Session 2aAA

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

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Troy, NY 12180

David H. Griesinger, Cochair
Research, David Griesinger Acoustics, Cambridge, MA 02138

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

Ana Sirovic, Cochair
Scripps Institution of Oceanography, La Jolla, CA 92093-0205

Lora J. Van Uffelen, Cochair
Ocean and Resources Engineering, University of Hawaii at Manoa, Honolulu, HI 96815

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

Martin D. Verweij, Cochair
Acoustical Wavefield Imaging, Delft University of Technology, Delft 2628 CJ, Netherlands

Koen W.A. van Dongen, Cochair
Acoustical Wavefield Imaging, Delft University of Technology, Delft 2600 GA, Netherlands

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

2a TUE. AM

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)

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
TU Berlin, Einsteinufer 25 TA, Berlin 10587, Germany

Klaus Genuit, Cochair
HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

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 /ṣ/ 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 /ṣ/ and /ɕ/ along the centroid dimension. The /ṣ/ 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' p^h t t' t^h s s' c c' c^h k k' k^h/ 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 ɤ]; 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

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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

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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.