohn (Department of Civil and Environmental Engineering, Korea Advanced Institute of Science and Technology, 291 Daehak-Ro, Yuseong-Gu, Daejeon 305-701, Republic of Korea, hoonsohn@kaist.ac.kr)

Recently, defect visualization techniques based on guided waves have been developed for pipe inspection. This study advances existing defect visualization techniques in two ways: (1) a fiber-guided laser ultrasonic system, which can operate under high radiation and temperature environments, is used to generate and measure broadband guided waves, and (2) a longitudinal mode instead of a torsional mode is used to detect axial cracks and wall thinning. Using optical fiber probes installed along a circumferential direction of a pipe with equal spacing, a pure longitudinal mode, L(0,2), is launched by axisymmetrically exciting a pipeline structure. The generated L(0,2) subsequently interacts with scattering sources such as defects or pipe boundaries and generate reflected L(0,2) and higher-order modes, L(n,2), (n>0). The reflected modes, L(0,2) and L(n,2), are measured in a pulse-echo manner using the same fiber probes and synthetically processed. By back propagating the dispersive L(0,2) and L(n,2) modes in time and space, this study reconstructs dispersion-compensated L(0,2) and L(n,2) at the scattering sources and thereby visually locates the defects. Numerical simulation and experimental studies are performed to validate the effectiveness of the proposed technique.

TUESDAY MORNING, 23 OCTOBER 2012 TRUMAN A/B, 8:00 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication: Cross-Language Production and Perception of Speech (Poster Session)

Wendy Herd, Chair
Mississippi State University, Mississippi State, MS 39762

Contributed Papers

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.

2aSC1. Comparing L1 and L2 phoneme trajectories in a feature space of sound and midsagittal ultrasound tongue images. Keita Sano, Yuichi Yaguchi, and Ian Wilson (University of Aizu, Ikkityo Kamega Fujiwara 198, Aizuwakamatsu, Fukushima 965-0005, Japan, ksano065@gmail.com)

To support the development of pronunciation training systems for non-native (L2) speakers, past research has proposed visualization of a speaker's tongue using ultrasound as feedback showing differences between L2 and native (L1) speakers. However, there has been little or no quantitative assessment combining temporal variation of speech sounds and ultrasound tongue images. We propose a mining method to analyze such temporal differences between L1 and L2 speakers. We firstly construct two eigenspaces: one made from feature vectors of speech sounds using Spectrum Vector Field (SVF) and the other from ultrasound tongue images using Histogram of Oriented Gradients (HOG). Next, we compare the movements of L1 and L2 trajectories. Furthermore, we model the connection of phonemes by finding tongue shapes from adjacent speech sounds, and we indicate the differences between L1 and L2 speakers to make a clear intermediate representation from the feature space. In our experiment, we analyze the differences between L1 and L2 pronunciation by focusing on the temporal trajectories of the feature space. These trajectory differences between L1 and L2 speakers' speech sounds will be presented. We will also present the feature space of ultrasound tongue images that indicate the intermediate tongue shapes mentioned above.

2aSC2. Delayed feedback disrupts optimal strategies during foreign speech sound learning. Bharath Chandrasekaran, Han-Gyol Yi (Communication Sciences and Disorders, University of Texas, Austin, TX 78712, bchandra@mail.utexas.edu), and W. Todd Maddox (Psychology, University of Texas, Austin, TX)

The Competition between Verbal and Implicit Systems (COVIS) model posits that an explicit hypothesis-testing system competes with an implicit procedural-based system to mediate category learning. During early learning, the hypothesis-testing system is dominant, whereas in later learning, the procedural system dominates. In visual category learning, delayed feedback is known to impair the procedural but not the hypothesis-testing system. We tested the COVIS model in natural auditory category learning. Young adult native English speakers learned to categorize lexical tones in Mandarin syllables. Timing of feedback was either immediate (0 s) or delayed (500 or 1000 ms). Consistent with COVIS, delay feedback affected accuracy only in later learning. Further, modeling analysis revealed that participants were more likely to adopt procedural strategies during later learning, but this transition was disrupted by delayed feedback. These results will be discussed in the context of developing methods to optimize foreign speech sound learning.

2aSC3. Training adult learners of English to hear the sounds of English. Charles S. Watson, James D. Miller, and Gary R. Kidd (Research, Communication Disorders Technology (CDT), Inc., Bloomington, IN 47404, jamdmill@indiana.edu)

Adult students of foreign languages frequently claim that native speakers of that language speak too rapidly. This is likely a result of the students' failure to achieve automaticity in recognition of speech sounds necessary for effortless speech perception. Research on the time course of auditory perceptual learning for both speech and non-speech sounds provides strong evidence that adults can, with appropriate training, achieve perceptual skills approximating those of native speakers, although they only rarely do so. Among the few adults who do achieve near-native conversational skills in an L2, many have had intensive recognition practice and training. The Speech Perception Assessment and Training System for students of English as a Second Language (SPATS-ESL) of CDT, Inc. provides such training. SPATS-ESL trains the identification of the 109 most common English syllable constituents (onsets, nuclei, and codas) and the recognition of meaningful sentences spoken by a variety of native speakers. Based on experience with over 200 ESL learners it has been found that near-native performance in the recognition of discrete English speech sounds and meaningful sentences spoken by many talkers is acquired by most ESL students after 15-30 hours of individualized computer-based training. (Watson and Miller are stockholders in CDT, Inc.)

2aSC4. High variability training increases mismatch negativity responses to L2 contrasts. Wendy Herd (English Dept., Mississippi State University, 100 Howell Hall, PO Box E, Mississippi State, MS 39762, wherd@english.msstate.edu), Robert Fiorentino, and Joan Sereno (Linguistics Dept., University of Kansas, Lawrence, KS)

Previous research established that high variability training improves both perception and production of novel L2 contrasts and that training noncontrastive sounds in subjects' L1 results in increased MMN responses to those sounds. However, it is unclear whether training novel contrasts in an L2 also results in increased amplitude of MMN responses to the contrasts. This study trained 10 American English learners of Spanish, for whom tap and /d/ are noncontrastive, to distinguish the phonemic tap-/d/ contrast in Spanish to determine if training also changed MMN responses to those sounds when presented in an oddball paradigm. First, the amplitude of native Spanish speakers' (N=10) MMN response to deviant tap was significantly more negative than to the standard, establishing this paradigm elicited canonical MMN responses. Second, trainees (N=10) and controls (N=10) did not exhibit significantly different responses to deviant and standard tap at pretest. Crucially, this was not the case at posttest. Trainees, like native Spanish speakers, exhibited a significant MMN response to deviant tap compared to the standard at posttest, but controls did not. The emergence of an MMN response in the trainees indicates it is possible to recategorize L1 contrasts when learning an L2. [Supported by NSF 0843653.]

2aSC5. Perception of speech-in-noise for second language learners and heritage speakers in both first language and second language. Michael Blasingame and Ann R. Bradlow (Linguistics, Northwestern University, 2016 Sheridan Road, Evanston, IL 60208, mblasingame@u.northwestern.edu)

This study asks whether speech recognition by bilingual listeners in each of their two languages follow complementary or supplementary patterns. Previous studies showed that early bilinguals are disproportionately affected by adverse listening conditions in L2 (Mayo et al., 1997; Shi et al, 2010; Bradlow & Alexander, 2007), but did not measure performance in L1. The current study extends these results using bilingual performance under adverse listening conditions in both languages to determine whether reduced use of the dominant language by relatively well-balanced bilinguals affects performance in L1 as well as L2. We examine two groups of English-Spanish bilinguals: Spanish learners (SL) and Spanish heritage speakers (SHS). Although both English dominant, crucial differences between these groups are L1 (SL=English, SHS=Spanish) and L1-L2 balance (SL=large imbalance, SHS=relatively balanced). Both groups were presented with sentences in English and Spanish in which final keywords varied on three factors: speech style (clear versus plain/conversational), contextual predictability (high versus low), and signal-to-noise ratio (easier versus harder). Results show SHS do not pattern like monolinguals in either language, yet average performance across both languages is higher than SL. This result suggests the overall system SHS maintain is "larger" than SL, but may be more susceptible to noise.

2aSC6. Idiosyncrasy and generalization in accent learning. Meghan Sumner and Ed King (Linguistics, Stanford University, 450 Serra Mall, Stanford, CA 94035, sumner@stanford.edu)

People understand speech well, despite pronunciation variation. Perceptual learning, where listeners are trained with acoustic features ambiguous between two phonemes and subsequently shift their perceived phoneme boundary, is one way listeners may compensate for variation (Norris et al 2003). These perceptual shifts, however, seem idiosyncratic to one speaker (Eisner & McQueen 2005, Kraljic & Samuel 2005), rarely generalizing to new speakers. We propose that lack of generalization is due to lack of experience mapping phonemes to specific continua; previous work uses continua like [s]-[f], whose midpoints rarely occur in speech. Ambiguous tokens that are never heard in real speech may be perceived as specific to the speaker used in training, preventing generalization. Use of a continuum occurring in accented speech, such as the mapping of English tenseness onto vowel duration, allows manipulation of the idiosyncrasy of the mapping. We train 13 listeners on an idiosyncratic duration mapping (lax to short duration, tense to ambiguous duration) and 11 on an Italian accent pattern (lax to ambiguous duration, tense to long), and test generalization to a different speaker. Listeners generalize the Italian accent, and generalize away from the idiosyncratic pattern. This suggests listeners generalize likely accents, treating unlikely patterns as idiosyncratic.

2aSC7. Functional significance of the acoustic change complex, mismatch negativity, and P3a for vowel processing in native-English listeners and late learners of English. Brett A. Martin, Valerie Shafer (Speech-Language-Hearing Science, Graduate Center of the City University of New York, 365 Fifth Avenue, New York, NY 10016, bmartin@gc.cuny.edu), Marcin Wroblewski (Communication Sciences & Disorders, University of Iowa), and Lee Jung An (Speech-Language-Hearing Science, Graduate Center of the City University of New York, New York, NY)

The acoustic change complex (ACC), mismatch negativity (MMN), and P3a all provide indices of the neural processing of the types of acoustic changes that underlie speech and language perception. The goal of this study was to compare neural correlates of vowel processing for contrasts that have been shown to be easy to perceive in native-English speakers but more difficult for native-Spanish speakers. Processing of a vowel change from /I/ to /E/ was compared in a group of late learners of English and a group of monolingual English listeners (n = 15 per group). Preliminary analyses suggest differences in processing of the vowel change from /I/ to /E/ across the groups. Monolinguals processed the vowel change more rapidly and more accurately than bilinguals. The obligatory response to vowel onset showed a larger N1 for the bilinguals compared to the monolinguals. In addition, group differences were obtained in mean global field power (MMN and P3a were longer for bilinguals) and topography (current source density showed group differences for ACC P2 component, MMN, and P3a). Therefore, ACC, MMN and P3a all showed the effects of native language experience; however, these effects were not identical for each component.

2aSC8. A moving target? Comparing within-talker variability in vowel production between native and non-native English speakers across two speech styles. Catherine L. Rogers, Amber Gordon, and Melitza Pizarro (Dept. of Communication Sciences and Disorders, University of South Florida, 4202 E. Fowler Ave., Tampa, FL 33620, crogers2@usf.edu)

Non-native English speakers may show greater variability in speech production than native talkers due to differences in their developing representations of second-language speech targets. Few studies have compared within-talker variability in speech production between native and non-native speakers. In the present study, vowels produced by four monolingual English speakers and four later learners of English as a second language (Spanish L1) were compared. Five repetitions of six target syllables ("bead, bid, bayed, bed, bad" and "bod"), produced in conversational and clear speech styles, were analyzed acoustically. Fundamental and formant frequencies were measured at 20, 50 and 80% of vowel duration. Standard deviations computed across the five repetitions of each vowel were compared across speaking styles and talker groups. Preliminary data analyses indicate greater within-talker variability for non-native than native talkers. Non-native talkers' within-talker variability also increased from conversational to clear speech for most measures. For some native talkers, within-talker variability was smaller for vowels with near neighbors in the vowel space than for vowels with more spectrally distant neighbors. This correlation was stronger in clear speech for talkers who showed a significant clear-speech intelligibility benefit in production in a related study. Implications for theories of vowel production will be discussed.

2aSC9. Processing reduced speech across languages and dialects. Natasha L. Warner, Daniel Brenner, Benjamin V. Tucker (Linguistics, University of Arizona, Tucson, AZ 85721-0028, nwarner@u.arizona.edu), Jae-Hyun Sung (Linguistics, University of Arizona, Tucson, AZ), Mirjam Ernestus (Centre for Language Studies, Radboud University Nijmegen, Nijmegen, Gelderland, Netherlands), Miquel Simonet (Spanish and Portuguese, University of Arizona, Tucson, AZ), and Ana Gonzalez (Linguistics, University of Arizona, Tucson, AZ)

Normal, spontaneous speech utilizes many reduced forms. Consonants in spontaneous speech frequently have a different manner or voicing than would be expected in clear speech (e.g. /d/ and /ŋ/ in "you doing" both being realized as glides or /dʒ/ in "just" as a fricative), and near or complete deletions are

also common (e.g. the flap in "a little"). Thus, listeners encounter and must process such pronunciations frequently. When speakers and listeners do not share the same dialect or native language, such reductions may hinder processing more than for native listeners of the same dialect. The current work reports a lexical decision experiment comparing listeners' processing of reduced vs. careful stops (e.g. /g/ in"baggy" pronounced as an approximant or as a stop), by several groups of listeners. Results show that listeners from both Arizona and Alberta, Canada can recognize speech by an Arizona speaker with reduced stops, but they recognize the words more easily when stops are clearly articulated. Speech style of the preceding frame sentence has little effect, suggesting that both groups can process the stops regardless of whether surrounding context leads them to expect reduced stops. Additional data from second-language learners and bilingual listeners is currently being collected.

2aSC10. Evidence of Spanish undershoot in a Mexican-American community. Arika B. Dean (English (Linguistics), North Carolina State University, Raleigh, NC 27603, abdean@ncsu.edu)

Previous phonetic work on Spanish vowels has suggested that undershoot does not occur in the Spanish vowel system. Quilis & Esgueva (1983) suggested that Spanish vowels were static with little to no articulatory variation. This study contributes to a growing body of phonetic research attempting to disprove these claims. The data comes from audio interviews conducted in a predominantly Hispanic community in Pearsall, Texas. Acoustic analysis is conducted on all vowels in the Spanish vowel range (/i/, /e/, /a/, /o/, /u/), and each token is measured from the midpoint of the vowel nucleus, as well as onset and offset. Subjects are Spanish speakers of different age, sex, language competency, and socioeconomic status. In response to Lindblöm's (1963) assertion that stress is more significant than duration in determining undershoot, this study raises that question again and finds that stress is more statistically significant than duration in the Spanish undershoot process, contrary to Lindblöm's findings. These results also contradict Willis (2005), who found that duration influences the F1 value for Spanish vowels. The effects of English substrate influence are considered, and the vowels of monolingual Mexican Spanish speakers are analyzed, providing a control against the speakers in the Pearsall community.

2aSC11. Abstract withdrawn

2aSC12. Phonemic processing in compensatory responses of French and English speakers to formant shifted auditory feedback. Takashi Mitsuya (Psychology, Queen's University, 62 Arch Street, Humphrey Hall, Kingston, ON K7L3N6, Canada, takashi.mitsuya@queensu.ca), Fabienne Samson (Psychology, Queen's University, Kingston, ON, Canada), Lucie Ménard (Linguistics, Université du Québec à Montréal, Montréal, QC, Canada), and Kevin G. Munhall (Psychology & Otolaryngology, Queen's University, Kingston, ON, Canada)

Past studies have shown that speakers modify their vowel formant production when auditory feedback is altered in order to make the feedback more consistent with the intended sound. This behavior was thought to minimize acoustic error overall; however, Mitsuya et al. (2011) showed different magnitudes of compensation for altered F1 across two language groups depending on the direction of perturbation. Their results seem to reflect how the target vowel is represented in relation to other vowels around it. From this observation, they proposed that compensation is to maintain perceptual identity of the produced vowel, requiring some phonological processes for error reduction. Yet, the results might have been specific to the language groups examined, and/or unique to F1 production. To generalize Mitsuya et al.'s hypothesis, the current study examined 1) different language groups and 2) F2 production. We compared compensatory behavior of F2 for /ɛ/ among French speakers (FRN) and English speakers (ENG) with decreased F2 feedback. With this perturbation, the feedback sounded like /œ/, which is phonemic in French but not in English. The preliminary data suggest that FRN compensated in response to smaller perturbations and showed greater maximum compensations than ENG.

2aSC13. Production of English vowels by speakers of Mandarin Chinese with prolonged exposure to English. Keelan Evanini and Becky Huang (Educational Testing Service, Rosedale Rd., Princeton, NJ 08541, kevanini@ets.org)

Previous studies of non-native production of English vowels have demonstrated that a native-like attainment of certain distinctions is not guaranteed for all speakers, despite prolonged exposure to the target (e.g., Munro et al. 1996, Flege et al. 1997). The current study examines the applicability of this finding to a group of non-native speakers from the same L1 background (Mandarin Chinese) who are all long-term residents in the USA (7 years minimum) and adult arrivals (> age 18). These non-native speakers (N=36) and a control group of native speakers (N=22) were recorded reading two sets of materials: the Stella paragraph (Weinberg 2012) and five sentences from Flege et al. (1999). Vowel formant measurements were extracted for all tokens from the following three pairs of vowels: [i] ~ [ɪ], [e] ~ [ɛ], and [a] ~ [ʌ]. Euclidean distances between the z-normalized (F1, F2) mean values for the two vowels in each pair for each speaker show that the non-native speakers produce each of the three pairs significantly less distinctly than the native speakers. This finding corroborates previous similar findings and suggests that a speaker's L1 continues to have a strong influence on vowel production, despite long-term exposure to the target.

2aSC14. Quantifying the consonantal voicing effect: Vowel duration in an Italian–American community. Ylana Beller-Marino and Dianne Bradley (Linguistics, CUNY Graduate Center, New York, NY 10010, ybeller@gc.cuny.edu)

Cross-linguistically, vowel duration preceding voiced consonants is greater than that preceding voiceless consonants, all else equal (Chen 1970, Mack 1982). Notably, this consonantal voicing effect is larger for English, a presumed instance of language-specific phonological enhancement of a basic phonetic process. The current study asks whether bilingual speakers maintain separate durational settings, and compares consonantal voicing effects across languages in two participant groups: foreign-born Italian speakers who acquired English as young adults, and US-born speakers from the same community who had simultaneous childhood exposure to Italian and English. The complete materials set employed familiar words, e.g., English *rib*/*rip*; Italian *cubico*/*cupola*, and sampled systematically over vowel height and consonantal place; data reported are drawn from the high-vowel materials subset only. For targets uttered within language-appropriate carrier phrases, both groups exhibited the consonantal voicing effect in each language; both also exhibited the same interaction with language, producing reliably larger effects in English, suggesting that language-specific settings were attained. But crucially, where foreign-born speakers produced a purely phonetic effect in Italian, US-born speakers suppressed phonological enhancement only partially. These findings, plausibly reflecting a degree of interplay between phonologies, are discussed in terms of the circumstances of language learning.

2aSC15. How tongue posture differences affect reduction in coronals: Differences between Spanish and English. Benjamin Parrell (University of Southern California, University of Southern California, Department of Linguistics, GFS 301, Los Angeles, CA 90089, parrell@usc.edu)

It has been suggested that both flapping of English coronal stops [e.g. Fukaya & Byrd, JIPA, 2005; De Jong, JPhon, 1998] and spirantization of Spanish voiced stops [e.g. Parrell, LabPhon, 2012] result from reductions in duration. If this is indeed the case, why would reducing duration in one language lead to spirantization (Spanish) and in another to flapping (English)? We suggest that these differences are the result of different ways the tongue is used to attain oral closure in the two languages: in Spanish, coronal stops are made with blade of the tongue at the teeth; in English, with the tongue tip placed at the alveolar ridge. Because of this difference, the tongue tip is oriented differently in the two languages: upward in English and downwards in Spanish, leading to differing articulatory and acoustic outcomes as duration is shortened. We examine these postural differences using tongue movement data, which allows for direct and dynamic examination of tongue posture and shaping of coronals in both languages. Differences between the two languages will be modeled using TaDA [Nam et al., JASA, 2004] to test how they may lead to different articulatory and acoustic outcomes as duration is reduced. [Supported by NIH.]

2aSC16. American Chinese learners' acquisition of L2 Chinese affricates /ts/ and /tsʰ/. Jiang Liu and Allard Jongman (Linguistics, University of Kansas, 1541 Lilac Lane, Lawrence, KS 66044, liujiang@ku.edu)

Many studies on L2 speech learning focused on testing the L1 transfer hypothesis. In general, L2 phonemes were found to be merged with similar L1 phoneme to different degrees (Flege 1995). Few studies examined

whether non-phonemic phonetic categories such as consonantal clusters in L1 help or block the formation of new phonetic categories in L2. The current study examined the effect of L1 English consonantal clusters [ts] (e.g., the ending of the plural noun 'fruits') and [dz] (e.g., the ending of the plural noun 'foods') on learning L2 Chinese affricates /ts/ and /ts^h/. We studied duration and center of gravity (m1) of L2 Chinese affricates /ts/ and /ts^h/ produced by native Chinese speakers, novice American Chinese learners and advanced learners. In terms of duration, both learner groups showed contrast between L2 /ts/ and /ts^h/, which is similar to native Chinese speakers' production. However, for m1, only the advanced learner group showed contrast between L2 /ts/ and /ts^h/, which is similar to native speakers' production while the novice learner group did not show m1 contrast between the two L2 affricates. The duration result can be accounted for by the existence of durational difference between L1 English [ts] and [dz] whereas the lack of m1 contrast between the two L2 affricates for the novice learner group can be accounted for by the absence of m1 difference between L1 English [ts] and [dz].

2aSC17. The production and perception of English stops in a coda position by Thai speakers. Siriporn Lerdpaisalwong (Dept. of Linguistics, University of Wisconsin-Milwaukee, Milwaukee, WI 53201, siriporn@uwm.edu)

This paper reports results from a pilot study on the production and perception of English stops in a coda position by native speakers of Thai with different length of residency (LOR) in the US. This study explores three important issues in second language (L2) acquisition: typological markedness (Eckman, 1997), the relationship between production and perception of speech sounds (Flege, 1988 and 1999), and the length of learning L2 sounds (Flege 1999). There were 13 Thai-speaker participants whose LORs ranged from 1 year to 23 years. They participated in two tasks: sentence reading in a production task, and sentence listening in a perception task. Preliminary results show that participants produced all English voiced stops less accurately than voiceless stops. However, in the perception task, only /g/ was perceived less accurately than voiceless stops. The speakers perceived /b/ better than /k/ and perceived /d/ better than /p/ and /k/. The more accurate the speakers can perceive the sounds, the better they can produce it. The Thai speakers with a longer LOR perceived and produced English stops in the coda position more accurately than those with a shorter LOR. The results found raise our awareness of to which sounds should be paid special attention and the benefit of enough language input. Also, the study suggests the pattern of English stops acquired by native speakers of Thai in both production and perception processes.

2aSC18. Acoustic correlates of stop consonant voicing in English and Spanish. Olga Dmitrieva (Linguistics, Stanford University, 450 Serra Mall, Stanford, CA 94305, dmitro@stanford.edu), Amanda A. Shultz (Linguistics program, Purdue University, West Lafayette, IN), Fernando Llanos (School of Languages and Cultures, Purdue University, West Lafayette, IN), and Alexander L. Francis (Department of Speech, Language, and Hearing Sciences, Purdue University, West Lafayette, IN)

In English, fundamental frequency at the onset of voicing (onset f0) covaries with the Voice Onset Time (VOT) of initial stops and provides an additional perceptual cue to the phonetic feature of voicing, especially when VOT is ambiguous. However, aerodynamic and physiological explanations of the onset f0/VOT relationship suggest that onset f0 should correlate with voicing only in languages such as English that contrast short lag (voiceless unaspirated) with long lag (voiceless aspirated) consonants, and not in languages such as Spanish that contrast prevoiced with short lag stops. Previous perceptual research supports this prediction: Spanish speakers with little English experience do not incorporate onset f0 in making voicing decisions, suggesting lack of a correlation in their ambient language. In contrast, Spanish speakers with extensive experience with English show an English-like pattern of onset f0 use, suggesting that exposure to the English pattern of covariation has influenced their perceptual weighting of these two cues. The present study compares the distribution and correspondence between VOT and onset f0 in syllable-initial bilabial stops ([b] - [p]) in Spanish and English. Implications for the typology of voicing contrasts and perceptual strategies for sound categorization in non-native language environments are discussed.

2aSC19. Modeling learning of the English voicing contrast by Spanish listeners living in the United States. Fernando Llanos (Spanish & Portuguese, Purdue University, West Lafayette, IN), Alexander L. Francis (Speech, Language & Hearing Sciences, Purdue University, SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, francisa@purdue.edu), Olga Dmitrieva (Linguistics, Stanford University, West Lafayette, Indiana), Amanda A. Shultz (Linguistics, Purdue University, West Lafayette, IN), and Rachel Chapman (Speech, Language & Hearing Sciences, Purdue University, West Lafayette, IN)

The importance of cue covariation in phonetic learning is explored through four experiments investigating perception of stop consonant voicing. Spanish and English show different uses of voice onset time (VOT; the time between consonant burst release and vocalic voicing onset) in cuing voicing perception. English contrasts short lag (<20 ms) with long lag (>20 ms) stops, whereas Spanish contrasts prevoicing (<0 ms) with short lag (>0 ms) stops. Secondary cues may also differ. In English, VOT and onset f0 (fundamental frequency at voicing onset) are positively correlated, and onset f0 plays a role in voicing perception. In Spanish these properties may not be as strongly correlated, meaning that onset f0 may be less relevant to voicing perception. As predicted, Spanish listeners tested in Spain showed a 0 ms VOT boundary with little use of onset f0, whereas English listeners tested in the US showed a 20 ms boundary with moderate use of onset f0. Significantly, Spanish listeners tested in the US showed an English-like VOT boundary, but made even greater use of onset f0 than did English listeners. Computational Hebbian modeling suggests a role for differences in each groups' experience with specific patterns of VOT/onset f0 covariation.

2aSC20. Perception of American English final consonants by speakers of New York-Dominican Spanish. Shari S. Berkowitz (Communication Disorders, Mercy College, 555 Broadway, Main Hall, G-14-B, Dobbs Ferry, NY 10522, shariellen@gmail.com)

The English language uses many final consonants and final clusters to convey meaning, especially for morphological endings. The Spanish language employs fewer final consonants than the English language, and Caribbean Spanish speakers treat many final consonants as optional. In this experiment, speakers originating from the Dominican Republic (N = 25) participated in a listening task in which they had to identify final consonants in fast and clear sentences in English (stimulus corpus Ito, K, 2011). A small group of native American English speakers was tested, and performed at ceiling. Spanish-speaking participants' performance on the experimental task varied from 40% to native-like accuracy and was statistically different from the native speakers' performance (Mann Whitney $U = 8$, $p < .003$). In addition, Spanish-speaking paritcpants' performance on the final consonant perception task correlated strongly with performance on a standardized aural/oral language battery known as the Versant Test (Pearson Corp, 2011)($r = .76$, $p < .001$); performance also correlated strongly with age of acquisition. The coarticulation of adjacent speech sounds played a role in which consonants were most difficult to perceive. Future directions, including the current testing of speakers of Puerto Rican Spanish, Kannada and Russian, and implications for intervention, will be discussed.

2aSC21. Featural enhancement of Spanish word-initial stops in clarifications of misheard words. Jessamyn L. Schertz (Linguistics, University of Arizona, Tucson, AZ 85721, jschertz@email.arizona.edu)

In an experiment exploring phonetic featural enhancement in Spanish, native speakers were asked to read words aloud, then repeat them when a supposed automatic speech recognizer "guessed" incorrectly (e.g. subject says "basta," computer displays (in Spanish) "Did you say 'pasta'?", subject repeats "basta"). In a previous experiment with the same paradigm, English speakers exaggerated VOT in the second repetition (longer prevoicing for voiced and longer aspiration for voiceless stops) when the incorrect guess was a minimal pair in voicing with the target word. Spanish speakers also had longer prevoicing durations for voiced stops, but unlike English speakers, showed no change in VOT for voiceless stops; in fact, VOT was shorter in the clarification, though not significantly. The differences in how speakers of the two languages manipulated the stops reflects cross-linguistic differences in the phonetic components of the stop contrast. Additionally, Spanish speakers produced fricatives for some of the word-initial voiced stops.

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Although Spanish voiced stops are realized as fricatives in many environments, they are not expected to be lenited following a pause. The results of this study confirm that speakers use inventory-specific featural manipulations to clarify contrasts, and demonstrate unexpected variability in post-pausal voiced stops in this dialect of Spanish.

2aSC22. The effects of first-language sound change on second-language speech production. Mi-Ryoung Kim (Practical English, Soongsil Cyber University, Dept. of Practical English, Soongsil Cyber University, 307 Jongno Biz-well, 34 Ikseon-dong, Seoul, OR 110-340, Republic of Korea, kmrg@mail.kcu.ac)

Recent studies have shown that the stop system of Korean is undergoing a sound change in which a consonantal opposition between lax and aspirated stops is merging in terms of voice onset time (VOT) whereas the contrast between the two stops is being maximized in terms of fundamental frequency (f0). This study investigates how the ongoing sound change of acoustic parameters in L1 Korean influences L2 English stop production. Results showed that, unlike the VOT merger in L1 Korean, it does not occur in L2 speech production. In contrast, similar to onset-f0 interaction in L1 Korean, there is a strong onset-f0 interaction in L2 English: voiced-low f0 and voiceless-high f0. Korean English learners use not only VOT but also f0 in contrasting an underlying [voice] distinction. The results suggest that f0 differences between lax and aspirated stops in L1 Korean are transferred to those between voiced and voiceless counterparts in L2 English. The findings are discussed with respect to cross-language phonetic effects and synchronic sound change.

2aSC23. Perception of place-of-articulation contrasts of English word-final consonants in connected speech by Japanese and Korean second language learners. Kikuyo Ito and JungMoon Hyun (Ph.D. Program in Speech-Language-Hearing Sciences, Graduate Center, CUNY, Grad Center, CUNY, 365 5th Ave., New York, NY 10016, kikuyoito@hotmail.com)

An extension of a previous study examining Japanese listeners' perception of place contrasts of English word-final stops in connected speech (Ito, 2010) was carried out by administering the same experiment to Korean listeners. Stimuli embedded in a carrier sentence and produced in fast casual speech were presented in a three-alternative forced choice identification test, adopting minimal triplets (e.g., sip-sit-sick, bib-bid-big, Kim-kin-king) followed by an adverb starting with /p/, /t/, or /k/. Data for 24 Korean listeners were compared with the previous data for 24 Japanese and 24 American English (AE) listeners. Whereas Japanese listeners had exhibited severe difficulty in perceiving place contrasts of nasal and voiceless stops, Korean listeners were expected to have much less perceptual difficulty on those contrasts because of the different L1 phonological rules of final stops. Results revealed that Koreans' response accuracy was much higher than that of Japanese on voiceless stops (Korean 82%, Japanese 67%, AE 90%) and nasal stops (Korean 96%, Japanese 66%, AE 98%), conforming to the predictions. The contrasting performance between Korean and Japanese listeners on nasal stops was especially remarkable, strongly supporting the notion that Japanese listeners' difficulty in perceiving place contrasts of word-final nasals is due to their L1phonological rules.

2aSC24. Non-native perception and production of Basque sibilant fricatives. Melissa M. Baese-Berk and Arthur G. Samuel (Basque Center on Cognition, Brain and Language, Paseo Mikeletegi, 69, Donostia, Guipuzkoa 20009, Spain, m.baese@bcbl.eu)

Differences in perception and production of non-native contrasts are thought to be driven by the relationship between sound inventories of the native and target languages (Best, McRoberts, and Goodell, 2001). The current study examines non-native perception and production of sibilant fricatives and affricates in Basque. Basque has a 3-way place contrast for sibilant fricatives and affricates (apico-alveolar /s̺/ and /ts̺/, lamino-alveolar /s̻/ and /ts̻/, and post-alveolar /ʃ/ and /tʃ/). In contrast, /s/ and /tʃ/ are the only voiceless sibilants that Spanish has in this region. The results suggest that in the case of Basque sibilant phonemes, similarity to an existing contrast (i.e., fricative-to-affricate contrasts) results in better perception and production. Native Spanish speakers performed better on discrimination and repetition of the /s̺/ - /ts̺/ contrast than the /s̺/ - /s̻/ or /s̺/ - /ʃ/ contrasts. Spanish speakers are able to leverage their ability to discriminate and produce /s/ and /tʃ/ in their native language to perceive and produce a new contrast in Basque, even though the contrast in Spanish differs by two features, rather than just one as in Basque. However, the lack of a contrast between sibilant fricatives prevents them from discriminating or producing the fricative-to-fricative contrasts.

2aSC25. Sibilant production patterns in three generations of Guoyu-Taiwanese bilinguals. Ya-ting Shih (Second and Foreign Language Education, The Ohio State University, Columbus, OH 43212, shih.68@buckeyemail.osu.edu), Jeffrey Kallay, and Jennifer Zhang (Linguistics, The Ohio State University, Columbus, OH)

This study investigates the effects of age and language dominance on sibilant production in a bilingual community. Guoyu (Taiwanese Mandarin) has 3 sibilants: alveolar /s/, retroflex /ʂ/ and alveolo-palatal /ɕ/, while Taiwanese (a Southern Min dialect) only has /s/, which is palatalized before /i/ and /io/. Productions of sibilant initial words were elicited using a word repetition task. Subjects were 30 adults in three age bands from 20-80 years, with the oldest being the most Taiwanese-dominant. The spectral centroid was obtained from the middle 40ms of each sibilant, along with the onset F2 of the following vowel. In the low-vowel /a/ context, the youngest speakers clearly separate /s/ in both languages from the Guoyu /ʂ/ and /ɕ/ along the centroid dimension. The /ʂ/ is then separated from /ɕ/ by F2. However, the oldest speakers show no clear separation of these sounds in terms of centroid, although Guoyu /ɕ/ can still be differentiated by F2. Also, the three generations demonstrated differences in the assimilation patterns of palatal sounds. The younger Guoyu-dominant speakers assimilated Taiwanese palatalized one to Guoyu /ɕ/ in both centroid and F2, while older speakers matched the Guoyu /s/~/ɕ/ distinction to that of the Taiwanese pattern.

2aSC26. Acoustic and perceptual similarities between Effutu and English fricatives: Implications for English as a second language. Charlotte F. Lomotey (Texas A&M University, Commerce, TX 75428, cefolatey@yahoo.com)

Volin & Skarnitzl (2010) describe a foreign accent as a set of pronunciation patterns, at both segmental and suprasegmental levels, which differ from pronunciation patterns found in the speech of native speakers (p.1010). Not only can these pronunciation patterns differ, they can also be similar in many ways. These similarities can be perceptual, acoustic and auditory, especially at the segmental level. This study investigates the acoustic and perceptual similarities between the fricatives /s/, /f/ and /ʃ/ of Effutu, a dialect of Awutu, and their English counterparts in the context of /a/ and /i/. Duration and spectral peak frequency are measured in order to achieve this. A discrimination task, Same-Different task, was administered to investigate listeners' perceived similarity (or difference) judgments between the pairs of fricatives. Preliminary findings show that there are perceptual and acoustic differences in the durations of these segments cross-linguistically. This study contributes to cross-linguistic investigation of fricatives, and to second language acquisition. The findings also show that the use of acoustic and perceptual cues helps to establish differences between speech sounds in different languages, and that, ESL teachers can use these to develop appropriate ways of teaching English sounds to learners.

2aSC27. Perception and production of second language sound inventory by English-speaking learners of Korean. Hanyong Park (Department of Linguistics, University of Wisconsin-Milwaukee, Curtin Hall 523, P.O. Box 413, Milwaukee, WI 53201, park27@uwm.edu)

This study investigates the perception of L2 sound inventory and its comparison with L2 production by eleven adult English-speaking novice learners of Korean in a classroom setting. We examined the perceptual identification and production accuracy of Korean consonants and vowels: eight monophthongs /i e ɛ ɨ u o ʌ ɑ/ both in isolation and following /p t k/ contexts, and fourteen consonants /p p' pʰ t t' tʰ s s' c c' cʰ k k' kʰ/ with /ɑ/ in word-initial position. Overall results indicated that most learners were better at production than perception. Such tendency was more apparent for consonants (except /s/) than vowels, for many learners exhibited a high performance in both perception and production of vowels. Results also showed that learners with more accurate production tended to exhibit more accurate perceptual identification. However, such observation applied only to vowels. Further, learners often had difficulty in both production and perception for the same vowels.

Findings suggest rate differences in L2 sound learning; learning takes longer in perception than in production, and in consonants than in vowels. Findings also suggest that production-perception link is stronger in L2 vowel development, at least in the case of English speaking learners of Korean.

2aSC28. Phonetic accommodation after passive exposure to native and nonnative speech. Midam Kim and Ann R. Bradlow (Linguistics, Northwestern University, 2016 Sheridan Road, Evanston, IL 60201, midamkim@gmail.com)

We investigated native English talkers' phonetic accommodation to a native or nonnative model talker in a passive auditory exposure setting. We performed a phonetic accommodation experiment, following the procedure of Goldinger & Azuma (2004). Specifically, the imitators read monosyllabic words, disyllabic words, and sentences before and after perceptual exposure to the stimuli. We found evidence of phonetic convergence both to native and nonnative model talkers from various acoustic measurements on words and sentences, and dynamic time warping analyses and XAB perception tests on sentences. We also found that dialect mismatch between participants and native model talkers inhibited phonetic convergence in some acoustic measurements. Additionally, the distances between model talkers and participants along the acoustic measurements before auditory exposure positively affected their degrees of phonetic convergence, regardless of the direction of the change; the farther the acoustic distance was before the auditory exposure, the larger the degree of phonetic convergence was. Moreover, the imitators generalized their accommodation patterns from exposed to unexposed items. Finally, XAB perception tests with the sentences revealed that imitators of all model talkers were perceived as converging towards their model talker, and importantly, this pattern of perceived accommodation was predicted by most of the sentence-based acoustic measurements.

2aSC29. Factors affecting the perception of foreign-accented speech by native and non-native listeners. Terrin N. Tamati (Linguistics, Indiana University, Bloomington, IN 47401, ttamati@indiana.edu)

Previous research has shown that several factors influence the perception of foreign-accented speech. Beyond talker-related factors, such as native language, length of residency, and age of acquisition, other factors, such as listener experience, listening context, and lexical characteristics, play an important role. To futher investigate these issues, the current study explored the perception of foreign-accented speech by native speakers of American English and Korean learners of English. In an accent rating task, listeners evaluated English sentences produced by native and non-native speakers (Korean and Mandarin) for strength of accent. Sentences contained three key words that varied by lexical frequency (high or low) and phonological neighborhood density (high or low). The same listeners also completed a sentence recognition task with a similar set of materials in which they listened to sentences and typed in the words they recognized. Results showed that lexical frequency and neighborhood density, overall, significantly influenced perceived accentedness and recognition accuracy for both groups. However, these effects were mediated by the native language of the talker and listener. These findings support previous research showing lexical frequency and density effects in the perception of foreign-accented speech and suggest that these effects may interact with talker and listener background.

2aSC30. Processing interactions between segmental and suprasegmental information in English and Mandarin Chinese. Mengxi Lin (Linguistics, Purdue University, West Lafayette, IN), Alexander L. Francis (Speech, Language & Hearing Sciences, Purdue University, SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, francisa@purdue.edu), Fernando Llanos (Spanish & Portuguese, Purdue University, West Lafayette, IN), Olga Dmitrieva (Linguistics, Stanford University, West Lafayette, Indiana), and Rachel Chapman (Speech, Language & Hearing Sciences, Purdue University, West Lafayette, IN)

In this study, a Garner selective attention task is used to identify cross-linguistic differences in attention to vowels, consonants and tones. In previous research, Tong et al. (2008) reported that, in Mandarin Chinese, consonantal and vocalic variability interfered more with tone processing than vice versa (asymmetric integrality), in contrast with the findings of earlier studies (Lee & Nusbaum, 1993; Repp & Lin, 1990). However, while these earlier studies examined both English and Mandarin Chinese listeners, Tong et al. (2008) did not study English speakers because of cross-linguistic differences in tone discrimination. The present study extends this work to examine interactions between these properties in English as well as Mandarin Chinese, using stimuli in which the consonantal, vocalic and tonal differences are linguistically meaningful in both languages, and normalizing for cross-linguistic differences in discriminability. It is predicted that Chinese results will replicate those of Tong, et al. (2008), while English listeners may show more symmetric integrality between segmental and tonal information than in previous studies since these pitch contours are prosodically meaningful and corrected for discriminability. Results will be discussed with respect to the role that linguistic knowledge plays in determining processing dependencies between segmental and suprasegmental information.

2aSC31. Delayed use of fundamental frequency (F0) rise in non-native speech segmentation. Caitlin E. Coughlin and Annie Tremblay (Linguistics, University of Kansas, 541 Lilac Lane Blake Hall, Lawrence, KS 60045-3129, atrembla.illinois@gmail.com)

Research has shown that second/foreign language (L2) speech segmentation is less efficient than native language (L1) segmentation, because L2 learners cannot suppress L1 segmentation routines. This study uses visual-world eye-tracking to determine whether English learners of French can learn to use F0 for locating word-final boundaries in French. F0 rise is often word-initial in English but often word-final in French. Native French speakers and English learners of French heard sentences where lexical competitors were created between the target noun and the following adjective (stimulus: chat grincheux 'cranky cat'; target: chat 'cat'; competitor: chagrin 'sorrow'). The target was either accented or unaccented, and the stimuli were either natural or resynthesized (swapped F0 between accented and unaccented targets). Participants selected the word they heard from given options (target, competitor, distracters), and fixations were recorded from target-word onset. Accuracy (word selection): learners, but not natives, were more accurate for target words with F0 rise than without it. Fixations: natives, but not learners, showed higher differential proportions of target and competitor fixations for target words with F0 rise than without it. Proficiency did not interact with the variables. This suggests that L2 learners' use of F0 rise is delayed compared to that of natives.

2aSC32. Cross-language assimilation of lexical tone. Jennifer Alexander and Yue Wang (Department of Linguistics, Simon Fraser University, Robert C Brown Hall Bldg, 8888 University Drive, Burnaby, BC V5A 1S6, Canada, jennifer_alexander@sfu.ca)

We extend to lexical-tone systems a model of second-language perception, the Perceptual Assimilation Model (PAM) (Best & Tyler, 2007), to examine whether/how native-language lexical-tone inventory composition influences perception of novel tone. Native listeners of Cantonese, Thai, and Mandarin perform a tone mapping-rating assimilation task. Listeners hear CV syllables bearing all tones of Cantonese, Thai, Mandarin, and Yoruba - languages with different tone inventories. They (1) map the tone they hear to the nearest native tone category, and (2) provide a goodness rating on a 5-point scale (5 = perfect). As predicted by the PAM, listeners assimilated non-native tones to the phonetically-closest native tone categories. Listeners attended primarily to pitch-contour, and secondarily to pitch-height, contrasts for the mappings. E.g., Mandarin listeners assimilated the Thai high "level" (phonetically mid-to-high-rising) tone to Mandarin rising tone 76% of the time, and to Mandarin high-level tone only 22% of the time. Also as predicted, all novel tones did not assimilate equally well to native categories; mappings received ratings between 2.9-4.1, averaging 3.5. The groups' different patterns of results indicate that novel-tone perception is influenced by experience with the native-language tone inventory, and that listeners attend to gradient phonetic detail to assimilate novel tones to native-tone categories. This work is supported by NSF grant 0965227 to J.A.

2aSC33. Effects of acoustic and linguistic aspects on Japanese pitch accent processing. Xianghua Wu, Saya Kawase, and Yue Wang (Linguistics, Simon Fraser University, 8888 University Drive, Burnaby, BC V5A1S6, Canada, xianghua_wu@sfu.ca)

This study investigates the hemispheric processing of Japanese pitch accent by native and non-native listeners. The non-natives differ in their first (L1) and second (L2) language experience with prosodic pitch, including Mandarin (tonal L1) and English (non-tonal L1) listeners with or without

Japanese learning experience. All listeners completed a dichotic listening test in which minimal pairs differing in pitch accent were presented. Overall, the results demonstrate a right hemisphere lateralization across groups, indicating holistic processing of temporal cues as the pitch accent patterns span across disyllabic domain. Moreover, the three pitch accent patterns reveal different degrees of hemispheric dominance, presumably attributable to the acoustic cues to each pattern which involve different hemispheric asymmetries. The results also reveal group difference, reflecting the effects of linguistic experience. Specifically, the English listeners with no Japanese background, compared to the other groups, exhibit greater variance in hemispheric dominance as a function of pitch accent difference, showing a greater reliance on acoustic cues when linguistic information is lacking. Together, the findings suggest an interplay of acoustic and linguistic aspects in the processing of Japanese pitch accent but showing a more prominent acoustic influence. [Research supported by NSERC.]

2aSC34. Tonal adaptation of English loanwords in Mandarin: The role of perception and factors of characters. Li-Ya Mar and Hanyong Park (Department of Linguistics, University of Wisconsin-Milwaukee, 4810 Marathon Dr., Madison, WI 53705, liyamar@uwm.edu)

The present study investigates the role of orthography and perceptual similarity between the English stress and the Mandarin tone during the borrowing process of English words among Taiwanese Mandarin speakers. We had 7 Mandarin speakers transliterate 40 unfamiliar disyllabic US city names using Chinese characters. Based on the results, we created 28 stimuli consisting of possible Chinese borrowings and the target English word for AXB identification tasks. Then, we had the subjects choose the more similar Chinese form to an English target word after the auditory presentation of the stimuli, first without and second with written representations of the stimuli on different days. The transliteration results indicate that a stressed syllable is usually adapted with tones with a high pitch. The AXB task results show a character frequency and a semantics override perceptual similarity; when the adapted forms include characters used infrequently or with negative meanings, another form is chosen despite the fact that it is not the most perceptually similar form. The findings suggest that perceptual similarity mapping takes place first and other factors, such as semantics or the character frequency, come into play when the output contains an infrequently-used or semantically-negative character in tonal adaptation.

2aSC35. Orthography modulates lexical recognition in a second language. Christine E. Shea (Spanish and Portuguese, University of Iowa, 412 Phillips Hall, Iowa City, IA 52242, cesshea@yahoo.com)

We use a cross-modal masked priming paradigm to investigate a) whether orthography is always activated during lexical recognition and b) when activated, whether orthography influences the perception of allophonic variants by adult L2 learners. L1 Spanish and L2 Spanish learners (n=60) were exposed to written Spanish primes with ‘b’ ‘d’ or ‘g’ in intervocalic position. In Spanish, the positional phones corresponding to these orthographic symbols are voiced fricatives [β ð ɣ]; in English they are voiced plosives. In the matched prime trials, written primes were paired to auditory targets with the expected voiced fricative (lado ‘[laðo] ‘side’). For the unmatched prime trials, the auditory target had medial plosives ([lado]. Orthographic prime durations were either 33ms (implicit, Condition 1) or 67ms (explicit, Condition 2). Accuracy and reaction times were registered on lexical decision to the auditory target. Preliminary RT results indicate a three-way interaction among group, trial type (matching or unmatching) and prime condition. Follow-up tests revealed a significant difference for the L2 listeners for prime conditions: significantly longer RTs were registered for the ‘matching’ trials when the orthographic prime was visible (Condition 2). These results suggest that L2 lexical recognition is modulated by orthographic information when it is explicitly available.

TUESDAY MORNING, 23 OCTOBER 2012 BENNIE MOTEN A/B, 8:15 A.M. TO 10:45 A.M.

Session 2aSP

Signal Processing in Acoustics: Methods for Underwater Acoustic Parameter Estimation and Tracking at Low Signal-to-Noise Ratios

Paul J. Gendron, Chair
Maritime Systems Div., SSC Pacific, San Diego, CA 92152

Chair’s Introduction—8:15

Invited Papers

8:20

2aSP1. Information-based performance measures for model-based estimation. Edmund J. Sullivan (Prometheus Inc., 46 Lawton brook Lane, Portsmouth, RI 02871, ejsul@fastmail.fm)

Classical estimation is conventionally evaluated via the Cramer-Rao Lower Bound on the estimate. When prior information is available, Bayesian estimation can be used if this information is available in statistical form. However, when the prior information is in the form of a physical model, such as in a tracking scheme, it is not clear how much improvement will be provided, since it is not in statistical form. Here it is shown how this problem can be dealt with using the Fisher information matrix by introducing the model into a Kalman estimator. Since the state error covariance provided by a steady-state Kalman estimator is the inverse of the Fisher matrix, it directly provides a statistical measure of the information provided by the model. Then by relating the Fisher information matrix to the Kullback-Liebler distance, it is shown how the Fisher matrix is scaled to provide its information in bits. The model can then be evaluated as to how much information it provides to the estimator. An example using a moving towed array as a bearing estimator will be presented. It will be quantitatively shown that inclusion of the array motion in the estimator will improve the estimation performance

8:40

2aSP2. A physical statistical clutter model for active sonar scenarios with variable signal-to-noise ratios. Roger C. Gauss and Joseph M. Fialkowski (Acoustics Division, Naval Research Laboratory, 4555 Overlook Ave., S.W., Washington, DC 20375-5350, roger.gauss@nrl.navy.mil)

Active sonar classification algorithms need to be robust in preventing operator overload while not being misled by false targets. This talk describes a new 3-parameter statistical sonar clutter model that not only provides a physical context for relating the characteristics of normalized matched-filter echo-data distributions to scatterer attributes, but scatterer information that is largely independent of its peak signal-to-noise ratio (SNR) value. It extends our 2-parameter Poisson-Rayleigh model (Fialkowski and Gauss, IEEE JOE, 2010) by adding a quantitative measure of scatterer spatial dispersion to its measures of scatterer density and relative strength. Maximum likelihood estimates of the clutter model's 3 parameters were derived from mid-frequency (1-5 kHz) shallow-water active sonar data containing returns from biologic, geologic and anthropogenic objects with differing spatial and scattering characteristics. The resulting clutter model's probability density functions not only fit the non-Rayleigh data well while displaying an insensitivity to SNR, but the dispersion parameter values were consistent with the known spatial characteristics of the scatterers and the values' ping-to-ping variance correlated strongly with clutter object class, all of which are encouraging with regard to developing robust physics-based active classification algorithms. [Work supported by ONR.]

9:00

2aSP3. Least squares channel estimation and adaptive equalization at low signal to noise ratios. James C. Preisig (AOPE, WHOI, MS #11, Woods Hole, MA 02540, jpreisig@whoi.edu)

Least squares based adaptive algorithms are among the most commonly used techniques for both channel estimation and adaptive equalization using signals that have propagated through underwater acoustic channels. Such channels are often characterized by long delay spreads meaning that the impulse response of the channel contains many "taps" or "weights" to be estimated or accommodated. In addition, the channel is often time varying which limits the duration of the averaging window that can be used by the algorithm's adaptation process. At low SNRs, these two factors pose a significant challenge to algorithm performance. This challenge is particularly severe in the context of adaptive equalization where the use of multichannel equalizers is often required to achieve reliable performance and the traditional approach of joint optimization of the feedforward filter tap weights on all receiver channels results in large dimensional optimization problems. This talk will contrast and compare the impact of low SNR on least squares based channel estimation and adaptive equalization algorithms. The role of dimensionality reduction will be examined and the response of multichannel equalizers to different signal and noise environments as well under different equalizer configurations will be examined.

9:20

2aSP4. Pulse compression in striation processing—Acoustic invariant as seen in the time domain. Paul Hursky (HLS Research Inc, 3366 North Torrey Pines Court, Suite 310, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

The acoustic invariant is well known to produce broadband interference or striation patterns in spectrograms. These have been used for a variety of applications, including geo-acoustic inversion and target tracking, in both passive and active settings. The processing to extract parameters from these broadband interference patterns has typically been performed on the spectrograms. However, spectrogram striations have energy that is spread across a wide band. This paper will present a time domain approach, in which the striations are pulse-compressed via a correlation process, before extracting the parameters of interest. Narrowband signals pose an interesting conundrum for striation processing - they are typically much stronger iin level than the underlying broadband interference pattern, and can be mistaken for striations, thus corrupting the parameter extraction. As in time delay estimation, pre-whitening is needed to suppress narrowband components. At the same time, the narrowband components best reveal the underlying interference pattern, albeit in a very narrow band, because they are of such high SNR. We will discuss how to exploit this information content as well.

Contributed Papers

9:40

2aSP5. Exploiting differences in underwater acoustic signal and noise distributions to improve signal detection in low signal-to-noise ratio. Andrew T. Pyzdek, R. L. Culver, and Brett E. Bissinger (Applied Research Laboratory, The Pennsylvania State University, PO Box 30, State College, PA 16804, atp5120@psu.edu)

Traditional models for acoustic signals and noise in underwater detection utilize assumptions about the underlying distributions of these quantities to make algorithms more analytically and computationally tractable. Easily estimated properties of the signal, like the mean amplitude or power, are then calculated and used to form predictions about the presence or absence of these signals. While appropriate for high SNR, quantities like the mean amplitude may not give reliable detection for SNR at or below 0 dB. Fluctuation based processors, utilizing additional statistics of received pressure, offer an alternative form of detection when features of the received signal beyond changes in mean amplitude are appreciably altered by the presence of a signal. An overview of fluctuation based processing will be given, with a focus on the underlying statistical phenomena that grant this method efficacy. Work sponsored by the Office of Naval Research in Undersea Signal Processing.

9:55–10:15 Break

10:15

2aSP6. Coherent processing of shipping noise for ocean monitoring. Shane W. Lani (School of Mech. Eng., Georgia Institute of Technology, Atlanta, GA 30332, lani.shane@gatech.edu), Karim G. Sabra (School of Mech. Eng., Georgia Institute of Technology, Atlanta, GA), Philippe Roux (Institut des Sciences de la Terre, Université Joseph Fourier, Grenoble, None, France), William Kuperman, and William Hodgkiss (University of California, San Diego, CA)

Extracting coherent wavefronts between passive receivers using cross-correlations of ambient noise may provide a means for ocean monitoring without conventional active sources. Hence applying this technique to continuous ambient noise recordings provided by existing or future ocean observing systems may contribute to the development of long-term ocean monitoring applications such as passive acoustic thermometry. To this end, we investigated the emergence rate of coherent wavefronts over 6 days using low-frequency ambient noise ($f < 1.5$ kHz) recorded on two vertical line arrays-separated by 500m- deployed off San-Diego CA in ~200m deep water. The recorded ambient noise was dominated by nonstationary distributed shipping activity with

the frequent occurrence of loud isolated ships. Noise data were first processed to mitigate the influence of these loud shipping events in order to primarily emphasize the more homogenous and continuous background ambient noise in the frequency band. Furthermore, the coherent noise field propagating between the VLAs was beamformed using spatio-temporal filters to enhance the emergence rate of specific coherent wavefronts. This presentation will discuss various strategies for the selection of these spatio-temporal filters (either data-derived or model-based) in order to improve the continuous tracking of these coherent wavefronts over 6 days.

10:30

2aSP7. Estimation of a broadband response with dilation process compensation at very low signal to noise ratios. Paul J. Gendron (Maritime Systems Division, SSC Pacific, A460, Bldg. 1, Bayside Campus, 53560 Hull St., San Diego, CA 92152, paul.gendron@navy.mil)

Challenges of estimating broadband acoustic response functions at low signal to noise ratio (SNR) are due to both their varying sparsity and the varying spatio-temporal dynamics of each acoustic arrival. Acoustic responses can be quite sparse over the delay-Doppler-angle domain exhibiting large regions that are relatively quiet. The arrivals may share significant Doppler processes due to platform motion or may be driven independently by boundary interactions. Because of this estimation must be adaptive across delay-Doppler and angle with any single fixed estimator inadequate. One means of constructing such an estimator is to view each angle-Doppler-frequency slot as either ensonified or not. A mixture model can be employed for this purpose to describe the behavior of the acoustic response over received signal duration, aperture, and bandwidth. The posterior mean is derived and shown to be soft shrinkage operator of the conventional Wiener filtered coefficients under each of the components of the mixture. This estimator can be employed for bulk dilation estimation as an alternative to a phase locked loop. The posterior variance is derived and compared conventional Wiener filtering. The resulting adaptive structure is applied to M-ary orthogonal signaling sets taken in diverse shallow water environments at very low SNR. This work was supported by the Naval Innovative Science and Engineering Program and the Office of Naval Research.

TUESDAY MORNING, 23 OCTOBER 2012 MARY LOU WILLIAMS A/B, 8:00 A.M. TO 11:45 A.M.

Session 2aUW

Underwater Acoustics and Acoustical Oceanography: Propagation Topics

Ralph A. Stephen, Chair
Woods Hole Oceanographic Institution, Woods Hole, MA 02543-1592

Contributed Papers

8:00

2aUW1. Nonlinear acoustic pulse propagation in range-dependent underwater environments. Joseph T. Maestas (Mechanical Engineering, Colorado School of Mines, 1500 Illinois Street, Golden, CO 80401, jmaestas@mines.edu) and Jon M. Collis (Applied Mathematics and Statistics, Colorado School of Mines, Golden, CO)

The nonlinear progressive wave equation (NPE) is a time-domain formulation of Euler's fluid equations designed to model low angle wave propagation using a wave-following computational domain [B. E. McDonald et al., JASA 81]. The wave-following frame of reference permits the simulation of long-range propagation that is useful in modeling the effects of blast waves in the ocean waveguide. The standard formulation consists of four separate mathematical quantities that physically represent refraction, nonlinear steepening, radial spreading, and diffraction. The latter two of these effects are linear whereas the steepening and refraction are nonlinear. This formulation recasts pressure, density, and velocity into a single variable, a dimensionless pressure perturbation, which allows for greater efficiency in calculations. Nonlinear effects such as weak shock formation are accurately captured with the NPE. The numerical implementation is a combination of two numerical schemes: a finite-difference Crank-Nicholson algorithm for the linear terms of the NPE and a flux-corrected transport algorithm for the nonlinear terms. While robust, solutions are not available for sloping seafloors. In this work, range-dependent environments, characterized by sloping bathymetry, are investigated and benchmarked using a rotated coordinate system approach.

8:15

2aUW2. A comprehensive study of the Bellhop algorithm for underwater acoustic channel modelings. Xiaopeng Huang (Dept. of Electrical and Computer Engineering, Stevens Institute of Technology, Castle Point on Hudson, Hoboken, NJ 07030, xiaopeng.huang0508@gmail.com)

Ray tracing is one of the most conventional methods for modeling underwater acoustic sound propagation, and the Bellhop algorithm is a highly effcient ray tracing program, written by Michael Porter as part of the Acoustic Toolbox. In this abstract, based on the introduced Bellhop algorithm, we select several typical underwater acoustic environments so as to study how to model their channels and analyze their channel properties. Simulation results will investigate the following aspects of channel modeling and perperties: ray tracing, eigen-ray tracing, coherent transmission loss, channel impulse reposne, coherence time. In addition, simulation results will compare the performance difference with variant environmental parameters, such as sound speed anomaly and wavy surface.

8:30

2aUW3. Information content of an acoustic field propagating in an ocean waveguide. Steven Finette (Acoustics Division, Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

It is intuitively clear that, in some sense, waves carry "information" concerning both their source characteristics and their interaction with boundaries and/or sound speed inhomogeneity in the propagation path. This presentation addresses the issue of how one can estimate the maximum rate that an acoustic field can transfer information in an ocean waveguide, based on the properties of wave propagation. Information theory is the natural framework for addressing this question, relating wave propagation and communication concepts; it is applied here for the example of a Pekeris waveguide. Using properties of the propagation operator, information-theoretic arguments applied to the propagated field allow for the transfer of information along independent communication channels in the waveguide and an explicit expression for the channel capacity is obtained. The latter represents an upper bound on the error-free transfer of information from a source to a point in the waveguide by use of the propagated field. Work supported by the Office of Naval Research.

8:45

2aUW4. Empirical and collocation-point methods for estimation of acoustic field and array response probability density functions. Thomas J. Hayward and Roger M. Oba (Naval Research Laboratory, 4555 Overlook Ave SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Recent research has investigated the representation of acoustic field uncertainties arising from uncertainties of the acoustic environment, with emphasis placed either on the representation of the acoustic field as a random process or on estimation of the acoustic field probability density function (pdf) at a given receiver location. The present work introduces two methods for estimating acoustic field and array response pdfs. The first method is based on the computation of the empirical characteristic function (ECF) of the acoustic field, derived from a random sample in the acoustic parameter space and computation of the corresponding acoustic fields. The acoustic field pdf estimate is obtained as the Fourier transform of the ECF. The second method is based on approximation of the characteristic function using collocation-point methods, which are based on orthogonal-polynomial approximations of the mapping from the parameter space to the acoustic field. Both methods are investigated in two examples: (1) a stratified shallow-water model with two-dimensional uncertainties of sound speed and attenuation coefficient; and (2) a shallow-water model with sound-speed fluctuations in the water column defined by a time-stationary internal wave field. Sampling requirements and convergence of the pdf estimates are investigated for both methods and compared. [Work supported by ONR.]

9:00

2aUW5. Modeling uncertain source depth in range-dependent environments. Kevin R. James and David R. Dowling (Mechanical Engineering, University of Michigan, Mechanical Engineering, Ann Arbor, MI 48105, drd@umich.edu)

Efficient and accurate estimation of the uncertainty in a transmission loss calculation is important for tactical applications of underwater acoustic propagation calculations. Uncertainty in source depth can contribute significantly to the overall transmission loss uncertainty. The unique relationship between source depth and transmission loss motivates a different approach to uncertainty estimation than that used for other environmental and sound-channel parameters. Prior research has shown that in a range-independent environment, source depth uncertainty can be efficiently modeled using the principles of reciprocity. This presentation describes a new approach to uncertainty estimation in range-dependent environments, based on the assumption that the relationship between source depth and transmission loss is approximately governed by the adiabatic approximation on a local scale. Transmission loss predictions are taken from RAMGEO results to solve for the unknowns in the resulting approximate formulation. By modeling the relationship between source depth and transmission loss, approximate uncertainty bounds can be generated for transmission loss predictions. Results are provided for simple up-sloping and down-sloping range-dependent environments, for frequencies from 100 Hz to several kHz, and for ranges of several kilometers. [Sponsored by the Office of Naval Research, Code 322OA.]

9:15

2aUW6. Mode coupling due to bathymetric variation. Charles E. White, Cathy A. Clark (Naval Undersea Warfare Center, 1176 Howell Street, Newport, RI 02841, charlie.e.white@navy.mil), and Gopu Potty (Ocean Engineering, University of Rhode Island, Narragansett, RI)

In shallow water the assumption of range independence fails in conditions of rapidly-varying bathymetry and/or horizontal sound speed. In these environments, the modes of vibration of the acoustic wave equation become coupled, with a transfer of energy between adjacent modes occurring upon traversing a horizontal change of environment. In this talk, we will consider some simple applications of mode conversions due to variable bathymetry. Results will be compared to closed form propagation solutions in constant-slope wedge environments. The ultimate goal of this research is the development of a fully non-adiabatic range-dependent mode solution which retains analytical integrity while executing in a time window that is tactically useful for warfare applications.

9:30

2aUW7. Transport theory applied to shallow water acoustics: The relative importance of surface scattering and linear internal waves. Kaustubha Raghukumar and John A. Colosi (Naval Postgraduate School, 833 Dyer Rd, Monterey, CA 93943, kraghuku@nps.edu)

Acoustic fields in shallow water have a statistical nature due to complex, time-evolving sound speed fields and scattering from rough boundaries. A coupled-mode transport theory [Creamer (1996), Colosi and Morozov (2009)] allows for the prediction of acoustic field second moments like mean intensity and coherence. This was previously applied to study low frequency acoustic fluctuations in an environment typical of that of the Shallow Water 2006 (SW06) experiment on the New Jersey Continental shelf. Here the propagation was found to be strongly adiabatic and random sound speed fluctuations from internal waves radically altered acoustic interactions with intense nonlinear internal wave packets. Here, we extend the SW06 study to examine the ability of transport equations to describe high frequency (>1 kHz) sound in shallow water. Mode coupling rates from internal waves are expected to be larger, and scattering effects from rough surfaces need treatment. The aforementioned transport theory is merged with the rough surface scattering transport theory of Thorsos et al (2009). Oceanographic and sea surface measurements are used to constrain the internal wave and sea surface models. The relative importance of linear internal waves and surface scattering effects are studied using transport theory and Monte Carlo simulations.

9:45

2aUW8. Theory of the sound field fluctuations in the presence of internal waves due to adiabatic mechanism of interaction. Boris Katsnelson (Dept. of Physics, Voronezh State Univ., 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru), Mohsen Badiey (College of Earth, Ocean and Environment, University of Delaware, Newark, DE), and Alexander Tckhoidze (Dept. of Physics, Voronezh State Univ., Haifa, Haifa, Israel)

In the presence of moving nonlinear internal waves character of interaction between sound field and internal waves depends on angle between direction of an acoustic track and wave front of internal waves (mode coupling, horizontal refraction or adiabatic regime) [JASA, vol. (122), pp. 747-760, 2007]. In particular, if this angle is about 15-20 degrees there should be adiabatic mechanism. Adiabatic regime of propagation means that variations of the sound field follow variations of the sound speed profile. In this paper similar situation is considered when wave front of the train of internal waves crosses the acoustic track at the angle about 15 degrees. Theoretical modeling shows specific features of adiabatic fluctuations: variations of shape of waveguide modes, fluctuations of amplitudes (excitation coefficients) of the corresponding modes and phase fluctuations. These fluctuations can be separated in time on dependence on position of the train. Results of modeling are compared with experimental data [shown in accompanying paper, Badiey et al.] and are in good agreement. This work was supported by ONR and RFBR.

10:00–10:15 Break

10:15

2aUW9. Acoustic frequency shifts due to internal tides and nonlinear internal waves. Altan Turgut and Peter C. Mignerey (Acoustics Div., Naval Research Lab, Acoustics Div., Washington, DC 20375, altan.turgut@nrl.navy.mil)

Significant frequency shifts of acoustic intensity level curves in broadband signal spectrograms were measured in the East China Sea during the summer of 2008. Broadband pulses at 270-330 Hz were transmitted from a fixed source and received at a bottomed horizontal array, located at 33 km range. The acoustic intensity level curves of the received signals indicate regular frequency shifts that are well correlated with the measured internal

tides and nonlinear internal waves. Regular frequency shifts due to nonlinear internal waves are observed only when their wave-fronts are nearly parallel to the acoustic propagation path, causing an effective change in the sound speed profile. Similar effects were observed in 3-D numerical simulation results when curved nonlinear internal wave fronts are used. These observations and simulations indicate the potential of monitoring internal tides and nonlinear internal waves using low-frequency acoustic signals when the acoustic source and receiver are strategically placed. [Work supported by the Office of Naval Research.]

10:30

2aUW10. Range dependent acoustic intensity scintillations due to focusing, defocusing, and scattering by sea swell and bottom sediment waves. Alexey A. Shmelev (WesternGeco, Schlumberger, Houston, TX 77057, alexey.a.shmelev@gmail.com), James F. Lynch, Ying-Tsong Lin, Arthur E. Newhall, and Timothy F. Duda (AOPE, Woods Hole Oceanographic Institution, Woods Hole, MA)

It is known that the waveguide depth variability causes horizontal refraction and coupling of acoustic normal modes. Presence of large bottom sediment waves and sea swell are examples of strongly anisotropic waveguides that result in range dependence of the acoustic scintillation index. In the directions parallel to the wave crests, three-dimensional effects of mutual horizontal focusing, defocusing and diffusion between such waves are the main mechanisms of intensity fluctuations. For acoustic propagation in the perpendicular to the wave crests directions, intensity fluctuations are mainly driven by random mode coupling and scattering. Analytical studies and numerical examples of the acoustic scintillation index, as well as its azimuthal and range dependence in the shallow water with both types of waves, will be provided. Directions for future studies will be discussed.

10:45

2aUW11. Deep seafloor arrivals in long range ocean acoustic propagation. Ralph A. Stephen, S. Thompson Bolmer, Matthew A. Dzieciuch (Geol & Geophys, WHOI, 360 Woods Hole Rd, Woods Hole, MA 02543-1592, rstephen@whoi.edu), Peter F. Worcester (IGPP, Scripps Institution of Oceanography, La Jolla, CA), Rex K. Andrew, James A. Mercer (Applied Physics Laboratory, University of Washington, Seattle, WA), John A. Colosi (Oceanography, Naval Postgraduate School, Seattle, WA), and Bruce M. Howe (Ocean and Resources Engineering, University of Hawaii at Manoa, Honolulu, HI)

Ocean bottom seismometer observations during the long-range ocean acoustic propagation experiment in the North Pacific in 2004 showed robust, coherent, late arrivals that were not observed on hydrophones suspended 750m and more above the seafloor and that were not readily explained by ocean acoustic propagation models. The DSFA arrival pattern on the OBSs near 5000m depth are a delayed replica, by about two seconds, of the arrival pattern on the deepest element of the DVLA at 4250m depth (DVLA-4250). Using a conversion factor from the seafloor vertical particle velocity to seafloor acoustic pressure, we have quantitatively compared signal and noise levels at the OBSs and DVLA-4250. Ambient noise and DSFA signal levels at the OBSs are so quiet that if the DSFA arrivals were propagating through the water column, perhaps on an out-of-plane bottom-diffracted-surface-reflected (BDSR) path, they would not appear on single, unprocessed DVLA channels. Nonetheless arrival time and horizontal phase velocity analysis rules out BDSR paths as a mechanism for DSFAs. Whatever the mechanism, the measured DSFAs demonstrate that acoustic signals and noise from distant sources can appear with significant strength on the seafloor at depths well below the conjugate depth.

11:00

2aUW12. Effects of fine-scale topographical change on mid-frequency bottom loss. Jie Yang (Applied Physics Lab, University of Washington, 1013 NE 40th St, Seattle, WA 98105, jieyang@apl.washington.edu) and Dajun Tang (Applied Physics Lab, University of Washington, Seattle, WA)

It was shown previously [Yang et al, J. Acoust. Soc. Am. 131(2), 1711-1721 (2012)] that forward scatter from topographical changes could alter bottom loss at mid-frequencies. In this more detailed study, fine-scale bottom bathymetry data from a multibeam survey are used as inputs to numerical experiments to investigate the effects of topographical variation on bottom loss (BL). Bottom reflection/forward scatter simulations in the frequency band of 2–5 kHz are carried out using several numerical methods which all include the effect of bathymetry variation. Bottom bounces including forward scatter are treated as if the bottom is flat and are used to estimate BL at different frequencies. It is found that small topographic changes can result in large deviations in BL estimates. Remedies for the effect of bottom topography change on BL are suggested. [Work supported by ONR.]

11:15

2aUW13. Modeling the effect of interface roughness on bottom loss from layered interfaces with finite elements. Marcia J. Isakson and Nicholas P. Chotiros (Applied Research Laboratories, University of Texas, 10000 Burnet Road, Austin, TX 78713, misakson@arlut.utexas.edu)

The bottom loss from a layered ocean sediment is determined using a finite element/boundary element (FE/BE) method. First, the pressure and its normal derivative are calculated on the top interface using finite elements. Then the field at a point outside of the domain is determined using the Helmholtz/Kirchhoff integral (BE). Bottom loss is then calculated by comparing the reflected/scattered energy to the incident energy. The finite element method makes no approximations to the Helmholtz equation and is exact within the limits of the discretization. Any number of layers including elastic layers with rough or smooth interfaces can be included. The results of the FE/BE approach will be compared to Geoacoustic Bottom Interaction Model (GABIM) for a number of test cases. [Jackson, et al., IEEE J. Ocean. Eng. 35(3), 603-617 (2010)] GABIM computes the layered reflection coefficient then includes scattering for one rough interface based on "a combination of the Kirchhoff approximation, first-order perturbation theory and an empirical expression for very rough seafloors". Lastly, the bottom loss of multiple rough interfaces will be compared to that of a single rough interface. [Work supported by ONR, Ocean Acoustics.]

11:30

2aUW14. Jurassic acoustics: Low frequency sound absorption in the ocean during past ages. David Browning (Physics Department, URI, Kingston, RI 02881, decibeldb@aol.com) and Peter M. Scheifele (Communication Sciences Dept., University of Cincinnati, Cincinnati, OH)

A major aspect of global warming is ocean acidification. To provide a baseline for future change, investigators have been able to track the geological record of ocean acidification back to 300 million years ago (mya). One of the key factors is tracing the history of boron isotopes. The principal low frequency sound absorption mechanism in seawater is a boron reaction which is pH dependent (the lower the pH, the lower the absorption), so this geological record can be used to estimate sound absorption in the ocean all the way back to the carboniferus period. The broad picture is that low frequency absorption in the ocean decreased from 300 mya to 200 mya, was relatively constant from 200 mya to 100 mya, and then has been increasing since. The present level is back to one similar to that 300 mya. Future global warming may reverse this trend and cause the absorption to decrease down to a level similar to when the dinosaurs roamed (100 mya).

Meeting of the Standards Committee Plenary Group
to be held jointly with the meetings of the
ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:
ISO/TC 43, Acoustics,
ISO/TC 43/SC 1, Noise,
ISO/TC 43/SC 3, Underwater acoustics
ISO/TC 108, Mechanical vibration, shock and condition monitoring,
ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures,
ISO/TC 108/SC 3, Use and calibration of vibration and shock measuring instruments,
ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machines,
and
IEC/TC 29, Electroacoustics

P.D. Schomer, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43 Acoustics and ISO/TC 43/SC 1 Noise
Schomer and Associates, 2117 Robert Drive, Champaign, Illinois 61821

M.A. Bahtiarian, Acting Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC 3 Underwater acoustics
Noise Control Engineering, Inc., 799 Middlesex Turnpike, Billerica, MA 01821

D.J. Evans, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108 Mechanical vibration, shock and condition monitoring, and ISO/TC 108/SC 3 Use and calibration of vibration and shock measuring devices
National Institute of Standards and Technology (NIST), 100 Bureau Drive, Stop 8220, Gaithersburg, MD 20899

W.C. Foiles, Co-Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
BP America, 501 Westlake Park Boulevard, Houston, TX 77079

R. Taddeo, Co-Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
NAVSEA, 1333 Isaac Hull Avenue, SE, Washington Navy Yard, Washington, DC 20376

D.D. Reynolds, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock
3939 Briar Crest Court, Las Vegas, Nevada 89120

D.J. Vendittis, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
701 Northeast Harbour Terrace, Boca Raton, FL 33431

R. Taddeo, Vice Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
NAVSEA, 1333 Isaac Hull Avenue, SE, Washington Navy Yard, Washington, DC 20376

V. Nedzelnitsky, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
National Institute of Standards and Technology (NIST), Sound Building, Room A147, 100 Bureau Drive, Stop 8221, Gaithersburg, MD 20899-8221

The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting.

The meeting of the Standards Committee Plenary Group will precede the meetings of the Accredited Standards Committees S1, S2, S3, S3/SC 1, and S12, which are scheduled to take place in the following sequence:

Tuesday, 23 October 2012	10:30 a.m.–11:30 a.m.	ASC S12, Noise
Tuesday, 23 October 2012	1:15 p.m.–2:15 p.m.	ASC S1, Acoustics
Tuesday, 23 October 2012	2:30 p.m.–3:45 p.m.	ASC S3, Bioacoustics
Tuesday, 23 October 2012	4:00 p.m.–5:00 p.m.	ASC S3/SC 1, Animal Bioacoustics
Wednesday, 24 October 2012	8:30 a.m.–9:45 a.m.	ASC S2, Mechanical Vibration & Shock

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3 and S12 are as follows:

U.S. TAG Chair/Vice Chair	**TC or SC**	**U.S. Parallel Committee**
ISO		
P.D. Schomer, Chair	**ISO/TC 43** Acoustics	ASC S1 and ASC S3
P.D. Schomer, Chair	**ISO/TC 43/SC 1** Noise	ASC S12
M.A. Bahtiarian, Acting Chair	**ISO/TC 43/SC 3** Underwater acoustics	ASC S1, ASC S3/SC 1 and ASC S12
D.J. Evans, Chair	**ISO/TC 108** Mechanical vibration, shock and condition monitoring	ASC S2
W.C. Foiles, Co-Chair R. Taddeo, Co-Chair	**ISO/TC 108/SC 2** Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures	ASC S2
D.J. Evans, Chair	**ISO/TC 108/SC 3** Use and calibration of vibration and shock measuring instruments	ASC S2
D.D. Reynolds, Chair	**ISO/TC 108/SC 4** Human exposure to mechanical vibration and shock	ASC S3
D.J. Vendittis, Chair R. Taddeo, Vice Chair	**ISO/TC 108/SC 5** Condition monitoring and diagnostics of machines	ASC S2
IEC		
V. Nedzelnitsky, U.S. TA	**IEC/TC 29** Electroacoustics	ASC S1 and ASC S3

TUESDAY MORNING, 23 OCTOBER 2012 — TRIANON E, 10:30 A.M. TO 11:30 A.M.

Meeting of Accredited Standards Committee (ASC) S12 Noise

W. J. Murphy, Chair, ASC S12

NIOSH, 4676 Columbia Parkway, Mail Stop C27, Cincinnati, OH 45226

S.J. Lind, Vice Chair, ASC S12

The Trane Co., 3600 Pammel Creek Road, Bldg. 12-1, La Crosse WI 54601-7599

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note—that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 23 October 2012.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control, including biological safety, tolerance and comfort, and physical acoustics as related to environmental and occupational noise.

Session 2pAA

Architectural Acoustics: Coordination of Architectural and Sound System Design in the Built Environment

Kenneth Roy, Cochair
Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604

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

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. Case studies of a method to integrate architectural acoustic and sound systems design. Gary W. Siebein and Hyun Paek (Siebein Associates, Inc., 625 NW 60th Street, Suite C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

A method to evaluate the integrated architectural acoustic and reinforced sound systems for medium to large size worship and performance spaces using computer modeling of the systems for new facilities and diagnostic impulse response based measurements in existing rooms is presented. Four case studies of varying degrees of design integration of architectural acoustic and sound systems are presented to illustrate the method. Case study 1 is a divisible multi-purpose worship space. The sound system equipment and design was done without considering the acoustical design of the room. It was decided to re-use the existing equipment and adjust the aiming and programming of the system to fit the acoustics and architecture of the room to optimize the balance of natural and reinforced sounds. The second case study covers the design of a theater with a distributed array system to reinforce theatrical and musical sounds on stage. The third case study covers the installation of a new sound system in a large worship space with an organ and a very long reverberation time. The fourth case study covers the diagnostics, architectural and sound system improvements in a unique space for symposia on a college campus.

1:25

2pAA2. Naturally you need a sound system. Rogers Dixson (Cape Dixson Associates, Inc., 4279 Roswell Rd., NE, Ste 102-135, Atlanta, GA 30342, rdixson@cdai.com)

Ignoring Room Acoustics in Audio System design can be just as problematic as ignoring the audio system in designing the acoustics of a space. Contrary to marketing claims by product manufacturers on either side of the design "aisle," neither system can overcome all of the shortcomings imposed by the poor design of the other. Both elements have to be designed in coordination with each other (as well as in coordination with other systems in the built environment). This can become particularly challenging in multipurpose performance and presentation spaces as there can be multiple audio systems installed in the same space. Sound Reinforcement, Program Playback (particularly multi-channel surround sound) systems are increasingly being installed in performing arts facilities where natural acoustics is a critical element. How these aspects should (and can) be coordinated is discussed in this paper.

Contributed Papers

1:45

2pAA3. Seamless integration of audio visual design into architecture for more successful projects. Felicia Doggett (Metropolitan Acoustics, LLC, 40 W. Evergreen Ave., Suite 108, Philadelphia, PA 19118, felicia@metropolitanacoustics.com)

The integration of audio visual systems into technology-enabled buildings has long been a struggle with the architectural community. In some instances, architects do not want any part of the systems to be visible within the space, like speakers, projectors and equipment racks. In other cases, like digital signage and way-finding technology, the systems are front and center in the room and must be as visible and as user-accessible as possible. Audio visual technology is present in some form in almost every building being constructed today including educational, corporate, houses of worship, legislative, hospitality, sports arenas, healthcare, museums, and retail. Much time is spent not just on the system design, but working with architects on integrating the technology into these built environments, each of which has its own requirements. This presentation focuses on various ways to integrate audio visual technology into buildings for more seamless and successful projects. Balancing the functionality and performance of any audio visual system installation with the aesthetic impact on the space takes experience, creativeness, and a willingness to coordinate with the design team.

2:00

2pAA4. Sound system installed in a central bus station. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

As a result of a new administration board planning in the Central long range Bus Station located in a mayor city, it was installed a public address system, together with a small modification of the visible building materials within the station, for the purpose of easy understanding of; bus departure

2p TUE. PM

confirmation, people and stuff localization and general information to passengers, workers and visitors of the station. The system involved a distributed sound system with dozens of loudspeakers strategically located with reasonable, non-disturbing, sound pressure level generation by each one, which enhanced the intelligibility of the spoken messages well over the average for the Central Bus Stations system in the country. The system performed at a comfortable sound level, just enough to overcome the noise level within the central, which together with a reduced reverberation time allowed for proper effortless understanding of the emitted messages. Mayor contributors for the obtained results were careful design, sound equipment quality, sound absorption materials, guided installation and luck.

2:15

2pAA5. Active acoustics in a restaurant: A case study. Pierre Germain and Roger W. Schwenke (Meyer Sound Laboratories, 2832 San Pablo Ave, Berkeley, CA 94702, rogers@meyersound.com)

Loud restaurants have become sufficiently common that newspapers have begun rating the sound levels of restaurants. A Zagat survey in 2011 found that diners ranked noise as their second-highest complaint behind service. Comal Restaurant in Berkeley California is the first to use a combination of active and passive acoustic treatments to control the reverberation time of the dining area.

2:30

2pAA6. Sight and sound: Visual aesthetics of loudspeakers. Ben Bridgewater, Bob Coffeen, and Jim Long (University of Kansas, Lawrence, KS 59741, BenBridgewater@gmail.com)

Loudspeakers are not only heard but often seen. A sound system's loudspeakers must be designed to meet the visual expectations of the architect, performers, and owners. A good designer can meet the audio requirements while not offending the architecture of the space. A case study of hidden, obvious, and ugly loudspeakers will be presented.

2:45

2pAA7. Review of an arena's acoustical and electroacoustical design upgrades. David Scheirman (Harman Professional, 8500 Balboa Blvd, Northridge, CA 91329, dscheirm@harman.com)

Sports arenas and stadiums are an important economic part of the urban centers in which they are located. For cost efficiency, acoustical elements of the original architectural design should complement and support the facility's requirements for communication and public address equipment. Multi-purpose use requirements are examined for one such facility, an indoor arena which measures 950,000 square feet (88,257.9 m^2) of total space, with a 94-foot (28.7 m) by 200-foot (61.0 m) arena floor. It hosts over 250 events and nearly 4,000,000 guests each year, seating 18,000-20,000 attendees per event based upon type and format. The timeline is reviewed over which acoustical and electroacoustical studies, upgrades and modifications were carried out. Original-construction sprayed cellulose acoustical insulation was enhanced by installation of lapendary panels to reduce reverberation time and sound intensity levels in target frequency bands. Finally, with the installation of a new multi-array line-array type loudspeaker system, the venue has achieved increased speech intelligibility and improved audience-perceived quality of sound reproduction for both sports and entertainment production events. The upgrades have helped to enhance its global reputation as a multi-functional space. The building was voted Venue of the Year at the 2011 Stadium Business Awards.

TUESDAY AFTERNOON, 23 OCTOBER 2012 JULIA LEE A/B, 1:00 P.M. TO 5:45 P.M.

Session 2pAB

Animal Bioacoustics: Arctic Bioacoustics

Michael A. Stocker, Chair
Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938

Chair's Introduction—1:00

Invited Papers

1:05

2pAB1. Soundscape of the North-Eastern Chukchi Sea. Bruce Martin (JASCO Applied Sciences, 32 Troop Avenue, Suite 202, Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), Julien Delarue, and David Hannay (JASCO Applied Sciences, Victoria, British Columbia, Canada)

The Chukchi Sea is a dynamic environment that cycles each year from open seas in summer to 100% ice cover in winter. The ice and sea conditions lead to a highly variable acoustic background. In summer the soundscape's backdrop are wind and wave sounds typical of shallow seas. In the winter, grinding ice sheets can create a cacophony of unusual sounds, some of which can easily be mistaken as being biological in origin. During ice formation and break-up sound levels can increase significantly due to combinations of wind and waves and ice floe collisions. Several marine mammal species move through this environment and their calls sometimes dominate the soundscape. Bearded seals are year-round residents whose calls overwhelm the soundscape in the spring during their mating season. Walrus follow the ice edge and fill their neighbourhoods with grunts, moans and knocks. Seasonal migrants such as bowhead and beluga whales are heard passing through in spring and fall. Summer visitors such as fin, killer, minke and gray whales add to the voices of this environment. Anthropogenic sounds mix into the soundscape during the open water season and are concentrated along the coast lines and in areas of interest for oil and gas exploration. This presentation will provide examples of all these sounds and will show spatial-temporal maps of the sounds distributions based on data from the Chukchi autonomous recording array operated by the Joint Studies Program since 2007.

1:25

2pAB2. Seals and sound in a changing Arctic: Ongoing psychoacoustic studies of spotted and ringed seals. Jillian Vitacco (Ocean Sciences, University of California at Santa Cruz, 100 Shaffer Rd, Santa Cruz, CA 95050, jillian.vitacco@gmail.com), Colleen Reichmuth, Asila Ghoul (Long Marine Laboratory, Institute of Marine Sciences, University of California, Santa Cruz, CA), and Brandon Southall (Southall Environmental Associates, Aptos, CA)

Arctic environments are changing rapidly as a result of climate warming and industrialization, and as sea ice recedes, activity associated with transportation and oil and gas production increases. Among the many concerns for ice-living seals in the region is the potential for behavioral or auditory effects resulting from noise exposure. Currently there are limited data available concerning the hearing sensitivity of arctic seals - some data exist for harp and ringed seals, while the most comprehensive data are for harbor seals. As the phylogenetic relationships among northern seals are not well resolved, extrapolation across species for management purposes is difficult. To this end, we are working to describe the species-typical hearing of spotted (*Phoca largha*) and ringed seals (*Pusa hispida*). Thus far, measurements of the underwater hearing sensitivity of spotted seals show best sensitivity between 3.2–25.6 kHz and peak sensitivity of 51 dB re 1 μPa at 25.6 kHz. Absolute thresholds for airborne tonal signals indicate acute sensitivity of <10 dB re 20 μPa from 0.80–12.8 kHz. Audiometric testing for ringed seals is ongoing, as are critical ratio measurements for both species. These studies will provide valuable insight into how arctic seals perceive acoustic signals, as well as inform management practices for these vulnerable species.

1:45

2pAB3. Simultaneous sound production in the bowhead whale Balaena mysticetus—Sexual selection and song complexity. Outi M. Tervo (Arctic Station, University of Copenhagen, Post box 504, Qeqertarsuaq 3905, Greenland, outiter@gmail.com), Lee A. Miller (Institute of Biology, University of Southern Denmark, Odense, Denmark), and Mads F. Christoffersen (Arctic Station, University of Copenhagen, Qeqertarsuaq, Qaasuitsup, Greenland)

Different components of bowhead whale Balaena mysticetus song were localized using hydrophone arrays. In 2008 recordings were made using two hydrophones spaced 20–35 m apart. In 2009 a linear GPS synchronized array of four hydrophones with an aperture of ~1400 m was used. The localization results confirm the co-location of the sound sources. The analyses show amplitude modulation of one signal caused by the onset of the second signal, which provides additional evidence of simultaneous sound production. Sound, when used as an indicator of fitness forces the vocalizing animal to improve the quality of its signal to compete with other vocalizing conspecifics. Several methods can be used to improve signal quality and these include 1) large repertoire size, 2) annually/seasonally changing repertoire, 3) broad frequency band, and 4) simultaneous sound production. Bowhead whales show all these features in their acoustic behaviour and we suggest that these complex songs have evolved as the result of sexual selection. Song complexity has been shown to be of importance in the sexual selection of many song bird species implying that sound complexity may be a key factor in the sexual behaviour of bowhead whales.

2:05

2pAB4. Bowhead whales and airgun pulses: Detecting a threshold of behavioral reaction. Susanna B. Blackwell (Greeneridge Sciences, Inc., Santa Barbara, CA 93117, susanna@greeneridge.com), Trent L. McDonald, Christopher S. Nations (WEST, Inc., Cheyenne, WY), Aaron M. Thode (Marine Physical Laboratory, Scripps Institution of Oceanography, San Diego, CA), Katherine H. Kim, Charles R. Greene (Greeneridge Sciences, Inc., Santa Barbara, CA), and Michael A. Macrander (Shell Exploration and Production Co., Anchorage, AK)

Previous work has shown that bowhead whales cease calling when near (<40 km) seismic exploration activities involving airguns. The aim of this study is to estimate the received level threshold for the onset of this behavior (cessation of calling). The analysis relied on data collected during late summer of 2007-2010 by up to 40 Directional Autonomous Seafloor Acoustic Recorders (DASARs) in the Beaufort Sea. About 98,000 localized calls and hundreds of thousands of airgun pulses were included in the analysis. For each 10-min period of data collected at each recorder, each year, the cumulative sound exposure level (CSEL) from airgun pulses was calculated and paired with the number of calls concurrently localized within ~3.5 km of each DASAR. Poisson regression was then used to estimate the threshold of airgun sound exposure received at the whales when call cessation begins. The CSEL threshold was found to be near 124 dB re 1 μPa^2 s (95% confidence intervals = 119-129 dB). For an airgun array firing every 10 sec, this corresponds to a received single pulse SEL at the whale of ~106 dB re 1 μPa^2 s. [Work supported by Shell Exploration and Production Company.]

2:25

2pAB5. Arctic marine mammal passive monitoring and tracking with a single acoustic sensor. Juan I. Arvelo (Applied Physics Laboratory, Johns Hopkins University, 11100 Johns Hopkins Rd., Laurel, MD 20723, juan.arvelo@jhuapl.edu)

The Arctic Ocean is a unique environment in the number of physical mechanisms that may be potentially exploited with much simpler acoustic systems than would be required in other oceans. The Arctic sound speed profile forms a surface duct with favorable cylindrical-spreading for near-continuous detection of marine mammal vocalizations. This ducted waveguide exhibit low seasonal variability, particularly under the ice cap, forcing under-ice sound to heavily interact with this rough elastic stratified boundary. The ice roughness introduces steeper slopes that enhance water-to-ice sound penetration [Arvelo, POMA 2012]. The ice elasticity is responsible for the excitation of a radially polarized longitudinal wave and a transverse-horizontal shear wave with group velocities around 2700-3000 m/s and 1550-1650 m/s, respectively. A third dispersive flexural vertical plate wave propagates at much slower speeds (<1200 m/s) at low frequencies [Stein, Euerle & Parinella, JGR 1998]. Vocalization distance may be estimated from the time delays between the three wave types via blind deconvolution, while an arctangent bearing-estimator may increase the azimuthal localization resolution for high SNR vocalizations [Maranda, Oceans 2003]. Therefore, the unique Arctic environment is well suited for passive marine mammal monitoring and tracking with just a single ice-embedded geophone or under-ice vector sensor.

2:45–3:00 Break

3:00

2pAB6. Long-range tracking of bowhead whale calls using directional autonomous seafloor acoustic recorders. Delphine Mathias, Aaron M. Thode (Scripps Institution of Oceanography, 9500 Gilman Drive, La Jolla, CA 92037-0238, delphine.mathias@gmail.com), Katherine H. Kim, Susanna B. Blackwell, Charles R. Greene (Greeneridge Sciences, Santa Barbara, CA), and Michael A. Macrander (Shell Exploration and Production Co., Anchorage, California)

Since 2007 "Directional Autonomous Seafloor Acoustic Recorders" (DASARs) have been deployed at five sites across a 280 km swath of the Beaufort Sea continental shelf to record bowhead whale (Balaena mysticetus) calls during their autumn migration. Composed of an omnidirectional pressure sensor and two horizontal directional sensors measuring particle motion, DASARs provide information for determining the bearing to a sound source. In previous analyses, bearings obtained from multiple DASARs within a single site have been used to localize calls, with a maximum baseline separation of 21 km between instruments. Here, we use data collected from two different sites to exploit a 45 km instrument separation for tracking bowhead whale calls detected under low ambient-noise situations in 2009 and 2011. These data sets have been manually analyzed to extract a series of calls from individual whales swimming from one site toward another. The fact that the tracking method does not require relative arrival time information makes matching calls between these widely separated shallow-water recordings practical. Both empirical and numerical transmission loss models are used to investigate the relationship between animal orientation, source level, and the presence of harmonics.

3:15

2pAB7. Acoustic monitoring of belugas (*Delphinapterus leucas*) in the eastern Chukchi Sea. Ellen C. Garland, Catherine Berchok (National Marine Mammal Lab, Alaska Fisheries Science Center, NOAA, AFSC/NOAA, 7600 Sand Point Way NE, Seattle, WA 98115, Ellen.Garland@noaa.gov), and Manuel Castellote (National Marine Mammal Lab, Alaska Fisheries Science Center, NOAA, Seattle, WA)

Beluga whales (*Delphinapterus leucas*) are highly vocal animals which make them ideal candidates for passive acoustic monitoring. Alaskan belugas overwinter in the Bering Sea, and in spring, two subpopulations migrate to their predictable summering grounds in the eastern Chukchi and eastern Beaufort Seas. In autumn, these subpopulations complete their annual migration by returning south to the Bering Sea. Additional information is required on the timing and migration routes in spring and autumn to assist in detecting these subpopulations as they transit between the two regions. Preliminary results are presented on the temporal distribution of Alaskan beluga based on acoustic detection (September 2010 to May 2011) from a passive acoustic recorder located 30 miles off Icy Cape, AK, in the eastern Chukchi Sea. Belugas were sporadically detected in autumn from mid-September to the start of December, with a peak in detections in late November. In spring, belugas were detected from mid-April to the end of May, with peaks in early and late May. Within each temporal peak in vocal activity we will investigate the common call types to assist in differentiating migratory streams, and potentially identify each subpopulation as they transit through this inshore area. [Work supported by the National Research Council and Bureau of Ocean Energy Management.]

3:30

2pAB8. Pacific walrus vocal repertoire in the northeastern Chukchi Sea: Call type description and relative proportion in time and space. Xavier Mouy (JASCO Applied Sciences, Victoria, BC, Canada), Julien Delarue (JASCO Applied Sciences, 202 - 32 Troop Ave, Dartmouth, NS B3B 1Z1, Canada, julien.delarue@jasco.com), Bruce Martin (JASCO Applied Sciences, Dartmouth, NS, Canada), and David Hannay (JASCO Applied Sciences, Victoria, BC, Canada)

Pacific walrus are present in the northeastern Chukchi Sea (NCS) from June to October. The study of their sounds has been largely restricted to the knock and bell sounds produced by males during the breeding season and in-air grunts and barks from mother and pups. A passive acoustic monitoring program conducted in the NCS since July 2006 has brought strong evidence that the underwater vocal repertoire of walrus is more diverse. Nine call types (including knocks and bells) and their variants identified over four years of acoustic monitoring will be described. Spectral measurements along with estimates of variability for high signal-to-noise ratio calls will be provided. The relative proportion of each call type across the study area and throughout the season is currently analyzed based on the identification of all calls in samples recorded multiple times per day in 2009 and 2010. Preliminary results suggest that the vocal repertoire of walrus is dominated by grunt-type calls, which is consistent with the NCS herds being mainly composed of females, pups and juveniles. The recurrent presence of knocking sounds indicates that either adult males routinely occur in the study area or that other age and sex classes also produce this call type.

3:45

2pAB9. Right whale versus bowhead whale gunshot calls in the Bering Sea. Catherine L. Berchok, Jessica L. Crance, Jessica L. Thompson, Stephanie L. Grassia, Phillip J. Clapham (National Marine Mammal Lab, NOAA/AFSC, 7600 Sand Point Way NE, Seattle, WA 98115, Catherine.Berchok@noaa.gov), Dana L. Wright (School of Fisheries and Ocean Sciences, University of Alaska Fairbanks, Fairbanks, AK), and Marc O. Lammers (Hawaii Institute of Marine Biology, University of Hawaii, Kaneohe, HI)

The Bering Sea is an important area for many cetaceans. It functions as both summer feeding grounds for some species like humpback and right whales, and wintering grounds for more northern species such as bowhead and beluga whales. Passive acoustics is an important tool for monitoring the presence of these and other species in the Bering Sea, although similarities in call characteristics among species can create confusion. For example, gunshot calls are produced by both right and bowhead whales. Recordings made in the Bering Sea (2007-2011) have been analyzed for the presence of gunshot calls. The spatio-temporal distribution of these calls is compared to that of more 'standard' bowhead and right whale calls. With the inclusion of information on historical range, migration patterns, and ice coverage, a tentative separation of the two species is proposed. Call characteristics and contextual information from the resulting subsets of data are then examined and compared with data confidently attributed to each species. Whether right and bowhead whale gunshot calls can be discriminated with sufficient reliability to include in spatio-temporal distribution analyses, and what added value this call type gives over using only 'standard' call types, will be discussed.

4:00

2pAB10. Cetacean vocalizations and anthropogenic noise levels in polar waters of the Atlantic. Sharon L. Nieukirk, Holger Klinck, Karolin Klinck, David K. Mellinger, Robert P. Dziak, and Haruyoshi Matsumoto (Cooperative Institute for Marine Resources Studies, Oregon State University and NOAA Pacific Marine Environmental Laboratory, 2030 SE Marine Science Drive, Newport, OR 97365, sharon.nieukirk@oregonstate.edu)

Obtaining baseline information on the distribution of endangered species in polar waters is important as climate change may adversely affect this fragile environment. In 2009 we began an acoustic survey in the Greenland Sea and Fram Strait to monitor the low-frequency calls from marine mammals using these waters. We also documented sources and levels of ambient noise as these will change as human use of the area increases. We recorded the vocalizations of numerous marine mammals; here we report our results for fin (Balaenoptera physalus), blue (B. musculus), sei (B. borealis) and sperm (Physeter macrocephalus) whales. The 20 Hz pulses of fin whales were recorded in the fall and early winter months. Sounds from blue whales, sperm whales and sei whales were recorded primarily in the summer and early fall. Background noise levels were dominated by the sounds from seismic airguns during the spring, summer and fall; during summer these sounds were recorded in all hours of the day and in all days of a month. Future increases in oil exploration and ship traffic coincident with melting sea ice will increase ambient noise levels, potentially affecting the numerous species of vocalizing whales using this area.

4:15

2pAB11. Spatio-temporal distribution of ice seals in the Chukchi Sea using underwater vocalizations: Special focus on male bearded seals. Heloise Frouin, Xavier Mouy, Julien Delarue (JASCO Applied Sciences, 2305 - 4464 Markham Street, Victoria, BC V8Z 7X8, Canada, heloise.frouin-mouy@jasco.com), Bruce Martin (JASCO Applied Sciences, Dartmouth, NS, Canada), and David Hannay (JASCO Applied Sciences, Victoria, BC, Canada)

Underwater vocalizations of ringed, ribbon and bearded seals were recorded over a wide region of the northeastern Chukchi Sea between July 2007 and October 2011. Ringed seals were identified by their barks and yelps, ribbon seals by their sweeps and puffing sounds and bearded seals by their trills, ascents and moans. Ringed seal vocalizations were detected in all months of the year, whereas vocalizations from ribbon seals occurred only in October and November. To determine the seasonal variation in the frequency of occurrence of male bearded seal vocalizations throughout a year 20-min recordings were analyzed to determine call counts between 02:00 and 06:00 every 3 days. Bearded seal acoustic detections increased progressively from August to March, peaked between April and June but were essentially absent in July. Outside of the mating period (April - June) the frequency of occurrence of vocalizations varied on a diel cycle and was higher during periods of darkness. To determine the influence of diel cycle, the duration of vocalizations and the proportion of each vocal type throughout the mating season 10-min recordings on a 17-20% duty cycle were analysed between April and June every 10 days. Results are currently under review and will be presented.

4:30

2pAB12. Spatio-temporal distribution of fin whales in the Bering Sea, 2007–2011. Jessica L. Thompson, Catherine L. Berchok, Phillip J. Clapham (National Marine Mammal Laboratory, NOAA/AFSC, 7600 Sand Point Way NE, Seattle, WA 98115, jessica.thompson@noaa.gov), Marc O. Lammers (Hawaii Institute of Marine Biology, University of Hawaii, Kaneohe, HI), and Sue E. Moore (National Marine Fisheries Service, Office of Science & Technology NOAA Fisheries, Seattle, WA)

The National Marine Mammal Laboratory has been collecting passive acoustic recordings of vocalizing marine mammals through much of the southeastern Bering Sea since 2000. The present analysis combines these recordings with those obtained in 2007 from a North Pacific Research Board-funded study (Stafford and Mellinger) to determine the long-term spatio-temporal distribution of fin whales throughout the Bering Sea shelf. A total of 28 moorings (19 Autonomous Underwater Recorders for Acoustic Listening (AURALs); 9 Ecological Acoustic Recorders (EARs)) were deployed year-round from October 2007 to September 2011 at varying depths and locations along the Bering Sea shelf, from Umnak Pass to just south of St. Lawrence Island. Preliminary analyses show an annual southward movement of calling fin whales in winter, seemingly associated with the expanding ice edge; these results will be compared with 1-day composite ice cover measured by the Advanced Microwave Scanning radiometer for the Earth Observing System (NOAA Coastwatch). Furthermore, the results show a considerably lower number of calls detected south of St. Matthews compared to the other mooring locations. In addition, overall fin whale seasonal calling trends along and between the 50m and 70m isobaths, and the species' use of Umnak Pass and Unimak Pass will be described.

4:45

2pAB13. Contributions of airgun activity to the ice-free ambient noise environment of the shallow Beaufort Sea between 2007 and 2011. Aaron Thode, Delphine Mathias, Katherine H. Kim, Susannna B. Blackwell (Scripps Institution of Oceanography, UCSD, 9500 Gilman Dr, La Jolla, CA 92093-0238, athode@ucsd.edu), Charles R. Greene (Greeneridge Sciences, Inc., Santa Barbara, CA), and Michael Macrander (Shell Exploration and Production Company, Anchorage, AK)

Every year since 2007 a collection of at least 35 "Directional Autonomous Seafloor Acoustic Recorders" (DASARs) have been deployed across a 280 km swath of the Beaufort Sea continental shelf, in water depths between 15 and 50 m. The ability of these instruments to estimate the arrival azimuth of transient signals has facilitated the development of an automated algorithm for the detection of airgun activity. This algorithm has been applied to five seasons of data, and in this presentation the contributions of this activity to the overall ambient noise background of the ice-free shallow-water Beaufort environment will be quantified with a variety of metrics, in terms of both level (peak-to-peak, rms, sound exposure level), frequency, and time (intervals and fraction of time present). During some years, up to four airgun operations could be detected simultaneously, and a random one-second time sample yielded a 40% chance of containing an airgun signal, but the levels detected are generally within the bounds of natural wind-driven ambient noise levels. This dataset provides useful empirical insight into discussions about the cumulative effects of anthropogenic activity on an environment extensively used by several marine mammal species. [Work sponsored by the Shell Exploration and Production Company.]

5:00–5:45 Panel Discussion

Session 2pBA

Biomedical Acoustics and Signal Processing in Acoustics: Biomedical Applications of Acoustic Standing Waves

Martin Wiklund, Chair
Applied Physics, Royal Institute of Technology, Stockholm 10691, Sweden

Chair's Introduction—1:00

Invited Papers

1:05

2pBA1. Acoustic focusing flow cytometry. Gregory Kaduchak, Gregory R. Goddard, and Michael D. Ward (Molecular Cell Biology Engineering, Life Technologies, 29851 Willow Creek Rd, Eugene, OR 97402, greg.kaduchak@lifetech.com)

Acoustic cytometry is a new technology that replaces or partly replaces hydrodynamic focusing of cells or particles in flow cytometry with forces derived from acoustic radiation pressure. The ability to focus cells into a tight line without relying on hydrodynamic forces allows many possibilities outside the scope of conventional flow cytometry. Dilute samples can be processed quickly. Flow velocities can be varied allowing control of particle delivery parameters such as laser interrogation time and volumetric sample input rates. Recently, Life Technologies unveiled a flow cytometer that directs particles into the laser interrogation region using acoustic radiation pressure. In this talk, the application of acoustic cytometry in flow cytometry systems from fundamental principles to details of its implementation will be presented. Data will be shown for both the operational implementation of the acoustic focusing device as well as demonstrating its ability to perform for complex biological assays.

1:25

2pBA2. Acoustophoretic cell sorting in microfluidic channels. Tom H. Soh and Allen Yang (Mechanical Engineering, UC-Santa Barbara, Santa Barbara, CA 93111, tsoh@engr.ucsb.edu)

In this work, we report the use of ultrasonic acoustophoresis for the label-free separation of viable and non-viable mammalian cells within a microfluidic device. This device exploits the fact that cells that have undergone apoptosis are physically smaller than viable cells, and achieves efficient sorting based on the strong size dependence of acoustic radiation forces in the microchannel. As a model, we have selectively enriched viable MCF-7 breast tumor cells from heterogeneous mixtures of viable and non-viable cells. We found that this mode of separation is gentle on cells, while enabling label-free separation at sample flow-rates of up to 12 mL/hr at a cell density of 10^6 cells/mL. We have extensively characterized the device and report the effects of piezoelectric voltage and sample flow-rate on device performance, and describe how these parameters can be tuned to optimize recovery, purity or throughput.

1:45

2pBA3. Standing surface acoustic wave technology for cell and organism manipulation. Sz-Chin S. Lin and Tony J. Huang (Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802, szchinlin@gmail.com)

Manipulation of particles, cells, and organisms is essential to many fundamental biological studies. For example, the ability to precisely control the physical settings of biological objects allows scientists to investigate the interactions between molecules, cells, or cell and its environment. Acoustic-based particle manipulation techniques possess non-contact and non-invasive natures and have rapidly become key enablers for many emerging lab-on-a-chip biomedical applications. We, BioNEMS lab at Penn State, have developed a series of standing surface acoustic wave (SAW) based acoustic tweezers that can dexterously manipulate a wide range of cellular-scale objects. This technology uses standing SAW induced acoustic radiation forces to trap suspended biological objects, whose physical location/orientation can then be dynamically changed by adjusting the parameters (e.g. frequency and amplitude) of the standing SAW. Simply by tuning the AC electrical signals, our system can perform sophisticated cell patterning, focusing, reorientation, separation, sorting, transportation, and stretching without direct contact. We show that our system can be seamlessly integrated with other on-chip devices and is fully compatible with fluorescence and confocal microscopies. The versatility, simplicity, low power consumption, non-contact and non-invasive natures render our system an excellent platform for a wide range of applications in the biological, chemical, and physical sciences.

2:05

2pBA4. Ultrasound standing wave fields for tissue engineering. Denise C. Hocking (Pharmacology and Physiology, University of Rochester, 601 Elmwood Ave., Box 711, Rochester, NY 14642, denise_hocking@urmc.rochester.edu), Kelley A. Garvin, and Diane Dalecki (Biomedical Engineering, University of Rochester, Rochester, NY)

The spatial organization of cells within native tissues contributes to proper tissue function. Thus, successful engineering of replacement tissues requires methods to control cell location within the engineered tissue. Our studies have focused on utilizing acoustic radiation forces associated with ultrasound standing wave fields (USWF) to rapidly and non-invasively organize cells within 3D collagen

gels. We have shown that USWF-induced alignment of fibroblasts into distinct multicellular planar bands increases cell contractility and enhances cell-mediated extracellular matrix reorganization. Additionally, USWF-induced patterning of endothelial cells accelerates the formation of capillary-like sprouts and supports the maturation of sprouts into lumen-containing, vascular networks throughout the volume of the collagen gel. Our recent studies have investigated the influence of various acoustic parameters on the USWF-induced spatial pattern of endothelial cells. The initial density of the USWF-induced cell bands affected both the rate of formation and the morphology of endothelial cell networks, indicating that different USWF-induced endothelial cell patterns can produce morphologically different vascular networks. Design of USWF, by choice of ultrasound frequency or use of multiple transducer geometries, can create more complex cell patterns within hydrogels. Thus, USWF technologies provide a novel approach to pattern large 3D engineered tissues in vitro.

2:25

2pBA5. Applications in acoustic manipulation of biological cells in micro-devices. Dyan N. Ankrett, Peter Glynne-Jones, and Martyn Hill (Engineering Sciences, University of Southampton, University of Southampton, University Road, Southampton, Hampshire SO17 1BJ, United Kingdom, M.Hill@soton.ac.uk)

Utilizing ultrasonic standing waves in biocompatible micro-fluidic devices, we are able to acoustically manipulate biological cells for a variety of potentially beneficial pharmaceutical, biomedical and environmental applications. Using a device designed to induce sonoporation in the absence of contrast agent micro-bubbles (CA-free sonoporation) we demonstrate both the uptake and efflux of differently sized, membrane impermeable molecules whilst maintaining high cell viability. Crucially we show that the cytotoxic action of several known pharmaceutical agents is significantly increase in porated cells compare with non-porated control cells, suggesting sonoporation-induced facilitated uptake of these agents. We also report on a device that acoustically excites tethered polymer-shelled micro-bubbles to induce micro-streaming around cardiomyocyte membranes in order to mimic myocardial infarction, ischaemia and induction of apoptosis. Initial results from a levitation culture system that continuously perfuses a pellet of ultrasonically suspended chondrocytes are also presented. The pellet geometry allows large numbers of cells to be cultured without developing a necrotic core. Primary chondrocytes and cell lines demonstrated good viability after more than ten days of levitation. Using low power, continuous ultrasonic excitation we have demonstrated significant reductions in biofilm growth in polymer fluid channels potentially increasing the lifespan of in-situ marine sensors.

2:45

2pBA6. Microfluidic air-liquid cavity acoustic transducers for point-of-care diagnostics applications. Abraham Lee and Maulik Patel (Biomedical Engineering, UC Irvine, 3120 Natural Sciences II, Irvine, CA 92697-2715, aplee@uci.edu)

Microfluidic devices with “side channels” that trap air enable acoustic energy coupling and acoustic streaming into the main channel. This basic configuration is versatile and can be designed as a microfluidic pump, mixer, particle trap, cell sorting switch, and sample separation component. These multiple functions are integrated on a microfluidic platform and provides rapid and specific diagnostics of infectious diseases based on the immune response detected in a drop of blood. A second implementation of this concept is to utilize microfluidic produced lipid microbubbles that respond to acoustic energy as ultrasound contrast agents with drug carrying payload and molecular ligand targeting that can detect and treat molecular diseases such as cancer and cardiovascular diseases. This talk will introduce both of these microfluidic acoustic transducer projects in my lab.

3:05–3:20 Break

3:20

2pBA7. Cell lysis using acoustic cavitation bubbles in microfluidics. Tandiono Tandiono, Siew-Wan Ohl (Fluid Dynamics, Institute of High Performance Computing, 1 Fusionopolis Way, Singapore 138632, tandiono@ihpc.a-star.edu.sg), Cara Sze-Hui Chin, Dave Siak-Wei Ow (Bioprocessing Technology Institute, Singapore), and Claus-Dieter Ohl (Physics and Applied Physics, Nanyang Technological University, Singapore)

Analysis of intracellular contents, such as proteins and nucleic acids, in a micro-scale system is gaining its importance in biomedical research. However, an efficient cell lysis needs to be achieved before the analysis can be carried out. The standard lysis methods in microfluidics, for example: by means of chemicals, thermal, or electrical lysis, suffer from the undesirable temperature increase and cross-contamination, which may lead to the denaturation of proteins or interfere with subsequent assays. Here, we present a technique to mechanically lyse microbial cells using acoustically driven cavitation bubbles in a polydimethylsiloxane based microfluidic channel attached on a glass slide. The cavitation bubbles are created by exciting gas-liquid interfaces in the microchannel into nonlinear interface instability with ultrasonic vibrations. The strongly oscillating bubbles create regions with intense mixing and high shear stress, which can deform and rupture the nearby cells. *Escherichia coli* (bacteria) and *Pichia pastoris* (yeast) cells are completely lysed in less than 0.4 seconds and 1.0 second, respectively. The temperature increase of the samples during the ultrasound exposures is less than 3.3 °C. Fluorescence intensity measurements and real-time polymerase chain reaction (qRT-PCR) analysis suggest that the functionality of the harvested protein and genomic DNA is maintained.

3:40

2pBA8. Structures formed by ultrasonic standing waves in active fluids and suspensions. Mauricio Hoyos (PMMH, CNRS, ESPCI, 10 rue Vauquelin, Paris 75005, France, hoyos@pmmh.espci.fr), Angelica Castro, Eric Clément, Annie Rousselet (PMMH, CNRS, Paris, Ile de France, France), Despina Bazou (Steel Lab, Massachusetts General Hospita, Charlestown, MA), Wei Wang, and Thomas Mallouk (Center for Solar Nanomaterials, Penn State University, State College, PA)

The acoustic radiation force concentrates particulate materials at the nodes or antinodes of an ultrasonic standing wave (USW), in function of different physicochemical parameters: size, shape, density or elastic properties. Thus, frequencies from 0.5 to 10 MHz are adapted for manipulating micron-sized particles, cells, bacteria, vesicles, drops, bubbles and even colloidal species. The interaction of different species with the ultrasonic radiation field generates levitation, aggregation or coalescence. In this presentation, new behaviors

will be presented in active fluids: bacteria bath and self-propelled metallic micro-cylinders. Leaving bacteria in culture medium undergoing the USW field show a dynamics inducing complex structures. USW propel, rotate, align and assemble metallic micro-rods (2 μm long and 330 nm diameter) in water as well as in solutions of high ionic strength, generating "self-acoustophoresis". Finally, new possibilities for controlling aggregation forming 2D and 3D particle and cancer cells structures using pulsed ultrasound will be shown.

4:00

2pBA9. Acoustic trapping with seed-particles for submicron particle enrichment. Björn Hammarström, Simon Ekström, Thomas Laurell, and Johan Nilsson (Measurement Technology and Industrial Electrical Engineering, Lund University, P.O. Box 118, Lund 221 00, Sweden, bjorn.hammarstrom@elmat.lth.se)

Acoustic trapping in disposable borosilicate capillaries utilize ultrasonic forces to capture/retain micro-particles or cells against fluid flow in a microfluidic-channel. A miniaturized ultrasonic transducer is used to locally excite a 4-MHz cross-sectional resonance in the capillary, creating an acoustic field gradient for retention of cells in non-contact mode. Due to competition between fluidic drag from induced acoustic streaming and primary radiation force the smallest particle size addressable with the trapping system is limited. Here, the typical transition occurs at single-micron particle diameters. However, trapping of single- or sub-micron biological species has highly relevant applications such as enrichment or purification of bacteria or viruses. This work investigates the influence of in-trap particle concentration on the trapping, and it is found that elevated concentrations allow capture of submicron particles. By preloading the acoustic trap with micron-sized seed-particles capture of submicron particles even at low concentrations is enabled. Using this technique, we demonstrate single event capture of bacteria as well as capture of 100nm particles. To provide analytical readout for identification/analysis of the trapped particles the acoustic trap is interfaced with a MALDI-MS instrument. Here, the acoustic trapping capillary is operated in aspirate/dispense mode allowing easy and flexible handling of small sample volumes.

4:20

2pBA10. Ultrasonic standing waves for dynamic micro-array cytometry. Martin Wiklund, Athanasia Christakou, Mathias Ohlin, and Björn Önfelt (Applied Physics, Royal Institute of Technology, KTH-Albanova, Stockholm 106 91, Sweden, martin.wiklund@biox.kth.se)

We describe a novel platform for dynamic micro-array cytometry (DMAC), i.e., parallel screening of individual cell-cell interactions based on ultrasonic standing wave aggregation and positioning of cells in a multi-well microplate. Upon ultrasound actuation, clusters containing one or a few cells are quickly formed and retained in a precise location synchronously in each of the 100 wells on the microplate. By combining the acoustic cell handling tool with high-resolution fluorescence microscopy, detailed time-lapse monitoring of individual cell-cell interactions in a highly parallel manner is possible. Of particular interest in our group is to study the long-term interaction between natural killer (NK) cells and different target cells at the level of single cells. In this talk we demonstrate both theoretically and experimentally how to design a microchip capable of trapping and positioning individual cells by ultrasound in a highly parallel manner, and with a spatial accuracy of the order of a cell diameter. We quantify the cell cluster motility with and without retained ultrasound exposure during 17 h, and we report on the viability of cells when exposed to continuous ultrasound for up to three days. Finally, we quantify the heterogeneity of NK cells' cytotoxicity against cancer cells.

Contributed Papers

4:40

2pBA11. Acoustic radiation force on a sphere in tissue. Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Applied Research Laboratories, University of Texas, Austin, TX 78713-8029, hamilton@mail.utexas.edu)

A theory is presented for the acoustic radiation force on a sphere embedded in a soft elastic medium that possesses a shear modulus μ several orders of magnitude smaller than its bulk modulus. Scattering of both compressional and shear waves is taken into account. There is no restriction on the size of the sphere or, apart from axisymmetry, the form of the incident compressional wave. The analysis employs the Piola-Kirchhoff pseudostress tensor and Lagrangian coordinates. In the linear approximation an analytical solution is obtained for the scattered waves. The nonlinear stress and full radiation force are calculated at the next order of approximation. For a small sphere and $\mu\approx 0$ the classical result for a particle in liquid is recovered. For small but finite shear modulus the radiation force is evaluated for a gas bubble driven at a frequency below resonance. The predicted magnitude of the radiation force on the bubble is found to be less than that in liquid by the factor $[1+(4/3)\mu/\gamma P_0]^{-1}$, where P_0 is the ambient pressure and γ the ratio of specific heats of the gas. Influence of the scattered shear wave in this limit is negligible. [Work supported by NIH DK070618.]

4:55

2pBA12. Development of a computational model to predict cranial resonance shifts due to changes in intracranial pressure. Andrew A. Piacsek and Sami Abdul-Wahid (Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

A possible method for noninvasively monitoring changes in intracranial pressure is to measure changes in skull resonance frequencies. Recent measurements of the vibrational response of a spherical aluminum shell clearly demonstrate that resonance frequencies shift higher as the internal pressure is increased [Piacsek et al, J. Acous. Soc. Am, 131, EL506-510 (2012)]. The frequency shift is approximately linear with the applied pressure, regardless of whether the shell is filled with air or water, and circumferential modes exhibit larger resonance shift than longitudinal modes. A computational model of a fluid-filled thin shell subject to acoustic stimulation was developed using the COMSOL multi-physics software to investigate the role of shell material and geometry in resonance shifts. The model predicts frequency shifts comparable to those observed in the spherical aluminum shell. Preliminary computational results for a spherical shell made of bone-like material, as well as for asymmetric and nonuniform shells, will be presented.

5:10

2pBA13. Numerical analysis of nonlinear standing waves in two-dimensional acoustic resonators. Fangli Ning (School of Mechanical Engineering, Northwestern Polytechnical University, 127 Youyi Xilu, Xi'an, Shaanxi 710072, China, ningfl@nwpu.edu.cn), Xiaofan Li (Department of Applied Mathematics, Illinois Institute of Technology, Chicago, IL), and Juan Wei (School of Communication Engineering, Xidian University, Xi'an, Shaanxi, China)

High amplitude nonlinear standing waves in acoustic resonators have been used in many engineering applications ranging from microfluidic devices for biomedical research, acoustic compression, to acoustic seal. The distribution of physical properties inside two-dimensional resonators is very useful for developing the application of nonlinear standing waves. Most of the previous numerical studies were limited to one-dimensional resonators, and analyzed standing waves with an assumption: the physical variables are assumed as finite sums of basic functions. In the study, the formation of the standing waves in the two-dimensional resonators is computed in the time domain from the initial position at rest without any predefined standing waves. The two-dimensional unsteady compressible Navier-Stokes equations and the state equation for an ideal gas are employed. This study extends the traditional pressure based finite volume SIMPLEC scheme for solving the equations. Initially, the pressure waves predicted in a two-dimensional cylindrical resonator are excellent agreement with the results obtained with pervious numerical methods in particular finite element and finite difference method. Next, we also investigate the velocity waveforms, and find that the sharp velocity spikes appear at the two ends of the resonator. Finally, the distribution of physical properties inside a two-dimensional cylindrical resonator is obtained.

5:25

2pBA14. Compound manipulation of micro-particles using a single device:Ultrasonic trapping, transporting, and rotating. Kun Jia, Keji Yang, Jian Chen, and Jianxin Meng (Department of Mechanical Engineering, Zhejiang University, Zheda Road No. 58, Hangzhou 310027, China, jiakun@zju.edu.cn)

Ultrasonic manipulation is widely used as a noncontact technology and recently small particles rotating on a vibrating substrate has been observed. In this report,a novel methodology which compounds the procedures of ultrasonic trapping, transporting and rotating micro-particles in fluid using a single device is investigated. Irregular micro-particles in a standing wave field experience both acoustic radiation force and torque, which drive the particles to pressure nodes and keep them in a balance posture. A prototype device has also been built according to this theory, which six phase-controlled piezoelectric transducers whose sound beam axes are arranged with an angle of 60 deg in the x-y plane are used to generate ultrasonic standing waves with arbitrary node. The transducers are divided to two groups, so the wave field can be rotated by switching between the two groups. The synthesized sound field is scanned using a needle hydrophone and 200 μm irregular SiO_2 particles are used to perform the compound manipulation ability of our device. The experimental results show good agreement with the theoretical calculation and the possible reason accounting for the small deviations is also discussed. This method may provide more complex and elaborate applications in micro-assembling and cell biomechanics.

TUESDAY AFTERNOON, 23 OCTOBER 2012 SALON 7 ROOSEVELT, 1:30 P.M. TO 3:30 P.M.

Session 2pEA

Engineering Acoustics: General Topics in Engineering Acoustics

Jerry H. Ginsberg, Chair
School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332

Contributed Papers

1:30

2pEA1. Collocation analysis of higher mode coupling in waveguides with discontinuous cross-section. Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332, jerry.ginsberg@comcast.net)

Standard techniques for analyzing transmission and reflection at discontinuities in a waveguide use weighted residuals to match modal series on each side of the junction. Convergence of the model equations generally is slow, which led Homenticovschi and Miles [JASA, 128 (2010) 628-638] to introduce a renormalization technique. Such a formulation does not lend itself well to implementation of a general computational algorithm. Slow convergence of the standard approach a consequence of using weighted residuals, which is a smoothing process, to interrogate a discontinuity. Evidence of this may be found in the analysis of scattering from cylinders having a discontinuous surface impedance [Ginsberg, JASA, 92 (1992) 1721-1728]. That work showed that a weighted residual formulation converges slowly, whereas a proper description of the field, including the singular nature of the velocity at the discontinity, is obtained from a collocation solution that satisfies the surface boundary condition at a discrete set of points. The present work uses standard modal series to represent the signal in each side of the discontinuity. The junction conditions of the waveguide are satisfied at a dscrete number of points. The rate of convergence will be assessed for a variety of schemes for selecting the collocation points, and the implementation of the formulation as part of a computational algorithm of networks will be discussed.

1:45

2pEA2. Dynamic impedance matching for active noise control in a cylindrical waveguide. Dong Joo Lee, Jae-Wan Lee (School of Mechanical Engineering, Yonsei Univ., Seoul, Republic of Korea), Jae Jin Jeon, Young Soo Seo (Agency for Defense Development, Changwon, Gyeongsangnam-do, Republic of Korea), and Won-Suk Ohm (School of Mechanical Engineering, Yonsei Univ., Engineering Building A586, Yonsei-ro, Seodaemun-gu, Seoul 120-749, Republic of Korea, ohm@yonsei.ac.kr)

In this paper, the feasibility of dynamic impedance matching for noise reduction in a cylindrical waveguide is demonstrated. An active acoustic coating, inserted parallel to the direction of wave propagation, is assumed to dynamically match the acoustic impedance of the incoming wave. The active coating appears as an acoustic branch containing the same fluid, and therefore part of the incoming wave can be diverted to and dissipated in the coating. The performance of the active coating is evaluated using a finite element analysis, where the coating is modeled as a dynamic impedance matching boundary. Simulations reveal that significant reductions in the downstream acoustic pressure can be achieved. Unlike the conventional active techniques that employ phase cancellation, dynamic impedance

matching has a number of advantages such as a relatively low power requirement.

2:00

2pEA3. Underwater sound transmission through water tunnel barriers. Matthew J. VanOverloop and David R. Dowling (Mechanical Engineering, University of Michigan, University of Michigan, Ann Arbor, MI 48109, mjvanoverloop@gmail.com)

In hydrodynamic test facilities with flowing water, receiving hydrophones are commonly placed behind a solid barrier to reduce flow noise from buffeting and turbulence. However, acoustic waves from the sound sources under study, such as cavitation bubbles, may propagate through the barrier as compression or shear waves, and the presence of the second wave type distorts signals recorded by the hydrophones. Such distortion depends on the structural characteristics of the barrier and the source-receiver geometry, and it may lead to sound-source detection and localization errors. This presentation describes results from a combined experimental and computational effort to understand the sound transmission characteristics of plastic and metal barriers typically used in water-tunnel testing. The experiments were conducted in a 1.0-meter-deep and 1.07-m-diameter cylindrical water tank using a single sound projector, a receiving array of 16 hydrophones, and impulsive (100 micro-second) signals having center frequencies from 30 kHz to 100 kHz. Computations intended to mimic the experiments are completed with the wave number integration software package OASES. Together the computations and experiments allow the most important barrier parameters to be identified. Dependencies of the received signal on the barrier parameters are presented. [Supported by NAVSEA through the Naval Engineering Education Center.]

2:15

2pEA4. Estimation of parameters of a head-related transfer function customization model. Kenneth J. Faller and Kathleen Hoang (Computer Engineering, California State University, Fullerton, 800 N. State College Blvd., Fullerton, CA 92831, jfaller@fullerton.edu)

Head-Related Transfer Functions (HRTFs) are special digital filters used to create the effect of three-dimensional (3D) virtual sound source placement over headphones. The two most common methods of obtaining HRTFs are to either individually measure the HRTFs on specialized equipment (individualized HRTFs) or to create a set of generic HRTFs by measuring them on a mannequin with average anatomical features (generic HRTFs). Individualized HRTFs required specialized equipment that is not readily available to the general public. Additionally, it is known that HRTFs are heavily dependent on our anatomical features. As a result, generic HRTFs produce significant localization errors. A multi-linear model is now available which uses simple anthropometric measurements of the intended user's anatomy to generate customized HRTFs. These customized HRTFs can be generated without specialized equipment and have improved spatialization over generic HRTFs. However, the anthropometric measurements, which are used as parameters for the customization model, are currently collected manually. In the present work, image processing techniques are used to automatically estimate a portion of the anthropometric measurements of the human pinnae. Analysis of the estimation technique's performance will also be conducted.

2:30

2pEA5. On the secondary path of headset active noise cancellation systems. Buye Xu, Jinjun Xiao (Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344, buye_xu@starkey.com), Thomas E. Miller, Erik Wiederholtz, Daniel M. Warren (Knowles Electronics, Itasca, IL), and Tao Zhang (Starkey Hearing Technologies, Eden Prairie, MN)

Active noise cancellation (ANC) technologies have been successfully applied in headsets to reduce ambient noise in the ear canal. The performance of such a technology depends on the characteristics of the secondary path (SP). In this paper, two different systems are studied: one system utilizes a moving-coil speaker as the secondary source and forms a closed cavity over the ear, while the other one uses a balanced-armature receiver inside the ear canal as the secondary source. The differences in the SP responses of these two systems are investigated both experimentally and theoretically. The implications for designing an effective ANC solution will be discussed based on numerical simulations.

2:45

2pEA6. Free-field reciprocity calibration of low-frequency ultrasonic transducers. M. R. Serbyn (Physics, Morgan State University, 1700 E. Cold Spring Ln., Baltimore, MD 21251, m_rserbyn@msn.com)

Techniques developed for the calibration of electroacoustic transducers provide many practical applications of the principles studied in advanced undergraduate and introductory graduate courses. This communication describes an implementation of the standard method of reciprocity calibration as presented in the popular text, Fundamentals of Acoustics by Kinsler et al. The measurements were performed on inexpensive off-the-shelf transducers designed to operate near 25 kHz and 40 kHz. The anechoic environment consisted of a 2-m3 box lined with absorbing material whose absorption coefficient was measured by the pulse-echo technique. The air density and the speed of sound were calculated using formulas available in the literature. The resulting values of transducer sensitivity (calibration factor) compared well, within 5 -10 %, with the values specified by the vendor. No accuracy statement was available for the transducers under test, not only because of their price, but mainly because no primary calibration services are available in the low ultrasonic range of frequencies, in contrast to the higher, MHz, range. A recent dissertation by N. Bouaoua, discovered in the course of this research, could fill this void should its procedures be adopted by a standards laboratory.

3:00

2pEA7. Acoustic supercoupling through a density-near-zero metamaterial channel. Romain Fleury, Caleb F. Sieck (Department of Electrical and Computer Engineering, University of Texas, Austin, TX), Michael R. Haberman (Applied Research Laboratories, University of Texas, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu), and Andrea Alù (Department of Electrical and Computer Engineering, University of Texas, Austin, TX)

Originally demonstrated with electromagnetic waves, supercoupling describes the extraordinary matched transmission, energy squeezing, and anomalous quasistatic tunneling through narrow channels. This behavior is the result of impedance matching achieved when the effective properties within the channel approach zero. For electromagnetic waves, supercoupling is observed when the electric permittivity in the channel approaches zero. These channels are accordingly known as epsilon-near-zero (ENZ) channels. This work shows that analogous behavior exists in the acoustic domain when the effective density is nearly zero, which can be achieved by tailoring the structure of the channel. Such channels are therefore known as density-near-zero (DNZ) metamaterial channels. Unlike tunneling based on Fabry-Perot resonances, DNZ transmission is independent of channel length and geometry and yields a uniform field along the entire length of the channel. Transmission-line theory is used to describe this peculiar phenomenon and finite element simulations are presented to confirm the unusual transmission properties of the metamaterial channel. It is further shown that acoustic waves may provide a unique possibility of squeezing acoustic energy through arbitrarily small channels in three dimensions, overcoming limitations that arise in the electromagnetic case.

3:15

2pEA8. Acoustic condition monitoring of wind turbines: Tip faults. Daniel J. Comboni and Bruno Fazenda (Acoustics Research Centre, University of Salford, Salford, Greater Manchester, United Kingdom, danielcomboni@gmail.com)

As a significant and growing source of the world's energy, wind turbine reliability is becoming a major concern. At least two fault detection techniques for condition monitoring of wind turbine blades have been reported in early literature, i.e. acoustic emissions and optical strain sensors. These require off-site measurement. The work presented here offers an alternative non-contact fault detection method based on the noise emission from the turbine during operation. An investigation has been carried out on a micro wind turbine under laboratory conditions. 4 severity levels for a fault have been planted in the form of added weight at the tip of one blade to simulate inhomogeneous debris or ice build up. Acoustic data is obtained at a single microphone placed in front of the rotor. Two prediction methods have been developed and tested on real data: one based on a single feature - rotational frequency spectral magnitude; and another based on a fuzzy logic interference using two inputs - spectral peak and rotational peak variation with time. Results show that the single spectral peak feature can be used to determine fault severity in ranges. The implementation of the fuzzy logic using a further input feature is shown to significantly improve the detection accuracy.

TUESDAY AFTERNOON, 23 OCTOBER 2012 ANDY KIRK A/B, 1:00 P.M. TO 2:00 P.M.

Session 2pED

Education in Acoustics: Take 5's

Jack Dostal, Chair
Physics, Wake Forest University, Winston-Salem, NC 27109

For a Take-Five session no abstract is required. We invite you to bring your favorite acoustics teaching ideas. Choose from the following: short demonstrations, teaching devices, or videos. The intent is to share teaching ideas with your colleagues. If possible, bring a brief, descriptive handout with enough copies for distribution. Spontaneous inspirations are also welcome. You sign up at the door for a five-minute slot before the session starts. If you have more than one demo, sign up for two non-consecutive slots.

TUESDAY AFTERNOON, 23 OCTOBER 2012 ANDY KIRK A/B, 2:00 P.M. TO 4:50 P.M.

2p TUE. PM

Session 2pMU

Musical Acoustics and Education in Acoustics: Teaching Musical Acoustics to Non-Science Majors

Jack Dostal, Chair
Physics, Wake Forest University, Winston-Salem, NC 27109

Invited Papers

2:00

2pMU1. An introduction to the physics of the clarinet for musicians and other non-science majors. Wilfried Kausel (Inst. of Music Acoustics, Univ. of Music and performing Arts, Anton-von-Webern-Platz 1, Vienna 1030, Austria, kausel@mdw.ac.at)

Teaching physics or more generally science related matters to non-science majors requires illustrative comparisons, fascinating application examples, live experiments, multi-medial presentation and simplified representations which can be understood without much background knowledge but which still remain relevant for the taught subject. But first of all, such lectures do require a very dedicated and alert lecturer who is able to captivate his audience by his outstanding presence. Following (some of) these principles a short generally understandable lecture on the acoustics of the clarinet will be given."

2:20

2pMU2. The Physics of Music course as an introduction to science. Gordon Ramsey (Physics, Loyola University Chicago, 6460 N Kenmore, Chicago, IL 60626, gprspinphys@yahoo.com)

Our Physics of Music course is an integration of physics and music. We start with the mathematical structure of music, including scales, harmonies and chords. We discuss musical styles and how they differ. After an introduction of physics concepts, including waves, resonances, forces, energy and fluid flow, the physical structure of instruments in the various groups are studied. Connection is made of the instruments and how they reproduce the mathematical nature of music. Finally, venue acoustics are investigated. The course integrates different styles of learning by integrating different learning modes. The classes include lecture/demonstration, discussion, in-class laboratories and a final individual project encompassing many course elements. The constant connection between the physics and the music, along with varied learning techniques, including hands-on experience, provides a motivating approach for non-science majors to experience science in a familiar context.

2:40

2pMU3. Fostering research in a general education acoustics course. Peter L. Hoekje (Physics and Astronomy, Baldwin Wallace University, 275 Eastland Rd., Berea, OH 44017, phoekje@bw.edu)

The general education science course often is the last science course an undergraduate takes. Liberal arts core requirements may include the goal that students come to understand or appreciate the nature of science and its impact on society. Incorporating research experience may be desirable, but has its challenges in a course for the non-major. A basic toolkit for student inquiry gives the capabilities for sound editing and analysis, filtering, sound level monitoring, sound and music synthesis, vowel and speech analysis, musical instrument design, and room acoustics analysis. These tools are available as low cost or free software. A course design compromise must be reached between delivery of content and support of student inquiry. However, a class of forty pursuing independent research projects in acoustics makes a rewarding experience for both instructor and students!

3:00

2pMU4. Acoustics at Berklee College of Music and elsewhere. Anthony K. Hoover (McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362, thoover@mchinc.com)

Students in the acoustics class at Berklee College of Music were non-science majors, but the course has done well, spanning over 50 semesters, and a number of those students have continued into graduate school and careers in acoustics. This paper will focus on the design and goals of the class, and will also discuss related experiences of teaching non-scientists in seminars, short courses, courses at other colleges, and the ASA-sponsored Continuing Education short course for architects.

Contributed Papers

3:20

2pMU5. Beats, ratios, and commas: Teaching tuning and temperament to music students. Stewart Carter (Music, Wake Forest University, 1833 Faculty Dr., Winston-Salem, NC 27106, carter@wfu.edu) and Jack Dostal (Physics, Wake Forest University, Winston-Salem, NC)

Undergraduate music students have many preconceived notions about tuning and temperament. For most of them, a piano tuned in equal temperament is perfectly “in tune” and A4 = 440 Hz is the only pitch standard that has ever existed. We find these naïve conceptions about tuning and temperament to be common among our introductory Physics of Music students–music majors, students pursuing majors in the sciences, and other non-science majors. In this paper we describe how we address some of these issues. We teach the mathematical underpinnings of equal, Pythagorean, just, and meantone temperaments, as well as their musical advantages and disadvantages, through the use of live demonstrations and recorded examples. We also describe our future plans to use recorded examples and available local resources—including reproductions of Baroque woodwinds tuned to A4 = 415 Hz and a 1799 organ at A4 = 409 Hz—to give students a broader understanding of historical and modern pitch standards.

3:35

2pMU6. Teaching and learning musical acoustics in a "music technology" program. Robert C. Maher (Electrical & Computer Engineering, Montana State University, 610 Cobleigh Hall, PO Box 173780, Bozeman, MT 59717-3780, rob.maher@montana.edu)

Helping math-averse students to learn musical acoustics can be challenging. Many universities offer a degree program entitled “Music Technology.” While the details of Music Tech programs differ, the general curricular focus includes music theory, electronic and computer music, audio recording and mixing, multimedia production, and computer applications in music composition. Music Tech programs also typically require a “Science of Sound” course that is intended to introduce the physical and mathematical concepts of musical sound and the basic principles of room acoustics. But unlike traditional acoustics students with a physics or engineering background, many students in the music tech programs are unconventional learners who do not tend to have much background in mathematics and the traditional science topics that would be the customary prerequisites for a formal acoustics class. This situation provides an interesting challenge–and a great opportunity–for the instructor to help students learn about acoustics and audio topics while side-stepping the students’ disdain for mathematical formulae. This paper presents one Music Tech lesson example: teaching musical instrument acoustics using a lumped source-filter-coupling model. The learning outcomes are sufficient for the students to understand and implement useful empirical models and simulations.

3:50

2pMU7. Measuring brass instruments: A “Physics of Music” lab exercise. Randy Worland (Physics, University of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Among the various aspects of brass instruments studied in a Physics of Music class are physical dimensions such as tube length and bore profile. Although textbooks list values for instrument lengths and describe the significance of the cylindrical, conical, and flared sections of tubing, these parameters are not visually obvious in the coiled instrument. A laboratory exercise is described in which students make simple length and diameter measurements of real brass instruments. Plots of tube diameter vs. distance are produced that effectively display what each instrument would look like if straightened out. Students also determine the air paths through valves (which are surprisingly difficult to visualize), and measure the added tubing associated with each valve. Mouthpieces are examined and measured in order to demonstrate the complex set of shapes involved. These hands-on measurements force students to look closely at the component pieces, air paths, and construction details of the instruments. Pedagogically, the use of real instruments in the lab engages students and helps them make connections between the physics of ideal tubes and the design of real brass instruments. Student results and feedback will be described.

4:05

2pMU8. Introducing the concept of resonance to non-science majors in a musical acoustics class. Andrew Morrison (Joliet Junior College, 1215 Houbolt Rd, Natural Science Department, Joliet, IL 60431, amorrison@jjc.edu)

Understanding the concept of resonance is one of the most important goals for a student in a musical acoustics course to achieve. In our musical acoustics course students examine the standing wave behavior of simple one-dimensional systems such as stretched strings and pipes with open ends or one closed end. The concept is integral to the discussion of how nearly all musical instrument work, but it is also a difficult concept for many students to understand. In our class, we discuss how looking at the boundary conditions of the systems we examine in class can be applied to predict what the resonant frequencies of the system are.

4:20

2pMU9. An integrated lecture-laboratory approach to teaching musical acoustics. Andrew A. Piacsek (Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

At Central Washington University, "Physics of Musical Sound" (PHYS103) is an introductory level course that satisfies a General Education Breadth requirement in the lab-based science category. Enrollment is capped at 40 students, of which 10-15 are typically music education majors. From 1998 to 2010, this five-credit course was taught in a traditional lecture format: four 50-minute lectures and a two-hour lab each week. Starting in 2011, the course was reformatted into three two-hour periods. The objective was to incorporate inquiry-based learning strategies adopted from physics education research. The two-hour meeting time provides flexibility to structure the class with a combination of lecture, tutorial, group problem solving, and experiment. Computational resources, including sound analysis software and online interactive simulations, are heavily utilized. Example activities will be described and lessons learned from the transition to the new format will be discussed.

4:35

2pMU10. Physics of music for musicians, architects, and science students: Designing a multi-level platform for an undergraduate course. Robert G. Hall (Music, Laurentian University, 935 Ramsey Lake Rd., Department of Music, Sudbury, ON P3E2C6, Canada, rhall@laurentian.ca)

This presentation will describe the development of a new course in the Physics of Music at Laurentian University in Sudbury, Ontario. While the creation of the course was spurred on by the opening of a new School of Architecture in 2013 and the desire to provide an elective containing acoustics for that program, the project also has the added challenges of: developing a course that is available to music students (many of whom have little or no science background), creating a course that is an attractive elective for students across campus, initiating a course that meets the elective requirements for engineering students, and building a course that contains sufficient physics so that it may be cross-listed as a physics course. The presentation will outline the specifics of each of the required parameters and give specifics as to the multi-level project choices within the course that have been developed to allow the different student constituencies to satisfy their pedagogical needs. Also addressed will be the requisite teaching methods used to allow on-line participation, while still encouraging and requiring attendance at the weekly lectures.

TUESDAY AFTERNOON, 23 OCTOBER 2012 TRIANON C/D, 1:30 P.M. TO 5:00 P.M.

Session 2pNS

Noise and Architectural Acoustics: Soundscape Workshop

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Sq., Vernon, CT 06066

Jian Kang, Cochair
School of Architecture, Univ. of Sheffield, Sheffield S10 2TN, United Kingdom

Chair's Introduction—1:30

Invited Papers

1:35

2pNS1. Introduction to workshop goals. Bennett M. Brooks (Brooks Acoustics Corporation, 30 Lafayette Square - Suite 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com), Brigitte Schulte-Fortkamp (Technical University Berlin, Berlin, Berlin, Germany), and Jian Kang (School of Architecture, University of Sheffield, Sheffield, South Yorkshire, United Kingdom)

The goal of this workshop is to work toward methodology standardization for the advancement of the developing field of soundscape measurement, analysis and design. The workshop will focus on the terminology lexicon used for soundscape subject interviews. Interviews of local experts, residents and other users and inhabitants of the sonic environment can yield insights into both personal reactions and universal observations. The specific terminology used in this process may significantly affect the outcomes. As the success of a new research or development project can depend on the lessons learned from previous projects, the standardization of interview techniques becomes increasingly important. Workshop participants are invited to develop a standardized lexicon of descriptors for field use in interview questionnaires. The lexicon development will be based on the results of earlier ASA soundscape workshops, and the concurrent activities in ISO and COST Working Groups. An introductory keynote address will review the topics, objectives, and procedures for the day's discussion. The participants will then break out into smaller subgroups to review key issues and to assess the available lexicon terms. The entire group will reassemble to report, assess, and prioritize the proposed actions.

2:00–3:00 Break Out Session

3:00–3:30 Break

3:30–4:30 Break Out Session

4:30–5:00 Panel Discussion

Session 2pPA

Physical Acoustics: Waves in Heterogeneous Solids II

Joseph A. Turner, Cochair
Dept. of Mechanical and Materials Engineering, Univ. of Nebraska-Lincoln, Lincoln, NE 68588-0526

Goutam Ghoshal, Cochair
Electrical and Computer Engineering, University of Illinois at Urbana-Champaign, Urbana, IL 61801

Invited Papers

1:00

2pPA1. Ultrasonic scattering for inverse characterization of complex microstructures. Stanislav I. Rokhlin (Department of Material Science and Engineering, The Ohio State University, Edison Joining Technology Center, 1248 Arthur E Adams Drive, Columbus, OH 43220, rokhlin.2@osu.edu)

This paper overviews resent theoretical and experimental results for inverse characterization of duplex polycrystalline microstructures from measurements of ultrasonic attenuation and backscattering. The duplex microstructures are formed by elongated regions of clustering crystallites with preferred orientations and are typical for titanium alloys. New insights into the dependences of the backscattering and attenuation on frequency and averaged ellipsoidal grain radii are obtained. In particular the dominant effect of the averaged ellipsoidal radius in the direction of wave propagation, instead of the ellipsoidal cross-section, is shown. Also the opposite effect on attenuation and backscattering of grain size in the direction of wave propagation is demonstrated by theory and experiment. Both attenuation and backscattering depend on effective elastic properties of clusters, their size and orientation and the measurement system parameters. It is shown that for material property inversion it is advantageous to decouple elastic and geometrical properties by a special series of measurements and data inversion: first determine grain geometry and size from relative directional and/or attenuation-to-backscattering ratio measurements; this is followed by absolute attenuation and backscattering measurements to obtain the crystallite orientation distribution function in the clusters.

1:20

2pPA2. Ambient noise seismic monitoring of the continuous deformation of the Earth. Michel Campillo (ISTerre, Université Joseph Fourier, BP 53, Grenoble 38041, France, Michel.Campillo@ujf-grenoble.fr)

The analogy between the Green function and the long-range correlation of the seismic ambient noise leads to new developments in imaging and monitoring. In the last years this approach allows for high-resolution surface wave tomography, and more recently its potential for body wave imaging was demonstrated. In the same time, the reconstruction of the scattered part of the Green function was analyzed and its possible use to detect slight variations in the medium was confirmed. Ambient noise monitoring allows for a continuous measure of seismic velocity changes related with the tectonic process affecting the lithosphere. I present some examples showing that a change at depth can be monitor with seismic noise records and that those changes are related to the deformation. We analyze in detail the transient deformation on a subduction zone and we discuss the relations between change of seismic velocity, slow slip events and tremors.

1:40

2pPA3. Multiple scattering in the spirit of Leslie Foldy. Paul A. Martin (Applied Mathematics and Statistics, Colorado School of Mines, Dept of Applied Math and Statistics, Colorado School of Mines, Golden, CO 80401-1887, pamartin@mines.edu)

Foldy's name is best known to wave theorists because of his 1945 paper on multiple scattering. This came out of wartime work on sound propagation through bubbly liquids. The paper itself contains a deterministic theory for scattering by a finite number of small objects, and a probabilistic theory for wave propagation through random arrangements of many small scatterers in which a certain closure assumption is invoked. Assumptions of this kind can be said to be unreasonably effective. We describe some of this early work and then review more recent work on a variety of multiple scattering problems.

Contributed Papers

2:00

2pPA4. Acoustic scattering from weakly coupled porous media. Max Denis, Kavitha Chandra, and Charles Thompson (University of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, max_f_denis@hotmail.com)

In this work, wave propagation and multiple acoustic scattering from porous media are examined. For a highly porous media, the inertial coupling between solid and fluid phases can be small. In such a case the dilatational and acoustic motions will be weakly coupled. The controlling factor is the ratio of effective inertial mass density of the porous media and the fluid mass density. Using Biot's analysis as the starting point, the dilatational and acoustic motions, as well as the scattered pressure that ensues are evaluated. The Kirchhoff-Helmholtz integral equation is used to this end. The integral equations are evaluated using a series expansion in terms of derived acoustic contrast properties. Using the local Padé approximant procedure detailed in [The Journal of the Acoustical Society of America, 128(5) EL274-278 (2010)]. Of particular interest, are high contrast media and wavelengths comparable to the average scatterer size.

2:15

2pPA5. Characterization of polycrystals by ultrasonic attenuation-to-backscattering ratio measurements. Jia Li and Stanislav I. Rokhlin (Department of Material Science and Engineering, The Ohio State University, Edison Joining Technology Center, 1248 Arthur E Adams Drive, Columbus, OH 43220, rokhlin.2@osu.edu)

Scattering of ultrasonic waves in polycrystalline materials depends on the relative misorientation of the crystallites, their elastic properties and the crystallite size and morphology. Those parameters affect both ultrasonic attenuation and backscattering. In structural material alloys the elastic properties of crystallites and microtextures are often unknown, thus complicating model-based microstructure size determination from ultrasonic measurements. It has been shown by analysis and simulation that the measured attenuation-to-backscattering ratio is independent of the unknown elastic material characteristics and thus may be used advantageously for inverse determination of mictostructural characterization of polycrystalline materials. The method is based on recently developed models for ultrasonic attenuation and backscattering in polycrystals with nonequiaxed grains of general ellipsoidal shape. Using the model approximation an effective size parameter is defined and obtained from the backscattering-to-attenuation ratio measurements. By this method the microstructure size from one side of the sample can be characterized. The method is demonstrated on experimental data for Ti alloy samples with duplex microstructure/microtexture formed by hexagonal alpha crystallites. Volumetric ultrasonic results are compared with direct surface microtexture measurements by orientation image microscopy.

2:30

2pPA6. Ultrasonic characterization of polymeric composites containing auxetic inclusions. Michael R. Haberman, Daniel T. Hook, Timothy D. Klatt (Applied Research Laboratories, University of Texas, 10000 Burnet Rd, Austin, TX 78758, haberman@arlut.utexas.edu), Trishian A. M. Hewage, Andrew Alderson, Kim L. Alderson (Institute for Materials Research and Innovation, University of Bolton, Bolton, 5AB, United Kingdom), and Fabrizio L. Scarpa (Bristol Laboratory of Advanced Dynamics Engineering (BLADE), Department of Aerospace Engineering, University of Bristol, Bristol, 1TR, United Kingdom)

Composite materials are often used as damping treatments or structural materials to mitigate the effects of unwanted vibration and sound. Recent work on materials displaying a negative Poisson's ratio, known as auxetic materials, indicate that composite materials consisting of a lossy matrix containing auxetic inclusions may lead to improved vibro-acoustic absorption capacity compared to composites containing positive Poisson's ratio inclusions. This work presents ultrasonic measurements of an epoxy matrix material (Epon E828/D400) containing volume fractions of α-cristobalite inclusions ranging from 5% to 25% by volume. The effective frequency dependent speed of sound and attenuation coefficient of each sample is measured from 1 to 10 MHz using ultrasonic immersion techniques. Ultrasonic test results are compared with Dynamic Mechanical Thermal Analysis and modal damping measurements of stiffness and loss behavior. This material is based upon work supported by the U. S. Army Research Office under grant number W911NF-11-1-0032.

2:45

2pPA7. Estimation the mean correlation length of metals by using mode-converted diffuse ultrasonic backscatter model. Ping Hu and Joseph A. Turner (Department of Mechanical and Materials Engineering, University of Nebraska-Lincoln, Lincoln, NE 68588, ping.hu@huskers.unl.edu)

Diffuse ultrasonic backscatter measurements can explain the phenomenon of the scattering at interfaces in heterogeneous materials. The theoretical models of diffuse ultrasonic backscatter for longitudinal (L-L) and shear (T-T) wave scattering within polycrystalline materials have recently been developed. A mean correlation length of metals is successfully calculated from the L-L model at normal incidence in a pulse-echo inspection. Above the first critical angle, mode-conversion scattering occurs when the longitudinal waves are converted to shear waves (L-T) at material grain boundaries. With a similar formalism, a mode-conversion scattering (L-T) model is presented to describe this process. The model is then to fit the experimental response for a pitch-catch transducer configuration and the correlation length is extracted by modifying the beam function. The mean correlation length from the L-T model is in agreement with both the L-L model and the results obtained from optical micrographs. This presentation outlines the theoretical framework and the method to extract the mean correlation length. Mode-converted backscatter removes the influence of the front-wall reflection and may lead to improvements in microstructural characterization and material property evaluation. [Research supported by FRA.]

3:00–3:15 Break

3:15

2pPA8. Detection of inclusions embedded in a polycrystalline medium: A diagrammatic approach. Lucas W. Koester and Joseph A. Turner (Mechanical and Materials Engineering, University of Nebraska-Lincoln, Lincoln, NE 68588, lucaswkoester@huskers.unl.edu)

Diffuse ultrasonic backscatter techniques are used to evaluate and to quantify polycrystalline microstructure in structural materials including steel, aluminum, and concrete. Modeling of the backscatter received from grain scattering is important for both microstructural characterization and flaw detection. Here, the problem of an inclusion embedded in a polycrystalline background is examined using a diagrammatic approach. Expressions are derived that for the attenuation and diffuse backscatter using a Green's function approach that incorporates Feynman diagrams for simplicity in the analysis. The resulting equations are solved in the single-scattering limit that couples geometric probabilities and correlation functions such that the impact of the inclusion can be quantified. The effects of frequency, average grain size, polycrystalline properties, inclusion properties and inclusion size are considered. Further work related to inspection with focused beams or immersion type ultrasonic inspections is also presented. This work is anticipated to impact the studies of model-assisted probability and inclusion content quantification in ultrasonic non-destructive evaluation.

3:30

2pPA9. Diffuse ultrasonic backscatter measurements for monitoring stress in rail. Devin T. Walentine, Christopher M. Kube, Joseph A. Turner (Mechanical and Materials Engineering, University of Nebraska, Lincoln, NE 68588, d.t.walentine@gmail.com), and Mahmood Fateh (Office of Research and Development, Federal Railroad Administration, Washington DC, Nebraska)

Recent research in polycrystalline materials, both theoretically and experimentally, has demonstrated a dependence of diffuse ultrasonic backscatter (DUB) on applied stress. This dependence has been used to develop a measurement device for monitoring longitudinal stress in continuous welded rail (CWR). However when moving this research from the laboratory setting to field applications many additional challenges are encountered with respect to the experimental method. In this presentation, results from initial field experiments are presented. This work is performed on a switch track from a railroad mainline to a short line railroad siding. Comparison between DUB measurements and actual stress in the rail is accomplished through the use of four stress modules (based on standard strain gauge technology) that were installed at four different locations along test site. In these field tests, DUB experiments using two orthogonal shear waves are investigated. Procedures to mitigate errors in our experiments and techniques to refine the measurement for more accurate results are discussed. These field experiments highlight the utility of this approach with respect to practical and financial considerations for determining induced stresses in CWR. [Research supported by FRA.]

3:45

2pPA10. Influence of stresses on grain size and attenuation parameters in ultrasonic scattering models. Christopher M. Kube and Joseph A. Turner (University of Nebraska-Lincoln, Lincoln, NE 68588, ckube@huskers.unl.edu)

Recent findings have shown the dependence of material stresses on the strength of ultrasonic backscatter. The stress dependence arrives when considering the covariance of effective elastic moduli of a medium with a randomly oriented grain structure. In some instances, the change in magnitude of the scattered response due to stress is significant and can be utilized as a technique for NDT stress monitoring. Conversely, ignoring the stress sate could result in significant error when attempting to extract micrsotructural parameters using ultrasonic NDE techniques. This presentation touches on the error generated in estimating the average grain size when using ultrasonic scattering models which include stress dependent backscatter coefficients. Results are given for three scattering modes; longitudinal to longitudinal, shear to shear, and (mode converted) longitudinal to shear under a variety of loading configurations. Furthermore, the stress dependence in the covariance influences common multiple echo attenuation measurements when grain scattering is present. Theoretical attenuation values of the different modes are given for various stress-states. Understanding how stress influences these parameters can potentially improve existing ultrasonic NDT techniques. [Research supported by the Federal Railroad Administration.]

4:00

2pPA11. Vibrations of composite bimorph cantilever with multidomain structures. David Sedorook (Department of Physics and Astronomy, University of Mississippi, University, MS), Andrew Nadtochiy (Department of Physics, Kyiv Shevchenko University, Kyiv, Ukraine), and Igor Ostrovskii (Dept. of Physics and NCPA, Univ of Mississippi, Lewis Hall, Rm. 108, University, MS 38677, iostrov@phy.olemiss.edu)

The vibrating piezoelectric cantilevers are important objects in contemporary physical acoustics and their applications including the hot topic of robotic flying insects, MEMS, and various actuators. In this work, we study different cantilevers consisting of the piezoelectric layers and a thin steel layer. The three-layer bimorph composite cantilever includes a central steel sheet covered with two oppositely polarized coatings made of a piezoelectric ceramic. We compare the amplitudes of tip vibrations in the bimorph composite structures with the simpler cases of a single piezoelectric layer with one to three ferroelectric domains sitting on a steel substrate as well as without the steel support. Theoretical calculations of the amplitude of acoustical vibrations of the actuator tip are performed with the help of a finite element method developed for piezoelectric media. The corresponding codes were verified and adopted by the experimental measurements from the 12-mm long and 0.5-mm thick PZT-cantilever vibrating on its fundamental mode at frequency 1.12 kHz. The vibration amplitude of a cantilever with 2 or 3 periodically poled domains is the highest at the 2nd or 3rd resonance frequency, respectively.

4:15

2pPA12. Spatial variation of ultrasonic attenuation coefficient in freshly excised human calvarium. Armando Garcia, Joseph A. Turner (Mechanical and Materials Engineering, University of Nebraska-Lincoln, Lincoln, NE 68588-0526, aagn17@gmail.com), and Robert S. Salzar (Center for Applied Biomechanics, University of Virginia, Charlottesville, VA)

The propagation of ultrasound through human calvarium poses a great challenge for transcranial diagnosis and treatment of several medical conditions. Moreover, better understanding of how sound waves are attenuated in the human calvarium is gaining relevance with the recent awareness of the problem of blast wave induced traumatic brain injury. In the present study, the spatial variability of ultrasonic properties was evaluated for relevant frequencies of 0.5, 1, and 2.25 MHz. A total of eighteen specimens from four donors were tested using a through-transmission configuration. With the aid of a two interface model, the ultrasonic attenuation coefficient was determined from the total energy loss at various locations on the specimens. Mean volumetric densities at various locations on the samples were determined from computed tomography images. The results show good correlation between attenuation and volumetric density, particularly for the higher frequencies. In addition, the spatial variability of the attenuation, within a single person and with respect to different people, was found to be much larger than expected. These results are anticipated to have a major impact on transcranial biomedical research. [Support of the Army Research Office for this work is gratefully acknowledged.]

4:30

2pPA13. Exact image theory for a three layer medium. Ambika Bhatta, Charles Thompson, Kavitha Chandra (Electrical and Computer Engineering, University of Massachusetts, 1 University Ave, Lowell, MA 01854, ambika_bhatta@student.uml.edu), and Vineet Mehta (MIT Lincoln Laboratory, Lexington, MA)

In this work an exact formulation and solution for the image source amplitude in a three layer medium is given. To do so a novel generalization of the Sommerfeld integral is employed. Of particular interest in this paper is the time-domain solution for the impulse response between source and observation points in the middle layer. It is shown that global boundary conditions can be accommodated for multiple reflections from media of having infinite extent. A branch integral formulation of inverse Laplace transform of integral powers of the Fresnel reflection coefficient is given for this case. It is shown that only odd-powers of the branch point argument contribute thereby reducing computational effort required to numerically evaluate the impulse response.

4:45

2pPA14. A sonar experiment to study sound propagation through flames. Mustafa Z. Abbasi, Preston S. Wilson (Mech. Eng. Dept. and Applied Res. Labs., University of Texas, Austin, TX 78712, mustafa_abbasi@utexas.edu), Ofodike A. Ezekoye, and Joelle I. Suits (Mech. Eng. Dept., Univ. of Texas at Austin, Austin, TX)

Disorientation is a major cause of firefighter death and injury. When a firefighter is trapped in a structure, there is a small window of time for rescue teams to locate a downed firefighter. We propose an acoustic navigation system based on sonar concepts to augment existing techniques (e.g. thermal imaging cameras). As a first step toward this system, a better understanding of acoustic wave propagation through the fire environment is needed. Therefore, a sonar experiment was developed to measure the distance through a flame. Since information in the literature suggests that transmission loss through fire increases with frequency, a parametric array was used to maintain narrow source directivity while remaining in the low frequency/low attenuation regime. Results of the sonar experiment will be presented for various source functions and flame conditions.

Session 2pSA

Structural Acoustics and Vibration: Guided Waves for Nondestructive Evaluation and Structural Health Monitoring II

Henrique Reis, Cochair
Industrial and Enterprise Systems Engineering, University of Illinois at Urbana-Champaign, Urbana, IL 61801

Michael J. Lowe, Cochair
Mechanical Engineering, Imperial College London, London SW7 2AZ, United Kingdom

Invited Papers

1:00

2pSA1. Integral structural health monitoring of helicopter tail booms by guided acoustic waves. Wolfgang Grill (Institute of Experimental Physics II, University of Leipzig, Linnéstr. 5, Leipzig, Germany 04312, Germany, grill@physik.uni-leipzig.de)

The general principles of structural health monitoring by ultrasonic waves with high resolution monitoring of the time dependence of the transport properties are presented. Reported are essential aspects of the theoretical background together with the results of modeling relating to the developed monitoring scheme and the experimental findings. The presented applications of the developed integral structural health monitoring concentrate on examples involving helicopter tails booms. These include conventional aluminum based bolted frame and stringer constructions as well as fiber compound structural components with foam inlays, in both cases for helicopter tail booms. The presentation includes experimental demonstrations on samples suitable for transport.

1:25

2pSA2. Hybrid microcontinuum field approach for intrinsic damage state quantification. Sourav Banerjee (Mechanical Engineering, University of South Carolina, 300 Main Street, Columbia, SC 29208, banerjes@cec.sc.edu)

The primary objective of this presentation is to demonstrate a systematic framework for the early stage diagnostics of materials under operation. Defying the conventional route of 'bottoms-up' multi-scale modeling approach for material diagnostics, a hybrid 'top-down' approach is presented. Structure specific diagnostics and prognostics have become extremely important due to variances in the real life performances of aerospace materials. Structures, structural components and engine components will need individual attention in the near future. Individual attention means that a parallel database must be maintained for each component for on-line digital certification. The 'digital-twin' software will have access to structure's detailed fleet and maintenance records, nondestructive test results, real-time sensory data etc. to certify the materials on-line, but it will need some 'quantifiable' parameters. Thus a parameter to quantify the incubation of damage at meso-scale is identified. The intrinsic length scale dependent 'parameter called 'damage entropy' closely resembles the material state due to fatigue, extreme environments, operational hazards or spatio-temporal variability. The proposed quantification process involves physics and statistics based predictive models. In this novel approach 'micromorphic' description of material is considered in spatio-temporal non-local sense and the nonlocal features are extracted from the real time signals for damage state quantification.

1:50

2pSA3. Mesh-free distributed point source method for modeling guided wave propagation in a viscous fluid layer trapped between two solid plates. Yuji Wada (Precision and Intelligence Laboratory, Tokyo Institute of Technology, Yokohama, Nagatsuta, Midori-ku, Japan), Tribikram Kundu, and Kentaro Nakamura (Civil Engineering and Engineering Mechanics, University of Arizona, 1209 E. 2nd Street, Bldg # 72, Tucson, AZ 85721, tkundu@email.arizona.edu)

Distributed Point Source Method (DPSM) is extended to model wave propagation in viscous fluids. Appropriate estimation on attenuation and boundary layer due to fluid viscosity is necessary for the ultrasonic devices that utilize acoustic streaming or ultrasonic levitation. Since the boundary layer is often much thinner than the wavelength, numerical simulations based on the finite element method suffer from large computational cost because very fine mesh is needed to trace the layer. DPSM can efficiently model this problem. The computational cost for modeling the viscous fluid with DPSM is reduced close to the cost of non-viscous fluid analysis. In this paper, equations for DPSM modeling in viscous fluids are derived by decomposing the linearized viscous fluid equations into two components - dilatational and rotational. By considering complex P-wave and S-wave numbers, the sound fields in viscous fluids can be calculated using the same calculation routines used for waves in solids. To verify the calculation precision, a comparison between approximated theory and DPSM generated results for a fundamental ultrasonic field problem is performed. The particle velocity profile parallel to the surface in a viscous fluid between two vibrating plates is calculated. Theoretical results agree well with the DPSM generated results.

2:15

2pSA4. Active structural health monitoring during sub-orbital space flight. Andrei N. Zagrai, William Reiser, Brandon Runnels, Chris White, Abraham Light-Marquez, Stephen Marinsek, Andrew Murray (Mechanical Engineering, New Mexico Institute of Mining and Technology, 801 Leroy Pl., Socorro, NM 87801, azagrai@nmt.edu), Stuart Stuart Taylor (The Engineering Institute, Los Alamos National Laboratory, Socorro, NM), Gyuhae Park, and Charles Farrar (The Engineering Institute, Los Alamos National Laboratory, Los Alamos, NM)

Increasing involvement of commercial enterprises in space activities is among leading forces behind a renewed interest in structural diagnostic methodologies promising potential for improving safety, operability and cost effectiveness of launch vehicles and spaceships. It is envisioned that unobtrusive, real time structural health monitoring (SHM) systems may assist in space vehicle's prelaunch qualification, orbital operation, safe disintegration during reentry or recertification for a next flight. SHM experiment utilizing piezoelectric wafer active sensors in conjunction with electro-mechanical impedance measurements has been develop to explore feasibility of active structural health monitoring during suborbital space flight. Details of experiment are discussed and some results obtained in real time for all segments of vehicle's trajectory are presented. Experimental data collected during suborbital space flight has shown feasibility of SHM in the challenging environment, utility of thin wafer piezoelectric sensors as active elements of spacecraft SHM system, and potential of the electro-mechanical impedance method for real time structural integrity assessment of the payload.

2:40

2pSA5. Modeling and signal processing for guided wave structural health monitoring. Anthony Croxford and Paul D. Wilcox (University of Bristol, Queens Building, Bristol, Avon BS8 1TR, United Kingdom, a.j.croxford@bristol.ac.uk)

Guided wave approaches are a key technique for the implementation of structural health monitoring with a sparse sensor array, offering good sensitivity over a reasonable range. There are however several difficulties, particularly how the system can be modelled to predict performance and how acquired data can be used most effectively post capture. This paper looks at both these issues, starting with a study of the modelling of the scattering of guided waves from features. Models of both the scattering from structural features and predicted damage types can then be used to determine the sensitivity of any network and the process of their derivation and use is described here. In order to maximise this sensitivity it is important to use the data in the most efficient way possible. This is accomplished through the use of statistical models of the changes in the signal and fitting maximum likelihood estimators to these. In choosing a good model of the signal a statistical detector can be designed to optimise the performance for a given use case, be that detection or localisation.

3:05–3:20 Break

Contributed Papers

3:20

2pSA6. Elastic wave propagation in coated pipelines. Ray Kirby, Zahari Zlatev (Mechanical Engineering, Brunel University, Uxbridge, Middlesex UB8 3PH, United Kingdom, ray.kirby@brunel.ac.uk), and Peter Mudge (NDT Technology Group, TWI Ltd., Cambridge, United Kingdom)

Guided elastic waves are commonly used in the non-destructive evaluation of oil and gas pipelines. In order to protect pipes from environmental damage a viscoelastic coating such as bitumen is often applied. Viscoelastic coatings do, however, attenuate travelling waves and it is not uncommon in a coated pipe for the signal reflected from a defect to be attenuated to the extent that it is no longer discernable above the background noise. Thus, viscoelastic coatings significantly impact on the effectiveness of non-destructive testing in pipelines, both in the ability to resolve defects but also in the length of pipeline that may be tested. In order to obtain a better understanding of guided wave propagation in coated pipes, a numerical model is presented here that seeks to analyse the propagation of torsional and longitudinal modes in a coated pipe with simple axisymmetric defects. Reflection coefficient predictions are compared with experimental data for a pipe coated with bitumen and good agreement is observed between the two, although only after first undertaking a curve fitting exercise to identity appropriate values for the phase speed and attenuation of bulk torsional waves in the viscoelastic material.

3:35

2pSA7. Characterization of porous materials using ultrasonic slow wave. Lin Lin (Engineering, University of Southern Maine, 37 college Ave., 131 John Mitchell center, Gorham, ME 04038, llin@usm.maine.edu), Michael Peterson (Mechanical Engineering, University of Maine, Orono, ME), and Alan Greenberg (Mechanical Engineering, University of Colorado at Boulder, Boulder, CO)

Porous materials are critical engineering components in many process industries. Although porous materials have been successfully utilized in many areas, characterization of porous structures is still a significant problem limiting the applications of porous materials, especially when the application involves the change of porous structure. Ultrasonic techniques have been reported for successful applications on material characterization, including porous materials. This research utilized an acoustic technique for permeability measurement by measuring the critical wave number of Biot's slow longitudinal wave. The slow longitudinal wave can serve as an important tool for investigating the ability of fluids to penetrate into porous materials. Since the slow longitudinal wave is associated with out-of-phase movement of the pore fluid relative to the matrix structure, it is very sensitive to the permeability of the porous formation. In Biot's theory the slow wave is highly attenuated below a critical frequency. The critical wave number can be directly related to the permeability of porous materials. The measurement of the critical wave number provides a unique method to obtain the permeability measurement, which can be apply to structure monitoring, quality control, etc.

3:50

2pSA8. Guided wave approach for inline photovoltaic module component inspection. Rico Meier (Module Reliability, Fraunhofer CSP, Walter-Huelse-Strasse 1, Halle, Saxony-Anhalt 06120, Germany, rico.meier@csp.fraunhofer.de)

Reliability of photovoltaic modules is one key factor for being financially attractive for customers all over the world. The further reduction in manufacturing costs lead to increased demands on module components and their materials to maintain acceptable mechanical yields and module reliability. Thus fast, economic and preferably non-destructive component characterization and manufacturing process control methods come more and more into focus. In the present work lamb wave based approaches for material characterization of plate-like photovoltaic module components were evaluated with respect to their precision and inline capabilities. The second-order elastic constants (Young's modulus and Poisson's ratio) were determined by automated numeric fitting of the Rayleigh-Lamb dispersion model on experimental data. Furthermore, the influence of mechanical stress on the ultrasound transport properties was investigated. Therefore the components

were systematically loaded by mechanical and thermal stress. The resulting changes in ultrasound transport were correlated with results from mechanical testing. Lamb wave approaches turned out to be well suited methods for inline material characterization of plate-like module components. The elastic constants can be determined with high accuracy. The usage of plate-like structures in photovoltaic modules makes lamb waves an important tool for inline and non-destructive material characterization.

4:05

2pSA9. Determination of leaky Lamb wave modal attenuation coefficient on a solid plate. Wan-Gu Kim and Suk Wang Yoon (Dept. of Physics, SungKyunKwan University, 300 Chunchun-dong, Jangan-ku, Suwon, Suwon 440-746, Republic of Korea, swyoon@skku.ac.kr)

Leaky Lamb wave is practically important in an immersion technique of nondestructive evaluation for a solid plate. It is necessary to determine its modal attenuation coefficient in order to evaluate plate defects in industry and to diagnose osteoporosis in medical applications. In this study we present a method to determine the modal attenuation coefficient of leaky Lamb wave using two-dimensional Fourier filtering. A complex leaky Lamb wave signal from a solid plate can be decomposed into several modal signals on frequency-space domain through two-dimensional Fourier filtering. It makes possible to experimentally determine the modal attenuation coefficient along an aluminum plate in water. It is well explained by theoretical estimation with the dispersion relation of leaky Lamb wave.

4:20

2pSA10. Simultaneous thickness, velocity, density, and attenuation measurement of a thin layer by time-resolved acoustic microscopy. Jian Chen, Xiaolong Bai, Keji Yang, Bingfeng Ju, and Jianxing Meng (The State Key Laboratory of Fluid Power Transmission and Control, Zhejiang University, Yuquan Campus, Hangzhou, Zhejiang Province 310027, China, yi03072004@stu.xjtu.edu.cn)

An ultrasonic method for simultaneous determination of thickness, velocity, density and attenuation of thin layer using a time-resolved acoustic microscopy is proposed. Reflection from the thin layer is represented as a function of three dimensionless parameters which are determined from the experimentally normal incidence component of the two dimensional reflection spectrum $R(\theta,\omega)$ derived based on the inversion of V(z,t) technique with time-resolved acoustic microscopy. The thickness of the thin layer is derived from the signals received from the layer itself considering the geometrical relations when the lens focused on the sample's surfaces. The simultaneous determination of thickness, velocity, density and attenuation of thin stainless steel plate by using a point-focusing transducer with nominal frequency of around 50MHz were carried out. The determined material properties are comparable, in which the thickness, velocity and density can be measured with a percentage biases less than 5% and the attenuation is close to its real value. The present preliminary work shows the high efficiency, viability and capability of the new non-destructive technique in simultaneously characterize basic mechanical and geometrical properties of thin layers.

4:35

2pSA11. Simultaneous measurement of local longitudinal and transverse wave velocities, attenuation, density, and thickness of a layer by using point-focus ultrasonic spectroscopy. Jian Chen, Xiaolong Bai, Bingfeng Ju, and Jianxing Meng (The State Key Laboratory of Fluid Power Transmission and Control, Zhejiang University, Yuquan Campus, Hangzhou, Zhejiang Province 310027, China, yi03072004@stu.xjtu.edu.cn)

This paper presented an ultrasonic technique for simultaneous determination of the complete set of acoustical and geometrical properties of a layer embedded between two known materials using point-focus ultrasonic spectroscopy, which provides a high lateral resolution. The theoretical model of the two-dimensional spectrum Rt (θ, ω) of the layer is calculated as a function of six parameters of longitudinal and transverse velocities cl, ct, attenuation αl, αt, density ρ and thickness h, which fully determined the layer properties. The experimental spectrum Re (θ, ω) can be measured by V(z, t) technique. A two-step algorithm is presented to decompose the searching process of parameters from one six-dimensional to two three dimensional spaces. The sensitivity of the two-dimensional spectrum to individual properties and its stability against experimental noise are studied. The full set properties of a 250 μm thick stainless steel plate immersed in water are determined. The proposed technique used a point focus transducer, which makes the set-up simple and reliable. It allows measurement of the local properties of the layer and enables precision material characterization.

TUESDAY AFTERNOON, 23 OCTOBER 2012 TRUMAN A/B, 1:00 P.M. TO 5:00 P.M.

Session 2pSC

Speech Communication: Speech Perception I: Vowels and Consonants (Poster Session)

Yuwen Lai, Chair
Foreign Languages and Literatures, National Chiao-Tung University, Taiwan, Hsinchu 30010, Taiwan

Contributed Papers

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

2pSC1. Context matters: Analyzing the necessity of context-dependent speech perception in complex models. Keith S. Apfelbaum and Bob McMurray (Psychology, University of Iowa, Iowa City, IA 52242, keith-apfelbaum@uiowa.edu)

Speech perception faces variability in the realization of phonemes from vocal tract differences, coarticulation, and speaking rate. While the mechanisms accommodating this variability have long been debated, recent computational work suggests that listeners succeed by processing input relative to expectations about how context affects acoustics. McMurray and Jongman (2011) measured 24 cues for fricatives in 2880 tokens. When effects of talker voice and vowel context were removed with a relative encoding scheme, simple cue-integration/prototype models identified the fricatives at near-human levels of performance. These models were less successful with raw cue values, suggesting some form of compensation is required. However, contingent encoding may not be needed in all categorization architectures. Exemplar models account for context by holistically comparing incoming tokens to others in memory, and connectionist back-propagation models encode stimulus-specific information through their hidden units for better performance. We conducted several simulations using these classes of models. After training the models to identify fricatives from the McMurray and Jongman corpus, we assessed model performance on held-out tokens for which human categorization data was available. Both classes of models more closely matched human performance when trained on context-adjusted information than raw cues. This offers firmer support for context-dependent speech perception.

2pSC2. Voicing in English revisited: Measurement of acoustic features signaling word-medial voicing in trochees. Joseph C. Toscano (Beckman Institute for Advanced Science and Technology, University of Illinois at Urbana-Champaign, 405 N Mathews Ave, Urbana, IL 61801, jtoscano@illinois.edu) and Bob McMurray (Dept. of Psychology and Dept. of Communication Sciences & Disorders, University of Iowa, Iowa City, IA)

A great deal of work in speech has argued that invariant acoustic cues do not exist, leading many researchers to conclude that listeners use specialized representations, such as talkers' inferred gestures, instead. Other work has emphasized that many phonological distinctions are signaled by multiple cues; Lisker (1986, Language and Speech, 29, 3-11), for example, lists 16 cues to voicing. Yet, few studies have measured the reliability of multiple cues and asked whether combining them may provide a solution to the lack of invariance. Here, we present measurements of 12 potential cues to the voiced/voiceless distinction in stops (including many of those reported by Lisker, 1986) and use a statistical modeling approach to determine which ones distinguish the two categories and how reliable those cues are. We recorded two-syllable non-words (/sVCVs/) with one of six consonants (/b,p,d,t,g,k/) and one of four vowels (/a,æ,i,u/). We found that talkers used multiple cues, but that cues varied in their usefulness. In addition, a classifier trained on the cues was able to accurately identify voicing categories. We argue that by harnessing information from multiple cues, listeners can overcome ambiguity in individual cues in specific utterances, allowing them to recognize speech across talkers and phonological contexts.

2pSC3. Linguistic and social effects on perceptions of voice onset time in Korean stops. Robert Podesva, Annette D'Onofrio, Eric Acton, Sam Bowman, Jeremy Calder, Hsin-Chang Chen, Benjamin Lokshin, and Janneke Van Hofwegen (Linguistics, Stanford University, 450 Serra Mall, Building 460, Stanford, CA 94305, annetted@stanford.edu)

This paper investigates effects of linguistic and social factors on phoneme categorization of Seoul Korean stops. In an investigation of VOT in aspirated versus lenis stops of Korean, Oh (2011) reports VOT length in aspirated stops to be conditioned both linguistically and socially: bilabial stops exhibit shorter VOT than velars, following /a/ conditions shorter VOT than /i/, and female speakers exhibit shorter VOT than males. 10 native speakers of Seoul Korean (5 men, 5 women) were recorded producing bilabial and velar stops in the frame /CVn/. Recordings were manipulated to create a 10-step continuum of VOT length for each speaker, from 25ms to 115ms. 30 native speakers of Seoul Korean listened to each of these manipulated stimuli for every speaker and categorized them as containing either aspirated or lenis stops. Listeners were more likely to categorize a given VOT as aspirated when it occurred in a bilabial stop as opposed to a velar stop, when it preceded /a/ as opposed to /i/, and when it was produced by a female as opposed to a male. Results indicate that speakers exhibit knowledge of production patterns when categorizing phonemes, drawing upon both linguistic and social information.

2pSC4. Speakers of tonal and non-tonal Korean dialects use different cue weightings in the perception of the three-way laryngeal stop contrast. Hyunjung Lee, Allard Jongman, and Stephen Politzer-Ahles (Linguistics, University of Kansas, 1541 Lilac Ln., Lawrence, KS 66044, hyunjung@ku.edu)

The current study investigates if and how listeners from tonal and non-tonal varieties of Korean (Kyungsang and Seoul) use different cue weighting in their perception of the three-way distinction among Korean voiceless stops (fortis, lenis, and aspirated). Forty-two Korean listeners (21 each from Seoul and Kyungsang) were tested with stimuli in which VOT and F0 were systematically manipulated. Analyses of the perceptual identification functions show that VOT and F0 cues trade off each other for the perception of the three stops. However, the trading relationship differs between the two dialects. Logistic regression analysis reported interactions among VOT, F0 and Dialect factors for perceiving one stop over the others, indicating the two dialects use the perceptual cues differently. Specifically, while Seoul listeners rely primarily on F0 for making lenis responses and on VOT and F0 for aspirated responses, F0 plays a less important role in modulating both lenis and aspirated responses for Kyungsang speakers. A similar pattern has been observed in production [Lee and Jongman 2012, JIPA 42 (2)]. The results will be discussed in terms of the close link between perception and production across the two different dialects.

2pSC5. The waiting is the hardest part: How asynchronous acoustic cues are integrated for fricative voicing and place of articulation. Marcus E. Galle and Bob McMurray (Psychology, University of Iowa, Iowa City, IA 52240, marcus-galle@gmail.com)

A fundamental issue in speech perception is the fact that information is spread over time. This raises the question of how listeners integrate acoustic information in real-time. One possibility is that cues are utilized as soon as they arrive to partially activate lexical candidates. Alternatively, input could be a buffered, until sufficient information is available to make a decision. Between these extremes, listeners may vary depending on the usefulness of a given cue, and whether it directly cues a phonetic contrast, or serves as context (e.g., talker identity) for interpreting other cues. We examined this in word-final and initial fricative contrasts using the visual world paradigm. Participants selected a visual referent of an auditory word (ship), and we used the likelihood of fixating lexical competitors (sip/ship) at each point in time to determine when various factors affected higher level decision making. Several studies contrasted the order and utility of the cues (strong vs. weak in different positions), and direct cues vs. information like talker used for compensation. In general, listeners appear to use acoustic information immediately as it becomes available in real-time to lexical candidates.

2pSC6. The contribution of high-frequency information to fine-grained speech perception in cochlear-implant-simulated speech. Ashley Farris-Trimble and Bob McMurray (Dept. of Psychology, University of Iowa, Iowa City, IA 52242, ashley-trimble@uiowa.edu)

It has been observed that the contribution of information in the higher frequency ranges (>7000 Hz) to speech perception is negligible. This assumption has influenced cochlear implant processing strategies, which typically filter out high-frequency information. However, even if speech recognition is robust without high-frequency information, that information may nevertheless contribute to the actual processing of speech. We asked 15 normal-hearing participants to categorize words along 8-step b/p (e.g., beach to peach) and s/sh (e.g., self to shelf) continua in normal and CI-simulated speech. The CI simulations used a bandpass filter with 8 bands between 200 and 7000 Hz. The envelope for each band was extracted and used to amplify broadband noise within the band's frequency range. Results showed that participants' boundaries shifted significantly for both continua; the endpoints of each continuum were categorized consistently, but ambiguous tokens were more often categorized as sh and b when presented in CI-simulated speech. That is, eliminating high-frequency information during simulation also eliminated information necessary to the perception of sounds with high-frequency cues. These results suggest that while the loss of high-frequency information may not affect CI users' ability to recognize speech, it may compromise their fine-grained discrimination.

2pSC7. Perceptual cues in Korean fricatives. Goun Lee and Allard Jongman (Linguistics, University of Kansas, 1541 Lilac Lane, Lawrence, KS 66045-3129, cconni@ku.edu)

The current study explores the production and perception of two Korean fricatives - fortis [s'] and non-fortis (plain) [s]. Production data from 10 speakers were examined to investigate the acoustic cues that distinguish the two fricatives in different vowel contexts (high vowel /i/ vs. low vowel /a/). Measures included rise time, intensity, center of gravity (COG), F0, H1-H2, and CPP, as well as frication, aspiration, and subsequent vowel duration. COG and vowel duration consistently distinguished the two fricatives; additional cues varied across vowel contexts. For the /i/ context, intensity and F0 differed across the fricatives; for the /a/ context, rise time, H1-H2 and

CPP differed across the fricatives. Four perceptual identification experiments were conducted with sixty native Korean listeners. Experiment 1 established that listeners can distinguish the two fricatives in intact natural syllables. In Experiments 2 and 3, listeners heard only the excised consonantal or vocalic segment. For the /a/ context, fricative identification was successful based on both consonantal and vocalic segments. In the /i/ context, fricative identification exceeded chance level only for the consonantal segment. In Experiment 4, cross-spliced stimuli revealed that speakers are more sensitive to vocalic cues than consonantal cues, but only in the /a/ context.

2pSC8. Influence of lexical and acoustic context on phonetic categorization depends on listening situation. Lori L. Holt and Eva Reinisch (Psychology, Carnegie Mellon University, 5000 Forbes Avenue, Pittsburgh, PA 15213, evarei@andrew.cmu.edu)

Phonetic categorization is influenced by multiple sources of contextual information, but little is known about how different sources of information interact. We examined the relative influence of lexical versus acoustic contexts on phonetic categorization of sounds along [s]-[S] continua embedded in word-nonword pairs (e.g., a[S]amed-a[s]amed, ca[s]ino-ca[S]ino). These categorization targets were preceded by sequences of 12 nonspeech tones with mean frequencies a standard deviation above or below the spectral means of the endpoint fricatives. Listeners' [s]-[S] categorization was influenced by lexical information, exhibiting a Ganong effect with categorization shifted toward responses consistent with words, and also by acoustic context. The effect of the tone sequence was spectrally contrastive; there were more [S] responses (low spectral mean) following higher-frequency tones and more [s] responses (high spectral mean) following lower-frequency tones. In addition, the influence of acoustic relative to lexical context was modulated by listening environment. When the informational load of the lexical context was low (four word-nonword continua) acoustic context exerted a relatively greater influence than when the informational load of the lexical context was high (forty word-nonword continua). Multiple sources of context interact to influence phonetic categorization and the relative influence of different information sources is flexibly modulated by listening environment.

2pSC9. "Talker normalization" effects elicited with no change in talker. A. Davi Vitela, Andrew J. Lotto, and Brad H. Story (Speech, Language, and Hearing Sciences, University of Arizona, 1131 E. 2nd St., Tucson, AZ 85721, adv1@email.arizona.edu)

For more than fifty years, it has been known that listeners' perception of a target speech sound may shift as a result of a change in speaker of a preceding carrier phrase. The predominant theories explaining this phenomenon suggest that the listener must be extracting information specific to the speaker - either generating speaker-specific acoustic-phonemic mapping or an estimate of the dimensions of the speaker's vocal tract. To the extent that these processes derive representations that are specific to a speaker, one would not expect shifts in target perception when the apparent speaker remains unchanged. This prediction was explicitly tested by appending targets to different carrier phrases produced by the same speaker. Speaker characteristics were controlled by using an articulatory synthesizer that allows one to specify the source and dimensions of the vocal tract. Significant shifts in target perception were readily obtained, even when the speaker remained the same. These results suggest that these shifts are not tied to talker identity or anatomy but to more general aspects of the context acoustics. [Work supported by NIH-NIDCD.]

2pSC10. Efficient coding of redundancy among formant frequencies in vowels. Christian Stilp (Department of Psychological and Brain Sciences, University of Louisville, University of Louisville, 308 Life Sciences Building, Louisville, KY 40292, christian.stilp@gmail.com) and Keith Kluender (Speech, Language, and Hearing Sciences, Purdue University, West Lafayette, IN)

Stilp and colleagues (*Proc. Natl. Acad. Sci.* [2010]; *JASA* [2011]; *PLoS One* [2012]) demonstrated that auditory perception rapidly and automatically exploits redundancy among acoustic attributes in novel complex sounds. When stimuli exhibited robust covariance between acoustic dimensions (attack/decay, spectral shape), discrimination of sound pairs violating this pattern was initially poorer than that for sound pairs respecting the pattern. While results support efficient coding of statistical structure in the environment, evidence of its contribution to speech perception remains indirect. The present effort examines perceptual organization adhering to statistical regularities in speech sounds. Vowel stimuli (/ɑ/, "ah") were synthesized to reflect natural correlation between formant frequencies across talkers; as vocal tract length decreases (from men to women to children), formant center frequencies increase (here F_1-F_2 varied; others held constant). Listeners discriminated vowel pairs that either obeyed this correlation (16 pairs) or violated it (1 pair) in randomized AXB trials without feedback. Performance replicated results with nonspeech sounds. Vowels that violated natural redundancy between formant frequencies were discriminated poorer than vowels that obeyed this pattern. Results encourage an efficient coding approach to speech perception, as redundancy among stimulus attributes is exploited to facilitate perceptual organization and discrimination. [Supported by NIDCD.]

2pSC11. Timing of perception for all English diphones. Natasha L. Warner (Linguistics, University of Arizona, Box 210028, Dept. Ling, Univ. AZ, Tucson, AZ 85721-0028, nwarner@u.arizona.edu), James McQueen (Max Planck Institute for Psycholinguistics, Nijmegen, Gelderland, Netherlands), Priscilla Z. Liu, Maureen Hoffmann (Linguistics, University of Arizona, Tucson, AZ), and Anne Cutler (Max Planck Institute for Psycholinguistics, Nijmegen, Gelderland, Netherlands)

Information in speech does not unfold discretely over time; perceptual cues are gradient and overlapped. However, this varies greatly across segments and environments: listeners cannot identify the affricate in /ptʃ/ until the frication, but information about the vowel in /li/ begins early. Unlike most prior studies, which have concentrated on subsets of language sounds, this study tests perception of every English segment in every phonetic environment, sampling perceptual identification at six points in time (13,470 stimuli/listener; 20 listeners). Results show that information about consonants after another segment is most localized for affricates (almost entirely in the release), and most gradual for voiced stops. In comparison to stressed vowels, unstressed vowels have less information spreading to neighboring segments and are less well identified. Indeed, many vowels, especially lax ones, are poorly identified even by the end of the following segment. This may partly reflect listeners' familiarity with English vowels' dialectal variability. Diphthongs and diphthongal tense vowels show the most sudden improvement in identification, similar to affricates among the consonants, suggesting that information about segments defined by acoustic change is highly localized. This large dataset provides insights into speech perception and data for probabilistic modeling of spoken word recognition.

2pSC12. Importance of sentence-level and phoneme-level envelope modulations during vowels in interrupted speech. Daniel Fogerty (Communication Sciences and Disorders, University of South Carolina, 1621 Greene St, Columbia, SC 29208, fogerty@sc.edu)

Speech interrupted by noise has been used as a simplified case for listening to speech in the presence of a fluctuating masker. The present study investigated the importance of overall vowel amplitude and intrinsic vowel amplitude modulation to sentence intelligibility. Eight young normal-hearing listeners participated in the experiment. Sentences were processed according to three conditions that replaced vowel segments with noise matched to the long-term average speech spectrum. Vowels were replaced with (1) low-level noise that distorted the overall sentence envelope, (2) segment-level noise that restored the overall syllabic amplitude modulation of the sentence, and (3) segment-modulated noise that further restored faster temporal envelope modulations during the vowel. Results demonstrated incremental benefit with increasing resolution of the temporal envelope. An additional seven listeners participated in a separate experiment that replaced vowels with a vowel masker instead of noise. The vowel masker was modified to have a flattened fundamental frequency at the mean of the replaced segment. The vowel masker either had a standard temporal envelope or was modulated by the envelope of the replaced vowel. Listeners who heard vowel maskers performed more poorly than those who heard noise maskers. No benefit of segment modulation was observed with the vowel masker.

2pSC13. Refining bouba-kiki: Phonetic detail and object dimensionality in sound-shape correspondences. Annette D'Onofrio (Linguistics, Stanford University, 450 Serra Mall, Building 460, Stanford, CA 94305, annetted@stanford.edu)

Speakers cross-linguistically associate non-words that have round vowels, such as /buba/, with round shapes, and non-words without round vowels, such as /kike/, with spiky shapes (e.g. Kohler 1947). While this link has been attributed to cognitive associations between rounded vowel sounds and images of rounded lips, stimuli have conflated vowel roundness with other phonetic features that may also contribute to the correspondence. In this study, 200 listeners matched abstract objects with nonsense words that systematically varied by vowel frontness, consonant place of articulation, and consonant voicing. Listeners are significantly more likely to select a spiky shape over a round shape when given words with voiceless consonants, alveolar consonants, and front vowels; combinations of these features strengthen the effect. These findings are corroborated in the realm of real-world objects. 102 participants were more likely to name a rounded member of a real-world object class when hearing a word containing non-front vowels than when hearing a word containing front vowels. Further, two- versus three-dimensional object roundness influences the strength of this association. This study provides an empirically and phonetically refined perspective on the paradigm, demonstrating the benefit of considering both detailed phonetic correlates and refined object properties in work on sound symbolism.

2pSC14. Tone recognition in continuous Mandarin Chinese. Jiang Wu, Stephen A. Zahorian, and Hongbing Hu (Electrical & Computer Engineering, SUNY-Binghamton, 4400 Vestal Parkway East, Vestal, NY 13850, jiang.wu@binghamton.edu)

Tones are important characteristics of Mandarin Chinese for conveying lexical meaning. Thus tone recognition, either explicit or implicit, is required for automatic recognition of Mandarin. Most literature on machine recognition of tones is based on syllables spoken in isolation or even machine-synthesized voices. This is likely due to the difficulty of recognizing tones from syllables extracted from conversational speech, even for native speakers of Mandarin. In this study, human and machine recognition of tones from continuous speech is evaluated and compared for four conditions: 1, vowel portions of syllables; 2, complete syllables; 3, syllable pairs; 4, groupings of three syllables. The syllables are extracted from the RASC-863 continuous Mandarin Chinese database. The human listeners are all native speakers of Mandarin. The automatic recognition is based on either Hidden Markov Models, or neural networks, and a combination of spectral/temporal, energy, and pitch features. When very little context is used (i.e., vowel segments only) the human and machine performance is comparable. However, as the context interval is increased, the human performance is better than the best machine performance.

2pSC15. Lexical tone in Mandarin spoken word processing. Joan Sereno and Hyunjung Lee (Linguistics, University of Kansas, 1541 Lilac Ln., Lawrence, KS 66044, sereno@ku.edu)

Two priming experiments examined the separate contribution of lexical tone and segmental information in the processing of spoken words in Mandarin Chinese. Experiment 1 contrasted four types of prime-target pairs: tone-and-segment overlap (bo1-bo1), segment-only overlap (bo2-bo1), tone-only overlap (zhua1-bo1), and unrelated (han3-bo1) in an auditory lexical decision task with 48 native Mandarin listeners. Experiment 2 further investigated the minimal segmental overlap needed to trigger priming when tonal information is present. Four prime-target conditions were contrasted: tone-and-segment overlap (pa2-pa2), only onset segment overlap (ping2-pa2), only rime overlap (na2-pa2), and unrelated (kui4-pa2) in an auditory lexical decision task with 68 native Mandarin listeners. Results showed significant priming effects, as compared to the unrelated baseline, when both tonal and segmental information matched. Moreover, when only segmental information overlapped, priming was also observed while no priming for the tone-only overlap condition was found. Interestingly, partial segmental overlap, with matching tonal cues, did not produce priming. In fact, tonal overlap with matching onset segmental information resulted in significant inhibition. These data clearly indicate that tonal similarity only provides facilitation if there is complete segmental overlap. These findings will be discussed in terms of the segmental and suprasegmental information used in word recognition.

2pSC16. The effect of segmental makeup on Mandarin word and tone recognition. Yuwen Lai (Foreign Languages and Literatures, National Chiao-Tung University, Taiwan, 1001 University Road, Hsinchu, Taiwan 30010, Taiwan, yuwen.lai@gmail.com) and Sheng-Hung Wu (Foreign Languages and Literatures, National Chiao-Tung University, Taiwan, Hsinchu)

The present study adopts the gating paradigm to investigate the effect of onset sonorancy, and coda on Mandarin spoken word and tone recognition. Eight tonal quadruplets (all monosyllabic) with different initial segment (obstruent or sonorant), coda composition (no coda, alveolar nasal, or velar nasal) were used as the stimuli. The gates were formed by a 40-ms increment from the beginning of each word. Twenty native listeners from Taiwan were asked to listen to each stimulus, click the corresponding tone number using a mouse, write down the word and then give their confidence rating on the answer. The Isolation point (IP) based on correct word identification (both segment and tone) and the tone isolation point (TIP) were analyzed. The results indicated that tone recognition can be done before the offset of the stimuli. Tone 1 has an earlier IP than Tone 4, followed by Tones 3 and then Tone 2. Sonorant-initial words have a significant earlier IP than obstruent-initial ones. Words without coda have an earlier IP than alveolar nasal, followed by velar nasals. The processing time course for tone and segment and the effect of tonal features on word processing will be discussed.

2pSC17. Do preceding prosodic patterns influence word recognition in Spanish? Tuuli Morrill, Laura C. Dilley, and Jessica Navarro (Communicative Sciences and Disorders, Michigan State University, East Lansing, MI 48824, tmorrill@msu.edu)

Recent work shows lexical recognition in English is influenced by prosodic characteristics of preceding words (e.g., Dilley et al, 2010, *Journal of Memory and Language*). This study investigates whether preceding prosody affects lexical recognition in Spanish, a language with productive lexical stress contrasts (e.g., *saco* 'bag' vs. *sacó* 'he carried out'). In an experiment, each of 30 disyllabic test items was embedded in a carrier phrase; each item was consistent with one of two lexical interpretations that were distinguished by having primary stress on the initial vs. final syllable (e.g., *saco* vs. *sacó*). The prosody of words preceding the test item was resynthesized with one of two fundamental frequency and timing patterns predicted to generate distinct patterns of syllable prominence favoring perception of primary stress on either the initial or final syllable of the test item. The acoustic characteristics of the test item were identical across prosody conditions. If perception of lexical stress depends in part on prosodic characteristics of the context, then prosody preceding test items should influence lexical perception in Spanish. These findings have implications for understanding the perception of lexical stress during word recognition in languages exhibiting productive lexical stress contrasts.

2pSC18. Influence of recent linguistic exposure on the segmentation of an unfamiliar language. Jui Namjoshi (University of Illinois at Urbana-Champaign, Urbana, IL), Annie Tremblay (University of Kansas, 541 Lilac Lane Blake Hall, Room 427, Lawrence, KS 60045-3129, atrembla.illinois@gmail.com), Mirjam Broersma (Max Planck Institute for Psycholinguistics, Nijmegen, n/a, Netherlands), Sahyang Kim (Hongik University, Seoul, n/a, Republic of Korea), and Taehong Cho (Hanyang University, Seoul, n/a, Republic of Korea)

Studies have shown that listeners segmenting unfamiliar languages transfer native-language (L1) segmentation cues. These studies, however, conflated L1 and recent linguistic exposure. The present study investigates the relative influences of L1 and recent linguistic exposure on the use of prosodic cues for segmenting an artificial language (AL). Participants were L1-French listeners, high-proficiency L2-French L1-English listeners, and L1-English listeners without functional knowledge of French. The prosodic cue assessed was F0 rise, which is word-final in French, but in English tends to be word-initial. 30 participants heard a 20-minute AL speech stream with word-final boundaries marked by F0 rise, and decided in a subsequent listening task which of two words (without word-final F0 rise) had been heard in the speech stream. The analyses revealed a marginally significant effect of L1 (all listeners) and, importantly, a significant effect of recent linguistic exposure (L1-French and L2-French listeners): accuracy increased with decreasing time in the US since the listeners' last significant (3+ months) stay in a French-speaking environment. Interestingly, no effect of L2 proficiency was found (L2-French listeners).

2pSC19. Time-course of lexical knowledge use in processing pronunciation variants. Stanislav M. Sajin and Cynthia M. Connine (Psychology, Binghamton University, 4400 Vestal Parkway East, Binghamton, NY 13902, connine@binghamton.edu)

Three experiments examined how listeners process words that are produced with a context conditioned sound change, fricative assimilation. In fricative assimilation, a word final /s/ can change to a /ʃ / when a following word begins with an approximate segment (e.g., dress yacht –> dresh yacht). Fricative assimilation is blocked in other segmental environments such as plosives (e.g., dress boat will not alter the word final /s/). Previous research revealed that phonological variation is acceptable as long as it is licensed by the context in which it is embedded (Gaskell &Marslen-Wilson, 1996). However, there is indication that such licensing effects are largely consigned to particular types of assimilation (e.g., place assimilation) and that other word-final variations (e.g., flap variants), which happen more often in natural speech, have a separate phonological representation that is dependent on the frequency with which the variation occurs (Ranbom, Connine, &Yudman, 2009). The present research addresses how assimilated speech is processed in recognizing spoken words, and, specifically, how different sources of knowledge (lexical and phonological context) are utilized at different times during processing. The results point out that word knowledge is not essential for utilizing the phonological context during recognition, but it permits earlier use of the phonological context in recognizing the pronunciation variant.

2pSC20. The contribution of individual words to the meaning of sentences. Amber M. Veliz and Theodore S. Bell (Psychology, California State University, Los Angeles, P.O. Box 2476, Montebello, CA 90640, amber.veliz2476@yahoo.com)

The studies purpose was assessing the contribution of individual words of a sentence to its overall intelligibility. The dependent variable was number of target words in each sentence correctly identified at each of three possible word positions, initial, middle, and last. The question of interest was how degradation of audibility of the initial word would affect other words by decreasing the semantic context for those later words resulting from the audibility of the first word. Other variables in the design included presentation levels that resulted in overall performance from 50 to 100 percent correct. There was a three way interaction of word position, masker, and presentation level on performance scores. Noise levels were fixed at 60 dB SPL, and speech levels were varied to -3, -6 and -9 signal-to-noise ratio (SNR). Between subject factors in the ANOVA design were SNR (-3, -6, and -9 db), and masker (flat or enhanced).

2pSC21. Cochlea-scaled entropy predicts intelligibility of Mandarin Chinese sentences. Yue Jiang (Speech, Language, and Hearing Sciences, Purdue University, Heavilon Hall, 500 Oval Drive, West Lafayette, IN 47907-2038, jiang23y@gmail.com), Christian E. Stilp (Department of Psychological and Brain Sciences, University of Louisville, Louisville, KY), and Keith R. Kluender (Speech, Language, and Hearing Sciences, Purdue University, West Lafayette, IN)

Cochlea-scaled spectral entropy (CSE; Stilp & Kluender, *PNAS, 107(27)*:12387-12392 [2010]) is a measure of information-bearing change in complex acoustic signals such as speech. CSE robustly predicts English sentence intelligibility even amidst temporal distortion and widely different speaking rates (Stilp, Kiefte, Alexander, & Kluender, *JASA, 128(4)*:2112-2126). However, CSE does not explicitly capture changes in fundamental frequency ($f0$) in any way distinct from that for other aspects of spectral shape (e.g., formant patterns, slope). This property of CSE could limit its ability to predict intelligibility of tone languages that use $f0$ for phonetic and morphological distinctions. The present study assesses the predictive power of CSE for Mandarin Chinese sentence intelligibility. Twenty-five native-Mandarin listeners transcribed Mandarin sentences in which consonant-length (80-ms) and vowel-length (112-ms) segments with either high or low CSE were replaced with speech-shaped noise. CSE reliably predicted listener performance; as greater amounts of CSE were replaced by noise, performance worsened. Results encourage information-theoretic approaches to speech perception, as change and not physical acoustic measures best predict sentence intelligibility across tonal and nontonal languages. [Work supported by NIDCD.]

2pSC22. Effects of noise on adults' word learning. Min Kyung Han, Holly L. Storkel, and Casey Cox (Speech-Language-Hearing, University of Kansas, 3001 Dole Center 1000 Sunnyside Avenue, Lawrence, KS 66045-7555, minhan@ku.edu)

Neighborhood density, the number of similar sounding words, and phonotactic probability, the likelihood of occurrence of a sound sequence, influence word learning by adults in quiet listening conditions. Specifically, Storkel, Armbruster, and Hogan (2006) found that adults learned high density words and low probability words more accurately. This study examined how neighborhood density and phonotactic probability influence adults' word learning in noisy conditions. Fifty-two college students learned nonwords varying in neighborhood density and phonotactic probability at either +8 dB SNR or 0 dB SNR, and learning was measured in a picture naming task. Results showed a significant interaction of density, probability, and noise level. At +8 dB SNR, no effects were significant. At 0 dB SNR, density interacted with probability, showing better learning when density and probability converged. In other words, for low probability words, learning was better when density was also low (i.e., low-low optimal), whereas for high probability words, learning was better when density was also high (i.e., high-high optimal). These results indicate that noise dampens the effects of probability and density in a moderately noisy condition (i.e., +8 dB SNR) and requires a convergence of probability and density in the noisiest condition (i.e., 0 dB SNR).

2pSC23. The influence of a foreign accent on recall of spoken word lists. Marissa Ganeku and Tessa Bent (Department of Speech and Hearing Sciences, Indiana University, 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

Talker and speaking rate variability influence spoken word recall (Nygaard et al., 1995; Goldinger et al., 1991; Martin et al., 1989) suggesting that indexical information affects memory processes. The current study investigated how another type of speech variability - the presence of a foreign accent - influenced spoken word recall. Recall of 10 spoken word lists with 10 words each was assessed at two presentation rates. Listeners received words from a native-accented talker, a non-native talker with a strong foreign accent, or a non-native talker with a mild foreign accent. Neighborhood density, neighborhood frequency, and word frequency were balanced across each list; all words were 100% intelligible (Bent, 2010). Recall was analyzed by: (1) accuracy in the correct serial position and (2) accuracy regardless of serial position. The first analysis demonstrated that listeners recalled more words with the slower presentation rate and in the early and late list positions, replicating earlier findings, but there was no talker effect. The second analysis demonstrated that listeners recalled fewer words from the strongly foreign-accented talker than the native talker or the mildly foreign-accented talker. Thus, foreign-accented speech is another source of variability that can influence processing and encoding in memory.

2pSC24. Sentence recognition as a function of the number of talkers in competing multi-talker babble. Kristin Van Engen and Bharath Chandrasekaran (Communication Sciences and Disorders, University of Texas, Austin, TX 78712, kj.vanengen@gmail.com)

Multiple-talker babble is used in studies of speech perception and processing both as a tool for loading perceptual and cognitive tasks and for direct assessment of speech perception in noise. Single interfering talkers and babbles containing few talkers differ dramatically, however, from babbles with high numbers of talkers. While low N-talker babbles provide greater opportunities for "glimpsing" speech targets during spectral and temporal dips in the maskers (i.e., less energetic masking), the linguistic information available to listeners in the maskers themselves also imposes higher-level interference (i.e., informational masking) that may detract from the identification of target speech. In order to assess the relative overall masking effects of N-talker babble for sentence recognition, the current study utilizes a range of maskers varying in N (1-8), as well as speech-shaped noise. Data collected to date show that performance declines as talkers are added to the masker (that is, as energetic masking increases), but that performance in 6-8 talker babble is significantly better than in speech-shaped noise. Individual variability on speech perception in these various maskers is being assessed by comparing performance on the speech-in-noise task to performance on a range of cognitive and psychoacoustic tasks.

Session 2pSPa

Signal Processing in Acoustics: Cognitive Approaches to Acoustic Signal Processing

Grace A. Clark, Cochair
Engineering, Lawrence Livermore National Laboratory, Livermore, CA 94550

R. Lee Culver, Cochair
ARL, Penn State University, State College, PA 16804

Chair's Introduction—1:10

Invited Papers

1:15

2pSPa1. Adaptive modulation and power control for underwater acoustic communications. Andreja Radosevic (Department of Electrical and Computer Engineering, University of California, La Jolla, CA), Rameez Ahmed Rasheed Ahmed (Department of Electrical and Computer Engineering, Northeastern University, 360 Huntington Avenue, Boston, MA 02115, rarameez@ece.neu.edu), Tolga M. Duman (School of Electrical, Computer and Energy Engineering, Arizona State University, Tempe, AZ), John G. Proakis (Department of Electrical and Computer Engineering, University of California, La Jolla, Arizona), and Milica Stojanovic (Department of Electrical and Computer Engineering, Northeastern University, Boston, MA)

In this work we explore the feasibility of a cognitive acoustic communication system that exploits a dynamic closed loop between the transmitter and receiver with the goal of maximizing the information throughput. Specifically, we design a power control mechanism and couple it with an adaptive modulation method based on orthogonal frequency division multiplexing (OFDM). We propose two methods: the first method is adaptive overall power adjustment in which the transmitter modifies the power gain to provide a target SNR at the receiver by exploiting a feedback link in a time-varying channel. The second method adaptively adjusts the modulation level on each individual sub-carrier to achieve a target bit error rate (BER) at the receiver. Crucial to both of these methods is the ability to predict the acoustic channel and the signal-to-noise ratio (SNR) one travel time ahead, which enables adaptive adjustment of the transmit power and modulation level. The performance of the proposed algorithms is demonstrated using an in-air test bed and further verified with real-time at-sea experiments conducted off the coast of Kauai, HI, in June 2011. Experimental results obtained using real-time at-sea experiments show significant savings in power, as well as improvement in the overall information throughput (bit rate) as compared to conventional, non-adaptive modulation with the same power consumption and target BER.

1:40

2pSPa2. Optimizing receiver and source positioning in the ocean: Lessons from nature. Zoi-Heleni Michalopoulou (Department of Mathematical Sciences, New Jersey Institute of Technology, Newark, NJ 07102, michalop@njit.edu)

Extracting information from the ocean using acoustic signals as well as detecting and localizing sources is critical for both defense and environmental applications. Accurate parameter estimation strongly relies on optimal positioning of sources and receivers, whose locations are often the function of the ocean waveguide. Observing marine-mammal vocalizations, one notices that preference is given to transmission and reception at particular depths. A simple sound propagation analysis indicates that use of the specific locations may optimize sound reception. We can make similar choices with man-made systems, calibrating the locations of deployed sound transmission/reception equipment, with optimal detection and estimation in mind. [Work supported by ONR.]

2:05

2pSPa3. Illumination waveform design for non-Gaussian multi-hypothesis target classification in cognitive radar. Ke N. Wang (Electrical and Computer Engineering, Naval Postgraduate School, Livermore, CA), Grace A. Clark (Engineering, Lawrence Livermore National Laboratory, 7000 Ease Ave., L-130, Livermore, CA 94550, clarkga1@comcast.net), and Ric Romero (Electrical and Computer Engineering, Naval Postgraduate School, Monterey, CA)

A cognitive radar (CR) system is one that observes and learns from the environment; then uses a dynamic closed-loop feedback mechanism to adapt the illumination waveform so as to provide system performance improvements over traditional radar systems. Romero et al. recently developed a CR system that performs multiple hypothesis target classification and exploits the spectral sparsity of correlated narrowband target responses to achieve significant performance improvements over traditional radars that use wideband illumination pulses. This CR system was designed for Gaussian target responses. This research generalizes the CR system to deal effectively with arbitrary non-Gaussian distributed target responses via two key contributions: (1) an important statistical expected value operation that is usually evaluated in closed form is evaluated numerically using an ensemble averaging operation, and (2) a powerful new statistical sampling algorithm and a kernel density estimator are applied to draw complex target samples from target distributions specified by both a desired power spectral density and an arbitrary desired probability density function. Simulations using non-Gaussian targets demonstrate very effective algorithm performance.

Contributed Papers

2:30

2pSPa4. Speech enhancement via only mostly blind source separation. Richard Goldhor, Karen Chenausky (Speech Technology & Applied Research, 54 Middlesex Turnpike, Entrance D, Bedford, MA 01730, rgoldhor@sprynet.com), Suzanne Boyce (Communication Sciences and Disorders, University of Cincinnati, Cincinnati, OH), Keith Gilbert (Speech Technology & Applied Research, Bedford, MA), Sarah M. Hamilton (Communication Sciences and Disorders, University of Cincinnati, Cincinnati, OH), and Joseph Cin (Speech Technology & Applied Research, Bedford, MA)

In environments in which multiple simultaneously-active acoustic sources contribute to sensor responses, Blind Source Separation (BSS) signal processing techniques may be employed to separate (that is, estimate or reconstruct) the signal characteristics of "hidden" sources. Only Mostly Blind Source Separation (OMBSS) involves the estimation of similar sources in important contexts in which non-acoustic information is also available about one or more of the contributing sources. Recently-reported objective source separation performance measures confirm that non-acoustic information can be used effectively to support high-quality separation in situations in which traditional BSS methods perform poorly (e.g., when more sources are active than there are microphones available). Here we present the results of additional perceptual and objective tests showing that OMBSS processing enhances the intelligibility of speech recorded in the presence of multiple simultaneous speech-babble and non-speech maskers.

2:45

2pSPa5. Use of pattern classification algorithms to interpret acoustic echolocation data from a walking-speed robotic sensor platform. Eric A. Dieckman and Mark Hinders (Dept of Applied Science, College of William and Mary, P.O. Box 8795, Williamsburg, VA 23187, eric.dieckman@gmail.com)

An unresolved issue for autonomous walking-speed robots in unstructured outdoor environments is maintaining situational awareness. One strategy is combining information from different sensors so the robot can function in a variety of conditions and environments. The very low-cost Microsoft Kinect accessory incorporates active infrared and RGB video sensors to provide real-time depth information, as well as a 4-channel microphone array. We validate the Kinect sensors and investigate the combination of infrared (passive and active) and non-linear acoustic echolocation sensors on our mobile robotic sensor platform. By using an acoustic parametric array to generate the audible echolocation signal, a tightly-controlled beam of low-frequency sound can interrogate targets at long distances, while infrared imaging works well in the nearfield and in difficult weather conditions. Sophisticated signal processing techniques are required to combine and interpret the collected data; we present an example using pattern classification on the acoustic echolocation data to differentiate between vehicle types.

3:00

2pSPa6. Real-time active noise control of magnetic resonance imaging acoustic noise. Chiao-Ying Lin and Jyh-Horng Chen (Electrical Engineering, National Taiwan University, Taipei 104, Taiwan, d95921031@gmail.com)

Magnetic resonance imaging (MRI) is extensively used in clinical and medical researches because MRI is non-invasive and non-radiation. However, using the MRI cause the loud acoustic noise generated by gradient switching. The MRI noise is annoying patients. Therefore, active noise cancellation (ANC) is used to solve this problem in this article. In this article, the main work is to implement real-time system using DSP. The noise generated by each planar imaging (EPI), which is generally used in fMRI researches, is recorded from 3 Tesla MRI for testing ANC system. The EPI signal is regular and predictable, means the signal like a pattern, we using LMS algorithm to training the signal and learning predict the noise spectrum. After the calculated the noise, we output the reverse single to we home-built headset.

TUESDAY AFTERNOON, 23 OCTOBER 2012 — LESTER YOUNG A, 3:25 P.M. TO 4:30 P.M.

Session 2pSPb

Signal Processing in Acoustics: Acoustics for Forensics and Identification

Al Yonovitz, Chair
Communicative Sciences and Disorders, University of Montana, Missoula, MT 59812

Chair's Introduction—3:25

Contributed Papers

3:30

2pSPb1. Digital signal processing in forensic acoustics cases. Al Yonovitz (Communicative Sciences and Disorders, University of Montana, Curry Health Center, Lower Level, Missoula, MT 59812, al.yonovitz@umontana.edu), Herbert Joe (Graduate School of Business, University of Phoenix, Park City, UT), and Joshua Yonovitz (Law, Business and Arts, Charles Darwin University, Tallahassee, Florida)

Forensic acoustics typically considers speaker identification, authenticity analysis, audio enhancement and transcript verification by analyzing and processing audio or speech signals. Other applications in forensic acoustics include different analyses of non-speech audio signals. This presentation will discuss three contentious litigation cases in which various digital processing techniques formed the foundation of scientifically-based conclusions. The first is a products liability case involving the death of a firefighter and his Personal Alert Safety System. When movement of a firefighter ceases, a high-pitched audible alert is emitted. When numbers of these devices were recorded, the task was to determine if the deceased firefighter's device was included within the multiple devices examined. In the second case the order of the discharges of various caliber weapons based on a 9-1-1 call was determined via digital signal processing. The third study utilized processing techniques to differentiate and identify the type of sounds produced in a number of murder trials. These cases represent the varied challenges in forensic audio and acoustics investigations, and show the difficulty of applying broad theory to practice, the necessity for innovation in the field and the basis for establishing scientific reliability in order to secure admissibility into the courtroom.

3:45

2pSPb2. Randomized sample-level interpolation for audio content manipulation. Samarth Hosakere Shivaswamy, Stephen Roessner, Xiang Zhou, Gang Ren (Dept. of Electrical and Computer Engineering, Univ. of Rochester, Rochester, NY 14627, g.ren@rochester.edu), Mark Bocko, and Dave Headlam (Dept. of Electrical and Computer Engineering; Dept. of Music Theory, Eastman Sch. of Music, Univ. of Rochester, Rochester, NY)

In this paper we propose a novel audio signal manipulation algorithm based on sample-level interpolations that can generate multiple unique versions of an audio file without creating any perceptual difference. The proposed algorithm enables important applications such as digital rights management and file distribution tracking. The simplest sample-level interpolation method is based on time domain interpolation of fixed-length audio frames. The processing algorithm first segments the audio signal into fixed-length frames. For each frame, we perform a slight time compression or extension using an audio sample interpolation algorithm and then we recombine the manipulated audio samples to form a manipulated version of the original audio files. To enable better security features a randomization program is applied to control the frame-length and manipulation-length using pseudo-random sequences. The result of this algorithm is effectively a form of weak frequency modulation. If the fame size is larger than the compression/extension sample number, these compression/extension manipulations will not produce any audible difference. Various subjective evaluation experiments are conducted to decide the extent of the admissible processing parameters that will not cause noticeable difference in both fixed-length and randomized-length sample manipulation. The authors also provide several implementation examples and a brief summary.

4:00

2pSPb3. Phase discontinuity detection as a means of detecting tampering with speech recordings. Jerome Helffrich and John D. Harrison (Applied Physics, Southwest Research Institute, San Antonio, TX 78228, jhelffrich@swri.org)

In the forensic analysis of speech recordings, it is frequently necessary to validate the authenticity of the recording—for example, to determine if it has been altered for the purpose of changing the meaning of what was spoken. One method for doing this is to exploit an incidental steady tonal content in the recording, and look for sudden, discontinuous changes in the phase of that tone. We report on the development of an algorithm based on this idea, to be used for detecting edits in digital or analog speech recordings. Our development process included generating a corpus from the TIMIT speech database, designing an algorithm to exploit possible phase discontinuities in the recorded signals at edit sites, and testing the algorithm against the corpus with both edited and unedited copies of the sentence material. A statistical analysis of the performance of the algorithm on sentences contaminated by noise and compressed by various digital compression schemes will be given.

4:15

2pSPb4. Speech enhancement by maintaining phase continuity between consecutive analysis frames. Erdal Mehmetcik (ASELSAN, PK.1. Yenimahalle, Ankara 06200, Turkey, emehmetcik@aselsan.com.tr) and Tolga Ciloglu (Electrical and Electronics Engineering Department, Middle East Technical University, Ankara, Turkey)

The aim of speech enhancement algorithms is to increase the quality and intelligibility of noise degraded speech signals. Classical algorithms make modifications in the magnitude spectrum of the noise degraded signal and leave the phase spectrum unchanged. Leaving the phase spectrum unchanged relies on the results of the early listening tests, in which it is concluded that a better phase estimation does not have a significant effect on speech quality. However, a poor phase estimate causes a noticeable distortion in the reconstructed signal. In this work a new phase estimation method (in the voiced segments of speech) is proposed. This method is then incorporated into classical (magnitude based) enhancement algorithms and a new enhancement method is put forward. The performance of the proposed enhancement method is tested with the ITU-T standard PESQ measure and the results are presented.

TUESDAY AFTERNOON, 23 OCTOBER 2012 — MARY LOU WILLIAMS A/B, 1:30 P.M. TO 4:30 P.M.

Session 2pUW

Underwater Acoustics: Parabolic Equation Methods and Comparisons

Timothy F. Duda, Chair
Woods Hole Oceanographic Inst., Woods Hole, MA 02543

Contributed Papers

1:30

2pUW1. Exact parabolic equation for the sound field in inhomogeneous ocean. Nikolai Maltsev (Frontier Semiconductor, 2127 Ringwood Avenue, San Jose, CA 95131, admin@asymptotus.com)

A system of equations $\partial P/\partial x = i\omega\rho u, \partial u/\partial x = (i\omega\rho)^{-1}(-\triangle_{yz}+\nabla_{yz}\rho,\nabla_{yz})-(\omega/c)^2)P$ where $P(\mathbf{r}).u(\mathbf{r})$ are sound pressure and horizontal particle velocity, $c(\mathbf{r}),\rho(\mathbf{r})$-sound speed and density, ω-angular frequency and $\triangle_{yz},\nabla_{yz}$ are laplacian and gradient in the plane (y,z), are an exact equations for the $P(\mathbf{r})$ and $u(\mathbf{r})$, derived from Euler equations of small motion of compressible fluid. They have first order with respect to x and ready for different marching type algorithms like split-step and Crank-Nicholson, without any approximations of operators, unlike traditional PE scheme. Different examples, illustrating applications of this equations are presented.

1:45

2pUW2. Use of Galerkin's method using variable depth grids in the parabolic equation model. William Sanders (Seafloor Sciences, Naval Research Laboratory, Naval Research Laboratory, Stennis Space Center, MS 39529, wsanders@nrlssc.navy.mil)

Choice of a depth grid affects the accuracy of a parabolic equation (PE) propagation model, as well as the speed of execution. Choice of a fine grid step may result in lower discretization errors, but will lengthen computation times. Moreover, application of a sampling requirement (e.g. N samples per wavelength) to the water column results in oversampling in the bottom. Since an artificial absorbing later in the bottom is often employed, a large part of the computational domain is oversampled. Fine depth sampling is only needed in the region about the water sediment interface. Hence, Galerkin's method using a variable depth grid is implemented. This is demonstrated to achieve the same error as a uniform grid with fine spacing over the entire depth domain, while taking a fraction of the run time. This is particularly important in models where nested PE models are required, as with noise models (so called N by 2D runs), full 3D, or broadband models. The variable grid Galerkin's method is also used in an elastic PE model, in which sampling requirements for low-shear speed sediments require much finer sampling than that in the water column.

2:00

2pUW3. Applications of a higher order operator splitting scheme on parabolic-equation methods for modeling underwater sound propagation. Ying-Tsong Lin, Timothy F. Duda (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543, ytlin@whoi.edu), Jon M. Collis (Applied Mathematics & Statistics, Colorado School of Mines, Golden, CO), and Arthur E. Newhall (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA)

A higher order operator splitting scheme is presented to increase the accuracy of parabolic-equation (PE) methods. This splitting scheme is essentially applied to the square-root Helmholtz operator resulting from the PE approximation, and it can benefit both the split-step Fourier and split-step Padé methods for modeling sound propagation in three-dimensional environments. Note that the square-root Helmholtz operator in the split-step Fourier PE method is split into a free propagator and a phase anomaly, but in the split-step Padé PE method it is split into two one-dimensional transverse derivatives. Our higher order scheme can handle these two different types of operator splittings, and accuracy improvements come from the fact that our computation includes (to second order) cross terms of the split operators. Numerical underwater acoustic examples are provided to demonstrate the performance of this scheme and comparisons against other schemes. [Work supported by the Office of Naval Research.]

2:15

2pUW4. Propagation modeling of under-ice transmissions using the parabolic equation. Kevin D. Heaney, Richard L. Campbell (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com), and Lee F. Frietag (Woods Hole Oceanographic Institution, Woods Hole, MA)

A recent reformulation of the RAM Parabolic Equation model (OASIS' CRAM model) permits efficient Nx2D propagation modelling, including ACOMMS performance estimation. This model is currently set up as a 3-layer propagation model. The 3-layers commonly used are water-column, sediment and basement - where the water-column and sediment have a depth dependent compressional speed, density and attenuation. The basement is an acoustic half-space with a sponge. To model under-ice propagation, the 3-layers can set up as sea-ice, water-column and seafloor. The only loss of generality is the single half-space seafloor, although bottom interaction is not as important a feature in the iso-thermal, upward refracting sound speed profiles in northern latitudes. The ice is modeled as an iso-speed (1700 m/s, rho = .988 kg/m3, alpha= 0.3dB/lambda) fluid on-top of the seawater. The ice-water boundary is input into the code the same way a complex bathymetric profile is used. Although this approach doesn't include shear propagation - an important source of frequency dependent attenuation, it does accurately model the scattering within the water column induced by interactions with the complex under-ice morphology. Comparison of measurements with models will be made for 900 Hz broadband transmissions to ranges of 50 km.

2:30

2pUW5. Seismo-acoustic propagation near thin and low-shear speed ocean bottom sediments. Jon M. Collis (Applied Mathematics and Statistics, Colorado School of Mines, 1500 Illinois Street, Golden, CO 80401, jcollis@mines.edu), Adam M. Metzler (Applied Research Laboratory, University of Texas, Austin, TX), and William L. Siegmann (Mathematical Sciences, Rensselaer Polytechnic Institute, Troy, NY)

Accurate and efficient parabolic equation solutions exist for complex propagation environments with elastic sediments. Certain ocean acoustic environments (such as harbors or estuaries) can feature a seafloor interface consisting of partially consolidated sediments, which can be described as a transitional solid. These complex sediments are generally thin, with low-shear wave speeds, and can cause numerical instabilities to arise in parabolic equation solutions. These instabilities make it difficult to obtain accurate solutions. In the low-shear limit, this problem can be treated as a multiple-scale problem. In this talk, such an ocean environment is modeled as a water layer overlying a thin transitional solid sediment layer over an elastic basement. An elastic parabolic equation approach is developed using asymptotic solutions for the displacements in the transitional solid, which are then incorporated into existing seismo-acoustic parabolic equation solutions by explicitly enforcing fluid-solid and solid-solid interface conditions across the transitional-solid interfaces. Solutions are benchmarked against existing elastic parabolic equation and normal mode solutions for accuracy.

2:45

2pUW6. Three-dimensional numerical modeling of sound propagation and scattering in the deep ocean with elastic (shear) bottoms. Ilya A. Udovydchenkov (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 266 Woods Hole Rd., Woods Hole, MA 02543, ilya@whoi.edu), Ralph A. Stephen (Geology and Geophysics, Woods Hole Oceanographic Institution, Woods Hole, MA), Timothy F. Duda, Ying-Tsong Lin (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA), and Daniel Peter (Department of Geosciences, Princeton University, Princeton, NJ)

A major challenge in bottom-interacting ocean acoustics is to include an accurate description of environmental variability in a computationally feasible model. Wave field predictions are often difficult in environments with strong range dependence, with rapid bathymetric variations, with multiple scattering regions, with interface waves at fluid/solid boundaries, and/or with shear waves in the bottom. In this presentation, we are adapting an existing three-dimensional spectral-finite-element code (SPECFEM3D, distributed and supported by the NSF funded program, Computational Infrastructure for Geodynamics), originally developed for global seismology, to bottom interaction problems in underwater acoustics. Numerical results from SPECFEM3D are compared with the range-dependent acoustic/elastic wave propagation model based on the parabolic equation (PE) method, for a 10 Hz acoustic pulse propagating in the deep ocean. The importance of out-of-plane scattering and bottom shear properties on resulting wave fields are investigated. [Work supported by ONR.]

3:00–3:15 Break

3:15

2pUW7. Effects of seismic source and environment parameters on elastic bottom parabolic equation solutions. Scott D. Frank (Mathematics, Marist College, 3399 North Ave., Marist College Mathematics, Poughkeepsie, NY 12601, scott.frank@marist.edu), Jon M. Collis (Applied Mathematics and Statistics, Colorado School of Mines, Golden, CO), and Robert I. Odom (Applied Physics Laboratory, University of Washington, Seattle, WA)

Recently, two-types of elastic self-starters have been incorporated into parabolic equation solutions for range-dependent elastic bottom underwater acoustic problems. These source fields generate parabolic equation solutions that can be used to study development of oceanic T-phases via the process of downslope conversion. More general range-dependence has also been shown to scatter elastic wave energy into water column acoustic modes which then propagate as T-phases. In certain circumstances, sources in the elastic bottom can also cause interface waves at the ocean bottom that contribute to the ocean acoustic field. Both types of waves can propagate long distances and could be source mechanisms for unexplained acoustic signals recorded near the sea floor and below the ray-theoretic turning point. Parabolic equation solutions will be used to study effects of parameters such as frequency, source location, and source type on T-phase and interface wave generation and propagation.[Work supported by ONR.]

3:30

2pUW8. Modeling low-frequency seismo-acoustic propagation in the Arctic using a parabolic equation solution. Adam M. Metzler (Environmental Sciences Laboratory, Applied Research Laboratories: University of Texas, 10000 Burnet Rd, Austin, TX 78713, ametzler@arlut.utexas.edu), Jon M. Collis (Applied Mathematics and Statistics, Colorado School of Mines, Golden, CO), and William L. Siegmann (Mathematical Sciences, Rensselaer Polytechnic Institute, Troy, NY)

Propagation in Arctic environments is complicated by three-dimensional variations in the waveguide. The sound speed minimum occurs at or near the ice-covered surface, and the upward refracting profile causes long-range propagation to interact repeatedly with the ice cover. Propagation in the Arctic waveguide needs to include an elastic ice cover of variable thickness which may terminate, overlying an ocean layer and an elastic sediment bottom. Parabolic equation solutions are accurate and efficient for elastic layers, although currently, solutions do not exist for Arctic environments. In this paper, elastic parabolic equation solutions will be obtained for range-dependent Arctic waveguides with elastic ice cover and an elastic bottom. Particular interest will be directed to environments where the ice cover terminates. Results will be benchmarked against normal mode and wave number integration solutions.

3:45

2pUW9. Elastic coupled modes for range dependent propagation. Minkyu Park (Geophysics, Korean Polar Research Institute, Incheon, Republic of Korea), Robert I. Odom (Applied Physics Lab, University of Washington, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, odom@apl.washington.edu), Scott D. Frank (Mathematics, Marist College, Poughkeepsie, NY), and Jon M. Collis (Applied Mathematics and Statistics, Colorado School of Mines, Golden, CO)

The seismo-acoustic field in a range dependent fluid-elastic environment can be computed from elastic coupled modes. Range dependence breaks the strict mathematical orthogonality of the modes causing energy to be exchanged among the elements of the modal spectrum. The original theory is from Maupin (1988). Range-dependent propagation effects are illustrated for a 2-D model including a seamount. The mode coupling is directly proportional to the bathymetric slope of the range dependence. While the range-dependent environmental model is composed of vertical slices, no iteration is required to solve for both the transmitted and the reflected seismo-acoustic field, which is found by solving a matrix Riccati equation for the modal reflection and transmission matrices. A particular feature of the elastic coupled modes is the automatic inclusion of the interface Scholte wave modes, which are absent from any all-fluid environmental propagation model. The excitation of the Scholte modes may be important for observed deep shadow zone acoustic arrivals. The coupled mode results will be compared to the results from an elastic parabolic equation (PE). [Work supported by ONR.]

4:00

2pUW10. Effects of tropical cyclones on underwater sound propagation. Arthur E. Newhall, Ying-Tsong Lin (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543, anewhall@whoi.edu), Sen Jan (Institute of Oceanography, National Taiwan University, Taipei, Taiwan), and James F. Lynch (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA)

Environmental data collected near the continental shelf and the shelfbreak in the Southern East China Sea around Taiwan are utilized to study sound propagation effects of tropical cyclones (typhoons). These data were from the Conductivity-Temperature-Depth (CTD) profiling casts conducted during the Quantifying, Predicting, Exploiting (QPE) Uncertainty experiments in both 2008 and 2009. These CTD surveys provided observations for the normal physical oceanographic conditions in the area and, most importantly, the extreme conditions induced by a tropical cyclone Typhoon Morakot in the summer of 2009. Strong upwelling currents near the shelfbreak in the experiment area were observed after the typhoon past by, and the water-column stratification was changed. A sound speed variation of 10 m/s over 50 km in distance was measured in the upwelling area, and this can produce significant effects on sound propagation. Numerical models using parabolic-equation and ray tracing methods will be presented to demonstrate the underlying physics of the sound field variability. [Work supported by the Office of Naval Research.]

4:15

2pUW11. The effects of mode structure on coherence in shallow water propagation. Jennifer Wylie, Harry DeFerrari, and Felipe Lourenco (AMP, University of Miami, FL 33432, jennie.wylie@gmail.com)

In an ideal shallow water propagation channel the sound field is accurately described by normal modes and the mode structure is predictable with clean separated modes. However the real ocean environment is rarely ideal, and variations in bottom bathymetry and water column sound speed are usually present. When fluctuations are small and on the order of a fraction of an acoustic wavelength the sound field propagation is deterministic and any mode pattern deviations are small and slow and the phase response is linear. Here the propagation is considered phase coherent and spatial and/or temporal averaging will produce gain. When the fluctuations increase to the order of $1/2$ to 1 acoustic wavelength the mode patterns becomes distorted such that slow linear fluctuations in phase can no longer describe the propagation. Here coherence is reduced and in some cases completely lost. A unifying theory will be presented linking mode pattern deviations and coherence. PE models will be used to predict mode structure and individual mode correlation will be computed with the ideal case. These mode correlations will be used to estimate temporal and spatial coherence and the results will be compared with data from three shallow water experiments.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

P. Battenberg, Chair ASC S1
Quest Technologies, Inc., 1060 Corporate Center Drive, Oconomowoc, WI 53066-4828

R.J. Peppin, Vice Chair ASC S1
Scantek, Inc., 6430 Dobbin Road, #C, Columbia, MD 21045

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound, etc. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 23 October 2012.

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance and comfort.

TUESDAY AFTERNOON, 23 OCTOBER 2012 JAY MCSHAN A, 2:30 P.M. TO 3:45 P.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

C.J. Struck, Chair ASC S3
CJS Labs, 57 States Street, San Francisco CA 94114-1401

G.J. Frye, Vice Chair ASC S3
Frye Electronics, Inc., P.O. Box 23391, Tigard OR 97281

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 23 October 2012.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance and comfort.

TUESDAY AFTERNOON, 23 OCTOBER 2012 JAY MCSHAN A, 4:00 P.M. TO 5:00 P.M.

Meeting of Accredited Standards Committee (ASC) S3/SC 1, Animal Bioacoustics

D.K. Delaney, Chair ASC S3/SC 1
USA CERL, 2902 Newmark Drive, Champaign, IL 61822

M.C. Hastings, Vice Chair ASC S3/SC 1
Georgia Institute of Technology, G.W. Woodruff School of Mechanical Engineering,
126 Love Building, 771 Ferst Drive, Atlanta, GA 30332 0405

Accredited Standards Committee S3/SC 1 on Animal Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43/SC 1 Noise and ISO/TC 43/SC 3, Underwater acoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 23 October 2012.

Scope of S3/SC 1: Standards, specifications, methods of measurement and test, instrumentation and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria or aquariums; or free-ranging wild animals.

TUESDAY EVENING, 23 OCTOBER 2012 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Tuesday are as follows:

Acoustical Oceanography	Mary Lou Williams
Architectural Acoustics	Trianon C/D
Engineering Acoustics	Lester Young A
Musical Acoustics	Andy Kirk
Physical Acoustics	Basie A1
Psychological and Physiological Acoustics	Salon 7 Roosevelt
Structural Acoustics and Vibration	Trianon A/B