Session 5aAA

Architectural Acoustics, Noise, and Speech Communication: Classroom Acoustics in Honor of Michael Nixon

David Lubman, Cochair
DL Acoustics, 14301 Middletown Ln., Westminster, CA 92633-3908

Louis C. Sutherland, Cochair
27803 Longhill Dr., Rancho Palos Verdes, CA 90275-3908

Chair's Introduction—8:30

Invited Papers

8:35

5aAA1. Classroom acoustics: A first step toward education for all. Karen L. Anderson (4736 Tony Sound Ln., Tallahassee, FL 32309, karenlanderson@earthlink.net)

Education is primarily provided through the medium of verbal instruction. With ever greater emphasis on test scores and teacher accountability, it is important to recognize the effects of excessive background noise and reverberation on student learning. Nixon was a leader in the movement to raise awareness of classroom acoustic effects and to achieve national written standards. He crossed professional lines to become an active member of the Educational Audiology Association, providing information and advice to almost 1000 audiologists who work for U.S. school districts. Nixon was instrumental in not only raising awareness, but raised the bar for what education audiologists should know about classroom acoustics. This paper will provide information on the listening challenges of learners with hearing loss (the most common birth defect), most of whom are educated in typically noisy classrooms. Acousticians are invited to learn more about the synergistic effects of excessive noise and reverberation on speech perception of this growing number of children and what the hearing industry is doing to address these issues. Together, the voices of the acoustics and hearing industries are needed to champion the case for listening, learning, and a better future footsteps left by Nixon.

8:55

5aAA2. Next steps toward improving classroom acoustics for all. Peggy Nelson (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, peggynelson@umn.edu)

For the past ten years, the ASA has been very influential in the improvement of classroom acoustics, especially with the adoption of ANSI S12.60-2002. In fact, we have been a part of a worldwide effort to improve the learning environments for children, thanks in large part to the efforts of Nixon. Even after the adoption of the standard, we must still be involved in local and state policy decisions involving classroom acoustics, especially in urban districts with older school buildings, diverse students, and declining enrollment. We can and should form partnerships with personnel in local school districts to evaluate and fix acoustical problems in schools. Recently we worked with a local urban school district to evaluate schools and determine the best investment for improving acoustics. In some schools, sustained noise levels were well over 65 dBA, largely because of poor quality doors and windows that allowed in high levels of external noise from hallways and outside. Solutions included: damping ventilation vibration, adding door seals, and judicious use of amplification systems when higher-signal levels were needed. In most cases, reducing noise problems at the source was the best investment of district funds.

9:15

5aAA3. A green pathway to classroom acoustics: A comparison of classroom acoustic standards. Daniel Bruck (BRC Acoust. Technol. Consulting, 1741 First Ave. S., Seattle, WA 98134, danb@brcacoustics.com) and Alexis Kurtz (Arup, New York, NY 10013)

Towne seems to have started the modern classroom acoustics movement, and Nixon was one of its earliest volunteers. Nixon contributed to an important success of that movement, the development of ANSI standard for classroom acoustics, S12.60-2002. Some of Nixon's collaborations with Towne are remembered. Nixon's efforts are also honored by showing important ways in which the standard influences contemporary practice of school building. The U.S. Green Building Council has adopted elements of the ANSI standard in its evolving LEED for Schools. Why not all? The acoustical requirements of LEED for Schools are compared with S12.60-2002. Ongoing challenges of integrating good acoustics into school buildings in the context of sustainable design are discussed. The authors discuss the issues encountered in development of the LEED standard when acoustics vies with competing needs. The importance of involvement by members of the classroom acoustics community to further Nixon's work is also discussed.

9:35

5aAA4. Issues in the sound absorption treatment in classrooms. Louis Sutherland (Consultant in Acoust., 27803 Longhill Dr., Rancho Palos Verdes, CA 90275) and David Lubman (DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683)

Classrooms require adequate sound absorption to establish appropriate environments for listening and learning. Sound absorbing materials improve the classroom learning environment by reducing reverberation time and by reducing background noise levels. ANSI standard S12.60-2002 provides simple methods for determining the amount of sound absorption needed to limit reverberation time. One of those methods is briefly reviewed. Throughout his career, Nixon's forte was the sound absorption treatment of rooms. He and his colleague, the late Robin (Buzz) Towne, were pioneers—sounding the alarm that, with hardly anyones notice, poor classroom acoustics had become a widespread problem adversely impacting scholastic achievement. The two were also instrumental in motivating the Acoustical Society of America to lead the change toward a solution. The Society is indebted to Nixon both for his pioneering effort to awaken acousticians and others to the need for good classroom acoustics and for his diligent participation in the subsequent effort to develop an ANSI standard for that purpose.

9:55

5aAA5. Classroom acoustics: Moving toward needed regulation. Lois Thibault (U.S. Access Board, 1331 F St. NW, Ste. 1000, Washington, DC 20004)

This year, the U.S. Access Board voted to pursue regulation of classroom acoustics. The initiative born just a decade ago from Mike Nixon's advocacy for adequate listening conditions in schools can now move forward toward enforceability with a new ADA standard based on ANSI/ASA S12.60-2002, the voluntary standard recently reballotted by ASA. Today, Connecticut requires its new schools to meet the S12.60 standard; the LEED sustainability credential rewards good acoustics in its school evaluations, and California's High-Performing Schools movement recognizes acoustical performance. Armed with data Nixon collected and shared and connected by the classroom acoustics LISTSERV he established, stakeholders have been able to influence scores of jurisdictions to require new school designs to address classroom acoustic performance. Nixon's work in improving public understanding of the need for good classroom acoustics has made today's regulatory agenda possible: he provided both the tools and the energy used to build an effective constituency for quiet classrooms. The community of support Nixon encouraged among parents, educators, audiologists, acousticians, and advocates is a necessary underpinning for progress toward regulation. This paper will document the history and anticipate the future of classroom acoustics regulation, focusing on building and access code process and provisions.

10:15—10:30 Break

Contributed Papers

10:30

5aAA6. A survey of unoccupied and occupied acoustic conditions in existing modular classrooms. Norman H. Philipp and Lily M. Wang (Architectural Engr. Prog., Peter Kiewit Inst., Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0681, nphilipp@mail.unomaha.edu)

Modular classrooms are widely in use across the United States, and an addendum on their acoustic performance is being prepared to supplement the classroom acoustics guidelines given in ANSI S12.60-2002. This paper presents measurements made in a large number of existing modular classroom units in Omaha, NE, both in unoccupied and occupied conditions. In the unoccupied classrooms, the gathered data include (a) reverberation times; (b) background noise levels in heating, cooling, and ventilation modes; and (c) airborne sound attenuation across all four exterior walls. In the occupied classrooms, sound levels were logged in both the room interior and exterior throughout an entire school day. Correlations have been made between the interior and exterior data to determine how often the interior noise levels could be attributed to exterior sources. Summaries of the results will be provided in an effort to benchmark the current conditions of modular classroom constructions.

10:45

5aAA7. Comparing classroom acoustics in green and nongreen schools. Pamela Harght and Robert C. Coffeen (School of Architecture and Urban Planning, Univ. of Kansas, 1465 Jayhawk Blvd., Lawrence, KS 66045)

Minimal data are available on existing conditions regarding the acoustical environment in public schools throughout the U.S. Furthermore, rarely is there communication between those involved in the design/build process and the professionals who utilize these spaces on a daily basis. The research presented for this paper was conducted in two parts. First, an electronic survey was issued to teachers of public elementary, middle, and high schools in all 50 states in green and nongreen classrooms. This survey gathered results on questions on the present condition of the teachers' classrooms in terms of background noise levels, noise from adjacent spaces and the outdoors, speech intelligibility, and the level of importance of acoustics for a teacher with regard to other indoor environmental concerns, i.e., thermal comfort, daylighting, and layout of classroom. The second part of this research involved testing of classrooms, both green and nongreen. The testing complied with ANSI S12.60-2002 for testing procedures for reverberation time, background noise levels, transmission loss, and speech interference level. Finally, the results from both parts were compared for nongreen and green classrooms regarding acoustics and classroom design for potential future applications in the school and architecture communities.

11:00

5aAA8. Graphical representation of acoustic data. Michael Ermann and John Samuel Victor (School of Architecture + Design, Virginia Tech, 201 Cowgill Hall, Blacksburg, VA 24061-0205, mermann@vt.edu)

This line of research aims to graphically represent acoustic data for clear comparisons. With a particular focus on low-frequency sound absorption, noise reduction coefficient, and impact insulation class, acoustic data are presented visually and grouped. Trends emerge when looking at some data, and in other cases similar assemblies may have dissimilar acoustic values.

11:15

5aAA9. Gymnasium room acoustics. Comparisons of different metrics, criteria, measurements, and calculation methods (including acoustical modeling software). Joseph F. Bridger and Steven S. Stulgin (Stewart Acoust. Consultants, 7406 L Chapel Hill Rd., Raleigh, NC 27607)

Gyms are nondiffuse large boxes and have unusual room acoustics challenges related to safety, durability, and volume. Classic room acoustics analysis methods, such as that of Fitzroy and Sabine, measurements, as well as acoustical modeling software results are compared. Speech intelligility results from the room acoustics software are compared against traditional reverberation time criteria. What has been learned about criteria and calculation methods is shared.

11:30—11:45 Discussion

Session 5aNS

Noise: Topics in Noise—Active Noise, Product Noise, and Community Noise

Erica Ryherd, Chair
Georgia Institute of Technology, Mechanical Eng., 771 Ferst Dr., Atlanta, GA 30332-0405

Contributed Papers

9:00

5aNS1. Noise levels in computer data centers: Potential occupational noise hazard. Matthew Nobile (IBM Hudson Valley Acoust. Lab., M/S P226, Bldg. 704, Boardman Rd. Site, 2455 South Rd., Poughkeepsie, NY 12601)

The noise levels in modern high-density data centers are encroaching on occupational noise limits, such as those set by OSHA laws in the United States or EC Directives in Europe. This is surprising to many because the high-tech computer data center is not usually thought of as a workplace that can cause hearing damage. Over the past decade two trends have contributed to this. First, high-end servers are being packaged more and more densely into a single rack, where the cooling fans or blowers now have to operate at higher speeds and airflow volumes to properly cool them. Secondly, more and more of these systems are being installed on data center floors, with racks often butted one against the other "as far as the eye can see." An associated problem is as follows: How to predict ahead of time what the sound pressure levels might be in a data center? This involves solving the following emission-to-immission problem: How to translate the emission sound power levels of individual racks into resulting immission sound pressure levels in the room? This paper will present an initial assessment of the noise exposures that exist in today's data centers and present some initial modeling results.

9:15

5aNS2. Lowering the effect of low-frequency noise generated from boiler exhaust stacks. Kevin Richardson and Byron Davis (Vibro-Acoust. Consultants, 490 Post St., Ste. 1427, San Francisco, CA 94102, kevin@va-consult.com)

Low-frequency noise from exhaust boiler stacks at a semiconductor plant was measured and the low-frequency noise spectrum was predicted using an analytical impedance model. Nearby residents complained that noise was rattling household items, such as windows and light fixtures. The semiconductor plant speculated that noise from two boiler exhaust stacks was the source of the problem. Sound pressure level measurements conducted at the plant revealed strong tones in the infrasound and low-audible frequency range. The boiler exhaust stack noise source was theoretically characterized by deriving the mechanical impedance of the boiler exhaust stack and the radiation impedance of the top of the boiler exhaust stack as a simple unbaffled source. The analytical model predicted the resonant frequencies of the boiler exhaust stacks and these resonances closely matched the strong low-frequency tones measured. Using this model, possible changes in controlling parameters of common exhaust boiler stack design are described that could decrease the overall noise radiating from the exhaust stacks. As one example, the plant appreciably decreased the sound power contribution from the upper exhaust stack resonances by lowering the overall output of the exhaust boilers. This change lowered the overall noise levels without the use of silencers.

9:30

5aNS3. Active control of diffuse sound fields using generalized energy density. Buye Xu and Scott Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602, buye.xu@gmail.com)

In enclosures, the total acoustic energy density (ED) has been shown both theoretically and experimentally to be more spatially uniform than the squared pressure. The generalized energy density (GED) has even more uniform properties than the acoustic energy density. In active noise control applications, the standard approach taken is to minimize the squared pressure response in the field. However, the use of ED as the minimization quantity has been demonstrated to yield improved performance in low modal density acoustic fields, often resulting in improved global attenuation. For diffuse acoustic fields (high modal density), local "quiet zones" can be achieved, with the volume of this quiet zone typically being characterized as a sphere with a diameter of about one-tenth of a wavelength when the squared pressure is minimized. It has been found that this performance can also be improved through the use of GED. By controlling GED instead of squared pressure, one cannot only increase the size of the quiet zone but also decrease the acoustic power added into the system. Results will be shown to illustrate the improved performance.

9:45

5aNS4. Active noise control for a short duct. Ho-Wuk Kim and Sang-Kwon Lee (Dept. of Mech. Eng., Inha Univ., Incheon, Korea)

Finite impulse response (FIR) filter for an adaptive filter algorithm is mostly used for an active noise control system. However, an FIR filter needs to be larger in size of the filter length than of its infinite impulse response (IIR) filter. Therefore, the control system using the FIR adaptive filter has slow calculation time. In the active noise control system of the short duct, the reference signal can be affected by the output signal, so IIR filter for the ARMA system can be more suitable for the active noise control of the short duct than FIR filter for the MA system. In this paper, the recursive LMS filter, which is the adaptive IIR filter, is applied for the active noise control inside the short duct. For faster convergence and more accurate control, a variable step size algorithm is introduced for this recursive LMS filter (R-VSSLMS filter). Using this algorithm and considering the secondary path, the filtered-*u* R-VSSLMS is conducted successfully on the real experiment in the short duct. The performance of the active control using the filtered-*u* R-VSSLMS filter is compared with the performance of the active control using a filtered-*x* LMS filter.

10:00

5aNS5. Analysis of commonly witnessed vehicle accident sounds *in situ*. William Neale and Toby Terpstra (Kineticorp, 44 Cook St., Ste. 510, Denver, CO 80206)

Research by Harber and Harber demonstrates the inaccuracy human beings have when recounting their experience witnessing a vehicle accident. However while witness statements can be unreliable, what a witness might have seen or heard is nonetheless important and cannot simply be disregarded. A witnesss recollection of audible sounds such as tire screech-

ing, vehicle acceleration, or impacts between objects can profoundly affect how an accident sequence is interpreted by experts in vehicular accident reconstruction who may rely on witness statements when there is a dearth of physical evidence that properly defines the sequence of events. This paper provides a means for assessing the validity of what witnesses hear by recreating, recording, and measuring commonly heard accident sounds in real world conditions. Sounds, such as engine noise, tire screeching, and impacts, are recreated using various vehicles and in various environments. These sounds are recorded, observed, and analyzed to provide an understanding of how these sounds might be experienced by a witness. Among some of the testing variables are vehicle speed, roadway temperature, vehicle type, roadway surface, and background environment. These factors are evaluated in how they affect sound quality and clarity, frequency spectrum, sound level, and the sound's reverberation and directivity.

10:15

5aNS6. Statistical learning approach applied to road surface classification. Joel Paulo (DEETC, ISEL-Tecnical Inst. of Lisbon, 1959-007 Lisbon, Portugal, jpaulo@deetc.isel.ipl.pt) and José Bento Coelho (CAPS, Instituto Superior Tecnico, TU Lisbon, Lisboa, Portugal)

Measures aiming environmental noise abatement usually consider acoustic barriers alongside the road. However, the cost associated with these measures is usually considerably high and its performance in urban areas is reduced. The problem of the visual impact is another issue affecting the communities. Nowadays, road planners have started to consider silent surfaces as an alternative. These types of surfaces are constituted basically by changing the texture and/or porosity of the mixtures. In some conditions, noise level abatements up to 15 dB can be achieved. Therefore, a considerable variety of different surfaces are available. The main goal of this research is to identify and classify different types of road pavements by analyzing the noise profile, using the close proximity method. Feature extraction and selection is one of the first procedures on a classifier algorithm. Moreover, the accuracy of the results is strongly dependent on the right choice of the selected feature vector. Standard classifiers are being tested in order to establish guidelines for future developments of this research. In situations of net road surveillance, searching for inhomogeneities on the surface and the presentation of the results in a geographic map, showing the locations of the surface types and the noise levels, can improve the accuracy of the noise mapping models.

10:30

5aNS7. Investigation and analysis of urban noise for sustainability. Martha G. Orozco-Medina (Inst. de Med. Amb. y Com. Hum. Dep. de Cs. Amb., CUCBA, Univ. de Guadalajara, Km 15.5 Carretera a Nogales, Las Agujas Zapopan, Jalisco, Mexico.), Arturo Figueroa-Montao, and Javier Garca-Velazco (Univ. de Guadalajara, Jalisco, Mexico)

The concept of urban sustainability involves complex issues such as civil services, social participation, resilience, productivity, health, and development into a multilevel approach of environmental, social, and productivity sectors. Viewing environmental noise as part of urban dynamics is essential. However it is almost absent or underestimated in most current environmental, social, and health policies in the developing world. Efforts to investigate noise pollution toward analysis, regulation, fulfillment, inspection, and fines contribute in a positive way to improve acoustic quality within urban communities. Therefore the ultimate goal for decision makers is to consider noise as a key issue when establishing or discussing management policies of urban areas and to transfer the approach to lower levels with preventive and educational measures, rather than restrictive or prohibited, in order to attain sustainability of urban communities.

FRIDAY MORNING, 14 NOVEMBER 2008 LEGENDS 10, 8:30 TO 11:30 A.M.

Session 5aPA

Physical Acoustics: Outdoor Sound Propagation

Claus Hetzer, Chair

Univ. of Mississippi, Natl. Ctr. for Physical Acoustics, 1 Coliseum Dr., University, MS 38677

Contributed Papers

8:30

5aPA1. Applying well-known diffraction models to the sound field in the shadow zone of an isolated building. W. C. Kirkpatrick Alberts, II, John M. Noble, and Mark A. Coleman (U.S. Army Res. Lab., Attn. AMSRD-ARL-CI-ES, 2800 Powder Mill Rd., Adelphi, MD 20783)

An isolated building, a fundamental case of urban acoustics, has recently been the subject of an experimental effort to characterize the building's influence on propagating sound [Alberts and Noble, J. Acoust. Soc. Am. **121**, 3064 (2007)]. As a precursor to modeling efforts involving finite-difference time-domain simulations, two common diffraction models, the frequency domain model of Pierce and Medwin's extension of Biot and Tolstoy's time-domain model, have been utilized to estimate the sound pressure at various locations in the building's acoustic shadow. The discussion will include the models used and comparisons between calculated and measured data, demonstrating reasonable agreement between measured and calculated spectra in limited cases at some frequencies.

8:45

5aPA2. Finite-difference time domain simulations of outdoor sound propagation around manmade structures. Sandra L. Collier, W. C. Kirkpatrick Alberts, II, Leelinda Parker, and John M. Noble (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783)

We develop a finite-difference time domain (FDTD) model of the acoustic propagation around outdoor man-made structures that includes the effects of atmospheric turbulence and porous ground surfaces. The wind flow around the building is determined with a Navier–Stokes approach, whereas, the basic acoustic propagation model is based on the coupled first-order partial differential equations for linear acoustic propagation in a dynamic environment developed in by [Collier *et al.*, Proceedings of the 2005 MSS BAMS]. Special numerical techniques are required to model the sound interaction at the man-made structure, in particular, for buildings with complicated geometries. For two-dimensional propagation, this numerical model can be run on a standard desktop computer. However, for three-dimensional

propagation, high performance computers are needed. Here we present results from the numerical simulations and compare them to recently collected data [Alberts and Noble, J. Acoust. Soc. Am. **121**, 3064 (2007)].

9:00

5aPA3. Predicted and experimental attenuation of obstructed projectile shock waves. James Perea and Brad Libbey (Army RDECOM CERDEC NVESD, 10221 Burbeck Rd., Fort Belvoir, VA 22060)

Acoustic sniper localization algorithms are well established for open field detection; however, these algorithms are less accurate in urban environments due to reverberation and diffraction. Research is being performed to understand the effects of obstructions on shock wave propagation. Barrier attenuation calculations that use Fresnel number to characterize obscurity will be compared to measured attenuation of shock wave frequency components. An artificial building was set up and shock waves from bullets were recorded by microphones with varying degrees of obscurity. The experimental setup allowed for a range of Fresnel numbers between 5 and 11 resulting in attenuation up to 30 dB. Attenuation increased as obscurity and Fresnel number increased up to a threshold. Beyond this point the magnitude of attenuation decreases in both theory and experiment. These data support the relationship between attenuation and microphone obscurity and may provide a means to estimate the amplitude and *N*-wave slope of the shock wave prior to obstruction.

9:15

5aPA4. Numerical simulation of sonic boom propagation through atmospheric turbulence. Andrew Piacsek (Dept. Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422, piacsek@cwu.edu), Lance Locey, and Victor Sparrow (Penn State Univ., State College, PA 16802)

To better understand the mechanisms by which atmospheric turbulence alters sonic boom rise times and peak overpressures, numerical calculations of sonic boom propagation through atmospheric turbulence have been performed using the NPE time domain model. Turbulence is incorporated into the model as a perturbation of the ambient sound speed that has a random spatial distribution determined by a von Karman energy spectrum. Each simulation employs a new realization of the turbulent field. Output from the finite difference solution includes high resolution movies showing the evolution of a full two dimensional wave field as it propagates from the upper region of the turbulent boundary layer to the ground. The evolution of wave forms corresponding to specific locations along the wave front is also obtained. Results are presented that illustrate how the magnitude, complexity, and spatial and temporal variabilities of the sonic boom wave field depend on turbulence spectrum parameters that represent atmospheric conditions. [V. Sparrow supported by NASA through Wyle Laboratories and by the FAA/NASA/Transport Canada PARTNER Center of Excellence.]

9:30

5aPA5. Acoustic tomography of the atmosphere at the Boulder Atmospheric Observatory. Vladimir E. Ostashev, Alfred J. Bedard (NOAA/Earth System Res. Lab., Boulder, CO 80305), Sergey N. Vecherin, and D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH 03755)

An array for acoustic tomography of the atmosphere has been built at the NOAA Boulder Atmospheric Observatory. In this paper, a short description of the array and some acoustic tomography results are presented. The array consists of three speaker and five microphone towers located along the perimeter of a square with a side length of 80 m. The towers are 9.1 m high. The speakers and microphones can be located at different (multiple) levels on the towers to do three-dimensional tomography. The transducers are connected via cables with the central command and data acquisition computer. The array enables measurements of travel times of sound propagation between different pairs of speakers and microphones. The measurements are done repeatedly within a short time interval so that the information about the temporal change in the travel times can be employed in tomographic reconstruction. Then, these travel times are used as input data in a time-dependent stochastic inversion for reconstruction of temperature and wind velocity fields. Examples of the reconstructed turbulence fields are presented and discussed. [Work supported by the Army Research Office.]

9:45

5aPA6. Comparison of several acoustic models with measured data for a pure tone sound source. Bruce Ikelheimer, Micah Downing, and Michael James (13 1/2 W. Walnut St., Asheville, NC 28801)

A unique data set of outdoor acoustical measurements has been collected in a study area of over 800 sq km. Over 100 pure tone, high-level sources are uniformly distributed within this area. These sources were sounded simultaneously for 4 min on two consecutive days and were recorded by 24 sound level meters also distributed within the study area. In addition to the sound data, surface meteorological data were collected at the sound level meters site along with local area weather service data. The sound level meters collected third-octave band 1-s Leqs, while the weather data collected included wind speed, direction, temperature, humidity, and atmospheric pressure every 5 s. The terrain in the study area is hilly with varying degrees of vegetation and some large bodies of water. This data set provides a unique opportunity to compare various outdoor sound propagation models with measured data for propagation distances from 375 m up to several km. This presentation will compare the measured results with model calculations including simple and complex ray-tracing methods and a parabolic equation method.

10:00—10:15 Break

10:15

5aPA7. Infrasound studies of hurricanes. Claus Hetzer, Carrick Talmadge, Roger Waxler, and Kenneth Gilbert (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, claus@olemiss.edu)

Hurricanes, whether in the Gulf of Mexico or the open ocean, are known to produce infrasound in the microbarom band (around 0.2 Hz). Infrasound technology is an excellent choice for monitoring hurricanes because portable, sensitive, and relatively low-cost infrasound arrays can be deployed quickly in areas out of danger from storm winds and waves while still monitoring storm-related signals continuously. The National Center for Physical Acoustics is developing a Gulf Coast hurricane monitoring system involving permanent and portable infrasound arrays. This discussion will focus the results of infrasound studies of the 2008 hurricane season, along with background information on the history and theory of hurricane microbaroms. The latest results of theoretical calculations about the microbarom source and possible effects of storm winds on bearings will also be presented.

10:30

5aPA8. Frequency response, acoustic impedance and background noise of microbarometers. Damien Ponceau (CEA/DASE, Bruyres le Chtel 91297, Arpajon Cedex, France), Benoit Alcoverro (CEA/DEV, BP2 33114, Le Barp, France), and Serge Olivier (CEA/DASE, Bruyres le Chtel 91297, Arpajon Cedex, France)

Frequency response and background noise of microbarometers are derived from their principle of operation. Lumped element models are proposed to describe each part of these absolute infrasound sensors and are used to derive analytical expressions for frequency response, acoustic impedance, and background noise of microbarometers. A set of experimental methods to estimate all elements is discussed in this paper. Some of them need a specific instrumentation developed in our laboratories. Others are very simple as they require only common equipment and can be applied on field to calibrate these sensors. All these methods have been applied to a set of MARTEC MB2005 microbarometers. Results from the theory and measurements are compared in this paper.

10:45

5aPA9. Characteristics of porous-hose infrasonic wind-noise-reducing filters. Claus Hetzer, Jin So, Carrick Talmadge, Richard Raspet, Jeremy Webster (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, claus@olemiss.edu), and Douglas Shields (MilTec Res. and Technol., Oxford, MS 38655)

Spatial wind-noise-reducing filter technologies are widely used in infrasound recording because wind is the primary source of noise in this frequency band. While permanent infrasound arrays often make use of large solid-pipe rosette spatial filters, portable or temporary arrays must use other materials, the most popular of which is porous garden hose. While undeniably effective at reducing wind noise, porous hoses have distinct frequency-dependent effects on signal amplitudes and by changing the phase response of the microphone can have deleterious effects on bearing accuracy. In this presentation the effects of the use of porous-hose wind-noise-reducing filters are discussed, both in terms of the effectiveness of noise reduction and the effects on signal characteristics.

11:00

5aPA10. Subsurface windscreen for the measurement of outdoor infrasound. Qamar A. Shams, Cecil G. Burkett, Toby Comeaux (NASA Langley Res. Ctr., 4 Langley Blvd., M.S. 238, Hampton, VA 23681), Allan J. Zuckerwar (Analytical Services and Mater., Hampton, VA 23666), and George R. Weistroffer (Virginia Commonwealth Univ., Richmond, VA 23284)

A windscreen has been developed that features two advantages favorable for the measurement of outdoor infrasound. First, the subsurface location, with the top of the windscreen flush with the ground surface, minimizes the mean velocity of the impinging wind. Secondly, the windscreen material (closed cell polyurethane foam) has a sufficiently low acoustic impedance (222 times that of air) and wall thickness (0.0127 m) to provide a transmission coefficient of nearly unity over the infrasonic frequency range (0–20 Hz). The windscreen, a tightly sealed box having internal dimensions of $0.3048\times0.3048\times0.3556$ m^3, contains the microphone, preamplifier, and a cable feed thru to an external power supply. Provision is made for rain drainage and seismic isolation. A three-element array, configured as an equilateral triangle with 30.48 m spacing and operating continuously in the field, periodically receives highly coherent signals attributed to emissions from atmospheric turbulence. The time delays between infrasonic signals received at the microphones permit determination of the bearing and elevation of the source, which correlate well with locations of pilot reports within a 320 km radius about the array. The test results are interpreted to yield spectral information on infrasonic emissions from clear air turbulence.

11:15

5aPA11. High-order parallel discontinuous Galerkin method for time-domain acoustic simulations. Timo Lähivaara (Dept. of Phys., Univ. of Kuopio, P.O. Box 1627, Kuopio, FI-70211, Finland, timo.lahivaara@uku.fi), Tomi Huttunen (Kuava Ltd., Kuopio, FI-70210, Finland), and Simo-Pekka Simonaho (Univ. of Kuopio, Kuopio, FI-70211)

The modeling of acoustic waves in the time-domain poses a significant challenge in scientific computing. A promising candidate for solving the three dimensional wave equation is the discontinuous Galerkin (DG) method. Advantages of the DG method are the easy parallelization and a special matrix structure which can reduce the overall time and the computer memory needed for solving the problem. In this study, a high order parallel DG method is investigated. The DG solver is implemented using the C++ programming language. Communication between processors of the parallel computer is performed using the message passing interface. In the solver, the polynomial degree of the basis functions is chosen individually for each element of the computation mesh (up to ninth order polynomials can be used). The unbounded problem is truncated using the perfectly matched layers. The method is evaluated with numerical simulations that are performed on a personal computer cluster.

FRIDAY MORNING, 14 NOVEMBER 2008 — LEGENDS 7, 9:00 A.M. TO 12:00 NOON

Session 5aSC

Speech Communication: Second Language Perception and Production (Poster Session)

Kanae Nishi, Chair
Boys Town Natl. Research Hospital, 555 N. 30th St., Omaha, NE 68131

Contributed Papers

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

5aSC1. Foreign-accented speech in noise. Jonathan Dalby (Dept. of Audiol. and Speech Sci, Indiana-Purdue Fort Wayne, 2101 E. Coliseum, Ft. Wayne, IN 46805 and Commun. Disord. Technol., Inc, Bloomington, IN 47404, dalbyj@ipfw.edu)

Studies of the perception of foreign-accented English speech have shown that native-speaking listeners are able to adapt quite quickly and effectively to the phonetic and phonological characteristics of non-native speech [Bradlow and Bent, 2003, XVth ICPS Proceedings] and can do so even when the accent is unfamiliar [Clark and Garrett, J. Acoust. Soc. Am. **116** (2004)]. Such studies raise the question of how much cognitive effort underlies this adaptation as well as questions about the robustness of the adaptation under less-than-optimal listening conditions. Rogers *et al.* (2004) have shown that the intelligibility of connected speech, even that from highly proficient non-native speakers, was degraded more than that of native speech when presented in noise. This study attempts to replicate that noise effect for isolated words. Two Spanish-speaking adults with differing oral English proficiency and two American English speakers recorded 50 CVC words from each of three lists that have been shown to be of equivalent perceptual difficulty (NU lists 1, 2, and 4). Words were presented with no noise and at −5 and −10 dB SNR to native listeners. Early results suggest that word identification scores are similar to those found for sentence comprehension. CDT, Inc. markets a speech intelligibility training software. [Work supported by Purdue Research Foundation.]

5aSC2. Quantifying the contribution of contextual information in speech perception in noise for native and non-native listeners. Kanae Nishi, Jessica Lewis, Judy Kopun, and Patricia G. Stelmachowicz (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, nishik@boystown.org)

When speech is degraded, listeners tend to rely on contextual information, if available. The goal of the current study was to evaluate the use of lexical, syntactic, and semantic context for native and non-native listeners. Using an adaptive tracking method, listeners' reliance on contextual information was assessed as signal-to-noise ratios for 70%-correct speech perception performance (SNR70) for single words and short grammatically and semantically appropriate sentences. Adult and child native speakers of American English and adult Spanish speakers learning English as a non-native language served as listeners. For all groups, SNR70 was significantly higher for words than sentences. No group difference was found for sentence SNR70, but significantly higher word SNR70 was observed for the non-native group than the two native groups, indicating greater benefit of context for non-native listeners. When non-native listeners' SNR70 measures were subjected to discriminant analysis to determine similarity to native groups, three subgroups were identified: (1) nativelike for both words and sentences, (2) nativelike only for sentences, and (3) not nativelike for either words or sentences. Possible application of this method to determine candidacy for auditory training to improve speech perception in noise will be discussed. [Work supported by the NIH.]

5aSC3. Effects of speechreading on understanding mainstream American English by second-language listeners of English. Yori Kanekama and David Downs (Dept. of Commun. Sci. and Disord., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260-0075, yxkanekama@wichita.edu)

Cross-language speech perception research has shown that, at least across some languages, first-language speakers of English show more of a McGurk effect for English nonsense syllables than second-language speakers of English. This suggests less auditory-visual integration and auditory-visual incompatibility when listening to nonsense syllables of a different language. Researchers, however, have not studied the role of auditory-visual integration and auditory-visual incompatibility on cross-language perception of ongoing everyday speech. The purpose of this study was to measure the effects of speechreading on understanding mainstream American English (MAE) by second-language listeners of English. Specifically, participants were 30 first-language MAE speakers and 30 first-language Indian speakers recruited from the same graduate school engineering program. They participated in two experiments in which they listened to Central Institute for the Deaf Everyday Speech Sentences under auditory-only, visual-only, and auditory-visual conditions at different signal-to noise ratios. Visual enhancement and auditory enhancement scores were computed and compared between language groups. Results have (1) theoretical implications for understanding the role of auditory-visual integration and auditory-visual incompatibility on cross-language perception of connected speech, and (2) have practical implications for understanding whether speechreading helps or hinders listening to a second language in noisy environments.

5aSC4. The amount of information needed for listeners to detect a foreign accent. Hanyong Park, Kenneth de Jong (Dept. of Linguist., Indiana Univ., 404 Memorial Hall, 1021 E. 3rd St., Bloomington, IN 47405, hanypark@indiana.edu), and Isabelle Darcy (Indiana Univ., Bloomington, IN 47405)

This study examined how much information is needed for listeners to detect a foreign accent. Two factors were considered regarding the amount of information: stimulus length and $L1$ phonotactics. Four Korean-English bilinguals and two native speakers of American English produced different lengths, but still short stimuli: the vowel /ɑ/, monosyllabic and disyllabic English words. The monosyllabic corpus, in particular, included the stimuli having both natural (i.e., CV) and unnatural syllable structures (i.e., CCV, CVC, and CCVC) as well as various segments in terms of Korean phonotatics. After being presented with a stimulus, eight native listeners were asked to judge whether the speaker of the stimulus was a native or a non-native speakers of American English. The examination of d' value indicates that all the listeners detected a foreign accent from hearing the monosyllabic and disyllabic stimuli, but only some listeners did from hearing the vowel /ɑ/. Furthermore, the listeners detected a foreign accent more often from the stimuli having the coda segment. Lastly, d' values for the stimuli having /l/ were lower than those for other stimuli. All these suggest that listeners may not need much information to detect a foreign accent, which, in turn, is closely related to $L1$ phonotactics.

5aSC5. Patterns of cross subject correlation in second language learning: Skill structure and feature-level grouping in production and perceptual learning. Kenneth de Jong, Yen-Chen Hao, and Hanyong Park (Dept. of Linguist., Indiana Univ., 322 Memorial Hall, Bloomington, IN 47405)

This study examines the degree to which achievement of accuracy in distinguishing contrasts in one set of segments tends to correlate with achievement of accuracy in a related set of segments across listeners. Forty Korean listeners identified anterior obstruents as produced by four American English speakers before, after, and between instances of the vowel /a/. Accuracy rates for fricatives and stops, differing in voicing and in point of articulation, correlated across the listeners, indicating that some listeners were better at manner distinctions as a class. Similarly, accuracy in different voicing contrasts also correlated across listeners, though only when consonants were in the same prosodic location. Voicing accuracy did not correlate with manner accuracy, indicating that particular listeners were specifically good at particular featural contrasts. The productions of 20 Korean listeners of the same consonants in the same prosodic locations were identified by 10 American listeners. Even though the same range of accuracies was found for production as perception, accuracy for various segments typically did not correlate across talkers. These results suggest that, while perceptual learning tends to generalize across segments along featural lines, production learning is more specific to particular segments.

5aSC6. On the relationship between the perception and learning of Hindi voicing and place contrasts by native speakers of American English. James D. Harnsberger (Dept. of Commun. Sci. and Disord., Univ. of Florida, Gainesville, FL 32611)

Many non-native speech sounds are challenging to perceive and, ultimately, to acquire. Predicting specific learning outcomes from perceptual data has been hampered by (1) problems in quantifying acoustic-phonetic similarity between non-native and native sounds, (2) calculating the perceptual weighting of acoustic cues in the native language, and (3) the high-variability commonly observed in the perceptual assimilation of non-native contrasts to multiple native categories. This variability may reflect long-term persistent patterns in learning or it may represent only a brief early stage prior to fossilization. To examine these two possible accounts, the perceptual assimilation by 15 American English listeners of 7 voicing and place contrasts produced by 6 Hindi speakers was examined before and after training in a paired-associate word learning task. Training utilized tokens from four of the six talkers used in perceptual assimilation tasks. The purpose of the study was to determine whether or not assimilation patterns could be greatly modified and simplified by very limited experience in acquiring the contrasts via word learning. The results showed that limited laboratory training significantly reduced talker and token variability in perceptual assimilation and allowed for non-native contrasts to be categorized more cleanly in terms of their predicted difficulty in learning.

5aSC7. Acquiring novel perceptual categories in a third language: Bengali-English bilinguals' perception and learning of Malayalam consonants. Divya V. Gogoi (Program in Linguist., Univ. of Florida, Gainesville, FL 32611)

The present study examines the acquisition of novel non-native speech contrasts by adult bilingual speakers of Bengali and English. One of the underlying issues in this study is the role that phonetic features may play in the development of new phonetic categories. For instance, features utilized in native contrasts may generalize in the learning of novel non-native contrasts, even if they play a limited or no role in the initial perception of these non-native contrasts [Polka (1992); Harnsberger (1998)]. To explore this feature generalization hypothesis, a high variability consonant identification training paradigm was used with ten Bengali-English bilinguals who learned to identify 4 Malayalam place of articulation contrasts (dental versus retroflex nasals, dental versus retroflex lateral approximants, palatoalveolar versus retroflex voiceless fricatives, and alveolar tap versus retroflex approximant). The bilingual Bengali-English listeners were selected for their extensive experience with the relevant place features (e.g., dental, retroflex, palatoalveolar, and alveolar), though Bengali and English lack any direct correspondents to the Malayalam contrasts. Bilingual performance was analyzed in terms of both patterns of perceptual assimilation as well as rate of acquisition of the novel non-native contrasts.

5aSC8. Tracking non-native acquisition of the Spanish tap-trill distinction: Cross-modal priming differences between native and non-native Spanish speakers. Wendy Herd and Joan Sereno (Linguist. Dept., Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66044, wenherd@ku.edu)

English-speaking learners of Spanish often fail to achieve nativelike pronunciation of the tap-trill distinction in words like *caro* "expensive" and *carro* "car." The trill proves difficult because it is neither a phoneme nor an allophone in English. Although the tap exists as an allophone of /t/ and /d/ in American English, learners of Spanish must learn to process it as a phoneme rather than an allophone. Similarly, English learners of Spanish have difficulty acquiring the spirantization of voiced stops, i.e., /d/ spirantizes in *codo* "elbow," which occurs in the same environment as flapping. This study uses a cross-modal priming task to investigate whether L2 Spanish learners are able to process intervocalic tap, trill, /d/, and /t/ in the same way as L1 Spanish speakers. Using a cross-modal priming paradigm, eight English-speaking learners of Spanish were compared to eight native Spanish speakers. Results show that auditorily presented words with intervocalic taps resulted in faster response times for words like *cada* for learners of Spanish, but the same auditory stimuli resulted in faster response times for identically matching targets like *cara* for native Spanish speakers. These results suggest that cross-modal priming can be used to track L2 acquisition.

5aSC9. The English l sound produced by Korean students. Byunggon Yang (English Education Dept., Pusan Natl. Univ., 30 Changjundong Keumjunggu, Pusan 609-735, South Korea)

This study examined the lateral l sound produced by 16 Korean students in order to tap a possibility of using acoustical and perceptual criteria to distinguish lateral variants and eventually to assess student's English pronunciation skills. The subjects read a short story in a quiet office at normal speed. Those words with the lateral sound in onset or coda positions and before a vowel of the following word were analyzed using PRAAT. The following results are shown. First, the majority of the subjects produced the clear l regardless of the contexts. Some students produced the sound as the Korean flap or the English glide /r/. A few missing cases were also seen. Second, the dark l was mostly produced by the subjects of English majors in coda position with a few cases before a vowel in a phrase. Visual displays from the computer analysis were helpful in determining lateral variants but sometimes personal listening to the given sound after temporal manipulation would be necessary in the cases of fast and weak productions of the target words. Further studies would be desirable to compare native productions of the lateral sound with those of non-native speakers.

5aSC10. Effects of childhood exposure to a second language on the production of voice onset time and closure duration. Tetsuo Harada (Dept. of Education, Waseda Univ., 1-6-1 Nishi Waseda, Shinjuku, Tokyo 169-8050, Japan, tharada@waseda.jp)

This study compares the production of voice onset time (VOT) and closure duration for singletons and geminates in Japanese by English-speaking university students who were exposed to Japanese in an immersion program in childhood and those regular university students who had no previous exposure to Japanese. 20 informants enrolled in a third-year Japanese were asked to repeat several target words including initial /p, t, k/ for VOT, and medials /p, t, k/ and /pp, tt, kk/ for singletons and geminates in a sentence frame. Both VOT of the initial stops and closure duration of the medial stops were measured. The results show that the immersion graduates' VOT values in Japanese were shorter (i.e., more Japanese-like) than those of the learners who had had no exposure to Japanese in childhood ($p<0.05$). For the production of closure duration, although the learners of Japanese without childhood L2 experience did not distinguish singletons from geminates, the immersion graduates did ($p<0.005$). The findings may suggest that long-term benefits of L2 experience in childhood in the naturalistic setting, which were found in the work of Knightly *et al.* (2003), may also apply to the instructional setting like immersion education.

5aSC11. A computer graphic three-dimensional tongue and lip movement synchronized with English fricatives for Japanese learners. Toshiko Isei-Jaakkola (Dept. of English Lang. and Culture, Chubu Univ., 1200 Matsumoto, Kasugai, Aichi 487-8501, Japan, tiseij@isc.chubu.ac.jp), Shigeki Suzuki (Tokyo Univ. of Social Welfare, Nagoya, Aichi 460 0002, Japan), Shigeo Morishima (Waseda Univ., Shinjuku, Tokyo 169-8555, Japan), and Keikichi Hirose (Univ. of Tokyo, Bunkyo, Tokyo 113-8656, Japan)

Some English fricatives are difficult specifically for Japanese learners of English (JL2) to produce. Simultaneous articulation of the lip and teeth (as in lebiodentals), or the tongue and teeth (as in dentals), or protruding lips (as in postalveolars) do not exist in standard Japanese. To understand partially or completely, invisible articulatory movements are unavoidable for JL2 in order to produce these fricatives properly. Thus, as an aid for the basic pronunciation training, a visualized automatic lip and tongue movement program synchronized with these fricatives was developed, utilizing three-dimensional computer graphic technologies. In this program, not only the lips, teeth, and tongue but also the other necessary speech organs were made to be half transparent. Consequently, it enables the learner to listen to and repeatedly model the target segmental sound while looking at these articulatory movements vividly from all kinds of angles. This system changes the phonetic practice situation from two-dimensional paper-based learning/teaching methods with audio-visual tools into revolutionary user-friendly method in phonetic class and outside the classroom. In addition, this system is applicable to any language sounds in the near future. [Work supported by JSPS and Chubu University Grant (A).]

5aSC12. Adapting second language phonemic perception training to common instructional situations: Initial results. Thomas R. Sawallis and Michael W. Townley (English Dept., Univ. of Alabama, Tuscaloosa, AL 35487, tsawalli@bama.ua.edu)

Although current L2 pedagogy de-emphasizes phoneme-level pronunciation training, laboratory experiments demonstrate benefits from training non-natives in perception of difficult target-language phonemic contrasts. Specifically, evidence shows that learners' perceptual performance improves (Jamieson Morosan, 1986; Flege, 1995), improvements generalize to new talkers and words (Lively, Logan, Pisoni, 1993), perceptual training triggers production improvements (i.e., without production training, Bradlow *et al.*, 1997), and both perceptual (Lively *et al.*, 1994) and production improvements (Bradlow *et al.*, 1999) are maintained over several months. These laboratory studies typically used intensive protocols, with long sessions, several days per week, for 2–3 weeks. We have adapted such protocols for use in common L2 instructional situations, using short sessions spread over a longer study period, and have begun training Japanese and other Asian students on the English /l-r/ contrast using this new regimen. This paper reports on initial results and some comparisons with studies using the earlier more intense protocols.

5aSC13. A nonsense syllable confusion matrix task in Korean-English bilingual children. Seok-Youn Yoon (Dept. of Speech Hearing Sci., Univ. of Ilinois, 405 Mathews, Urbana, IL 61801, yoon5@uiuc.edu), Cynthia J. Johnson, and Jont B. Allen (Univ. of Illinois, Urbana, IL 61801)

Compared to extensive studies in bilinguals in adults and very young children, very few studies have investigated speech perception in school-aged Korean-English bilingual (KEB) children. Given that English (L2) and Korean (L1) have different contrast systems among consonants and vowels and children have different speech perception processing than adults, school-aged KEB children are expected to have more complicated patterns of L2 perception. The main goal of the present study was to discern which set of L2 sounds are salient enough not to experience interference from L1. Ten KEB children at age 8–13 years were asked to identify 30 nonsense syllables of L2 sounds. The results showed that dissimilar consonants to Korean (i.e., /f,θ/) are most confusing to perceive than similar consonants with familiar contrasts (/p,k,b,g/) next to similar consonants with unfamiliar contrasts (/ʧ,ʤ,s,z/). More interestingly, perception performance in vowels showed the opposite pattern: a similar vowel (/ɛ/) was less correctly identified than unfamiliar vowels (/I,æ/). The present study suggests that L2 consonants, which cannot be mapped to any of the L1 sounds, are the most challenging to perceive. Nonetheless, groups of sounds that are patterned together cannot simply be explained with the L1 influence. For example, /f,θ/ are most confusing to KEB as well as English-monolingual listeners (Phatak & Allen, 2007); yet confusion of /b/ with /θ/ is unique only to KEBs.

5aSC14. An epenthetic vowel between consonantal sequences in perception and production by Japanese. Mieko Sperbeck and Winifred Strange (Dept. of Linguist., City Univ. of New York-Graduate School and Univ. Ctr., 365 Fifth Ave., New York, NY 10016, msperbeck@gc.cuny.edu)

Native Japanese have a strong tendency to epenthesize a vowel when producing consonantal sequences in English. This study investigated how Japanese learners of English perceive and produce word-initial CCV versus CəCV contrasts (e.g., sport versus support). Two types of tasks were employed: a categorial ABX task for perception and a delayed imitation task for production. Nonsense words of the form /C1C2ani/ (e.g., spani) and /C1əC2ani/ (e.g., sepani) served as the stimuli where C1C2 combinations were /s-p/, /s-t/, and /s-k/. In the ABX task, participants heard three short sentences that contained the target words and answered whether the third target word was the same as the first or the second one. In the delayed imitation task, participants heard the production of a native speaker in a carrier sentence (e.g., say sepani now) twice and produced the target word in isolation and then in the carrier sentence. Japanese participants made both perception and production errors. Interestingly, the majority of errors in the production task were vowel deletion for the CəCani contexts rather than vowel epenthesis for the CCani contexts. The relationship between perception and production among $L2$ learners as well as influence of $L1/L2$ phonotactic differences will be discussed.

5aSC15. Production and perception of English vowel categories by native Korean speakers. Ga Yeon Son (Univ. of Pennsylvania, 619 Williams Hall, 255 S. 36 St., Philadelphia, PA 19104, gson@babel.ling.upenn.edu)

This study deals with the production and perception of a second language ($L2$) by non-native speakers assessing phonetic convergence in their production and perceptual divergence. Production and perception of nine American English monophthongs and eight Korean monophthongs by the experienced and inexperienced Korean speakers were acoustically analyzed. The experiments consist of the production test, identification tests, and discrimination test. The experienced group showed relatively successful production of English vowels that have dissimilar acoustic properties with Korean vowels, while the inexperienced group showed complete phonetic interlingual identification. However, in perception, both groups successfully scored in the identification and discrimination tests, not showing perceptual identification between English and Korean. It suggests that two language systems mutually influence in one native phonetic space, and linguistic experience with $L2$ brings the reorganization of the phonological system, adding new phonemic categories for dissimilar $L2$ phones. The biased results in production and perception tests demonstrate that production of $L2$ is strongly related to the native phonemic categories, but perception is not. Therefore, the findings suggest that two different phonological systems are incorporated into the same phonetic space and linguistic experience with $L2$ boosts this process.

5aSC16. The influence of signal-to-noise ratio and listener's native language on vowel intelligibility. Tessa Bent and Diane Kewley-Port (Dept. Speech and Hearing, Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47401, tbent@indiana.edu)

In a previous study, intelligibility rankings of American English (AE) talkers were different for native and non-native listeners [Bent, *et al.*, J. Acoust. Soc. Am. **122**, 3016 (2007)]. To equate overall performance, the native and non-native listeners were tested at two different signal-to-noise ratios (SNRs). Therefore, the differences in intelligibility rankings between the two listener groups may have been affected by the listener's native language and differences in SNR. The current study explored this issue by testing across-talker differences in AE vowel intelligibility for AE listeners and Korean listeners learning AE as a second language. Listeners heard recordings of ten AE vowels in /bVd/ context produced by ten talkers under three SNRs: -8, -5, and -3 dB. The words were presented for identification in a 10-alternative forced choice task. Results showed that when AE and Korean listeners' were tested under the same SNR, the across-talker intelligibility scores were highly correlated for the two listener groups. However, Korean listeners were less accurate at vowel identification compared to the AE listeners. Furthermore, Korean listeners' intelligibility rankings of the talkers were more variable across SNRs whereas the AE listeners were more consistent. (Work supported by NIH DC02229 and T32-DC00012).

5aSC17. Exposure effects on production and perception development for the learners of English as a second language. Seokhan Kang (Dept. of Linguist. and Philosophy, MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

Exposure effects for L2 production and perception development were examined. Two types of intervowel VCV stop voicing with different accents (iambus and trochee) were presented to ten English native speakers, ten English-learning Korean university students who had residence experience in America, and ten English-learning Korean university students who had no residence experience in America. Results showed that L2 patterns of Korean students' production and perception with residence experience in America were in between the English native speakers and the Korean students without residence experience. However, the asymmetrical development patterns between the production and perception for the two Korean groups could be found. In the production tests, the prosodic factor deeply influences on the asymmetrical development. The difference is in the trochee pattern rather than in the iambic pattern. It means that residence experience improves prosody knowledge in their production. Also this effect has some influence on both the temporal and fundamental frequency features but not on the formant features. The acoustic difference could be seen in the perception tests, too. We could not find any difference in the identification test of the word-medial iambic environment. Rather the difference could be found in the cue influence test in the trochaic environment.

5aSC18. Perceptual cues to English lexical stress: Comparison between native speakers and Mandarin second language learners. Yuwen Lai and Joan Sereno (Dept. of Linguist., Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Lawrence, KS 66044)

The present study investigates the role of $F0$, duration, and vowel quality in English lexical stress perception by second language learners with a tonal L1. Mandarin speakers learning English as a second language (advanced learners, n=25; beginning learners n=25) were compared to a control group of native English speakers (n=25). Resynthesized disyllabic nonwords (dada) were presented in a stress localization task. Simultaneously manipulating five different $F0$ ratios (ratio of maximum fundamental frequency values of the first and second vowels) and five different duration ratios (ratio of duration values of the first and second vowels), stimuli were synthesized based on acoustic measurements from a previous study [Y. Lai and J. Sereno, J. Acoust. Soc. Am. **121**, 3071 (2007)]. Both full-vowel and reduced-vowel stimuli were used. The results indicated that full vowels most often attracted stress across all three listener groups. More interesting, beginning second language listeners relied mainly on duration cues to determine the stressed syllables and advanced listeners focused more on $F0$ cues, while native listeners made use of both cues. The findings will be discussed in terms of the similarities and differences between the prosodic systems of Mandarin and English.

5aSC19. Intonational interference on the perception and production of Mandarin Chinese tones by English-speaking second language learners. Yen-Chen Hao (Dept. of Linguist., Indiana Univ., 2451 E. 10th St. Bloomington, IN 47408, yehao@indiana.edu)

Most previous studies on $L1$ transfer focus on segments. The present study investigated the effect of $L1$ intonation on English speakers' acquisition of Mandarin Chinese tones. Four intermediate learners read three sentence types: declaratives, yes/no questions, and a list sequence, both in English and Chinese. Their productions showed Tone 3 (low) to Tone 2 (rising), and Tone 4 (falling) to Tone 1 (high) errors, as expected if productions have a (English) terminal rise imposed on them. Nonfinal items in reading a list exhibited similar errors. In the perception task, subjects listened to Chinese productions of the different sentence types and identified the tone and whether the sentence is a statement or a question. When subjects identified a sentence as a question, they tended to judge the final tone to be Tone 2 (rising) or Tone 1 (high). Similarly, with final Tone 2, subjects usually called the sentence a question; and with final Tone 4, subjects often judged it as a declarative. English-speaking learners associate the rising tone with questions and falling tone with declaratives, even though they know the final word can carry any tone in Chinese. This study quantitatively documents an effect of intonation on the learning of lexical tones.

5aSC20. Identification of Mandarin coarticulated tones by inexperienced and experienced English learners of Mandarin. Yunjuan He and Ratree Wayland (Program in Linguist., Univ. of Florida, Box 115454, Gainesville, FL 32611-5454)

Two groups of native English speakers, relatively inexperienced (IE) ($N=14$) with 4 months of Mandarin study and relatively more experienced (EE) ($N=14$) with 12 months of study, were asked to identify coarticulated Mandarin lexical tones in disyllabic words. The results show that the EE group was better at identifying Mandarin tones than the IE group. Interestingly both groups were more accurate at identifying tones of the second (i.e., last) syllables than tones of the first syllables. Two types of errors were found in both groups: tonal direction misperception and tonal height misperception. EE committed fewer tonal direction errors than IE. However, EE still made considerable amount of tonal height errors. These results suggest the following: (1) The ability to perceive coarticulated tones improves with learning experience. (2) Due perhaps to a recency effect, final tones are remembered better than initial tones, and final syllable tones are misperceived less frequently than initial syllable tones. (3) The ability to identify tonal direction may improve faster than the ability to identify tonal height among English speakers and thus while tonal direction errors decrease with experience, misperception of tonal height remains even with increasing learning experience.

5aSC21. The perception of English lexical stress by native Thai speakers. Jirapat Jangjamras (Linguist. Program, Univ. of Florida, 4131 Turlington Hall, Gainesville, FL 32611-5454, jirapat@ufl.edu)

This study investigates the influence of $L1$ stress patterns and stress cues on the perception of $L2$ lexical stress. Thai and English were examined due to their prosodic differences such as the stress pattern and native perceptual correlates of stress. Thai, being a tone language, has a fixed stress pattern and employs the duration contrast as the primary stress cue. In contrast, English has a variable stress pattern and employs a combination of acoustic cues such as $F0$, duration, intensity contrast, and vowel reduction. Perceptual difficulties by listeners from a fixed-stress background have been reported in $L2$ stress discrimination and identification tasks. In this study, two groups of listeners, 35 Thai and 10 American English (AE), identified the stress location of disyllabic English nonwords produced by a trained phonetician using the above stress cues except vowel reduction. The preliminary results show similar mean scores between the two groups with a bias toward initial stress in misidentified tokens by AE listeners. Further analysis will report the reaction time analysis and the interaction between the syllabic structure and stress location.

FRIDAY MORNING, 14 NOVEMBER 2008 LEGENDS 12, 9:00 TO 11:15 A.M.

Session 5aSP

Signal Processing in Acoustics: Detection, Estimation, and Classification of Acoustic Signals

Angelo J. Campanella, Chair
Campanella Associates, 3201 Ridgewood Dr., Columbus, OH 43026-2453

Contributed Papers

9:00

5aSP1. Ocean surface degradation of shallow water acoustic communication. Geoffrey Edelmann (U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Code 7145, Washington, DC 20375, geoffrey.edelmann@nrl.navy.mil), Shaun Anderson (Georgia Inst. of Technol., Atlanta, GA 30332-0405), and Paul Gendron (U.S. Naval Res. Lab., Washington, DC 20375)

The accurate prediction of underwater acoustic communication system performance at high frequencies is essential to system risk mitigation prior to deployment. The deterioration of acoustic coherence by ocean surface dynamics will be predicted via three dimensional (3-D) ray tracing with the inclusion of interaction with a realistic dynamic ocean surface. Acoustic communication performance limits at high frequencies for most fixed platform systems are dominated by multipath interaction with the spectrally rich heaving ocean. The high-fidelity Wave-watch III surface spectra model is used here with a wave-number integration approach for surface time series modeling. A fully 3-D ray tracing model is used to model the propagation of sound under a dynamic surface boundary at the time and spatial scales that dominate coherence degrading effects for underwater acoustic communications. The proposed method will serve the rapid development of operating regimes for high-frequency underwater acoustic communication systems. Proposed applications include distributed underwater sensor networks and persistent deployable low-cost systems that meet a wide range of underwater sensing needs. This work seeks to solve problems in deploying advanced communication systems by predicting the degradation of signals due to ocean variability and automatically selecting appropriate signaling rates, schemes, and constants. [This work was supported by ONR.]

9:15

5aSP2. Acoustic monitoring of severe weather in the Northeast Pacific Ocean. Jeffrey Nystuen (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105, nystuen@apl.washington.edu) and Svein Vagle (Fisheries and Oceans Canada, Sidney, BC V8L 4B2, Canada)

Wind and rainfall are the principal physical processes responsible for the production of high-frequency (1–50 kHz) ambient sound in the ocean. The primary source of the sound is the resonant ringing of individual bubbles created during wave breaking and raindrop splashes. Larger bubbles (>300 μm diameter) quickly return to the surface, while smaller bubbles can be mixed downward at several meters. During severe weather a layer of smaller ambient bubble forms and effectively absorbs higher-frequency (>10 kHz) sound. These processes are revealed in a two-year record of ambient sound recorded from a subsurface mooring at 50N, 145W in the NE Pacific Ocean as part of the Canadian SOLAS program. The passive acoustic signal of wind, rain, and ambient bubble clouds are compared to the subsurface mooring data, including data from an upward looking 200 kHz active sonar and a 300 kHz ADCP. The acoustic signatures of light, moderate, and heavy rainfall are superimposed on the signature of high wind, demonstrating rainfall detection even in the presence of high wind. [Work supported by ONR, Fisheries and Oceans Canada, and the Canadian CFCAS NSERC.]

9:30

5aSP3. Model based target depth estimation using received signal statistics. Colin Jemmott (Grad. Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804, cwj112@psu.edu), R. Lee Culver, Brett Bissinger, and Nirmal Bose (Penn State Univ., State College, PA 16804)

The overall goal of our research is to develop model based sonar signal processing techniques that utilize predictions of the acoustic field while being robust to environmental variability and uncertainty. Specifically, we are estimating the depth of a target broadcasting a tonal signal in shallow water using a single passive hydrophone. Three different techniques, naïve Bayes, histogram filter, and estimator correlator, are examined. Each is applied to event S5 of the SWellEX-96 measurement that took place in shallow water off the coast of California. The data consist of sinusoidal signals transmitted simultaneously from moving sources at two different depths and received at a bottomed horizontal array. Received signal amplitudes are modeled using random access memory for the two sources at different depths, and these modeling results are used in the design of the target depth estimation algorithms. Thus, the ability of each of the techniques to estimate target depth depends on an accurate representation of the uncertainty in the acoustic field due to propagation effects. Despite each technique requiring different assumptions, general conclusions about their effectiveness to solving this problem can be drawn. [Work supported by ONR Undersea Signal Processing, Code 321US.]

9:45

5aSP4. Bat-inspired distance measurement using phase information. Said Assous, Peter Jackson, Clare Hopper, David Gunn, John Rees, and Mike Lovell (Ultrasound Res. Lab., Dept. of Geology, Univ. of Leicester, LE1 7RH, UK)

This paper shows the use of phase measurement to estimate the distance to a target. Inspiration for this work comes from the observation that bats have been shown to have exceptional resolution with regard to target detection when searching during flight. Au and Simmons ["Echolocation in dolphins and bats," Phys. Today **45(7)**, 40–45 (2007)] concluded bats with a center frequency of about 80 kHz (i.e., 4-mm wavelength), and 40-kHz bandwidth can have a resolution of distance in air approaching 20 μm. For this frequency, we see that the resolution achieved by the bat is about 200 times better than $\lambda/2$ (i.e., 2 mm at this frequency), which is usually used as a guide for resolution for analog systems. Moreover, Au and Simmons show, using time-frequency analysis, that there are essentially two frequencies present at any particular time within a single bat pulse. Considering this use of two frequencies we may infer a distance. A new bat-inspired algorithm is presented. This is based on phase measurement and, when applied to underwater acoustics, shows that a resolution of 1/50 of the wavelength can be achieved in practice.

10:00—10:15 Break

10:15

5aSP5. Contrast models and classification of active sonar signals. Charles F. Gaumond, Colin W. Jemmott, and Derek Brock (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320)

New features and algorithms are derived from human perception to improve the automation of active sonar signal classification. Data from human subject research on the perceived similarity of sound pairs can be assembled in the form of a dissimilarity matrix. A dissimilarity matrix can be analyzed in many ways, two of which are multidimensional scaling (MDS) and additive clustering (AC). Different underlying perception models can be inferred from each analysis technique: a feature vector space for MDS and a contrast model for AC. The relevance of each technique to the dissimilarity data is inferred from the quantification of model complexity to the measure of fitting the original data [D. J. Navarro and M. D. Lee, "Common and distinctive features in stimulus similarity," Psychon. Bull. Rev. **11**, pp. 961–974 (2004)]. An experimental dissimilarity matrix is shown and analyzed using MDS and AC. Potential features for each case are shown from singular value decomposition of various signal representations. Signal processing implications of real valued features in the contrast model and alternative implementation are discussed. [The authors gratefully acknowledge conversations with and results from Dr. Jason Summers. Work supported by the Office of Naval Research.]

10:30

5aSP6. Development and analysis of chirplet signal decomposition for detection, estimation, and classification in sonar and nondestructive evaluation (NDE) applications. Yufeng Lu (Dept. of Elec. and Comput. Eng., Bradley Univ., 1501 West Bradley Ave., Peoria, IL 61625, ylu2@bradley.edu), Erdal Oruklu, and Jafar Saniie (Illinois Inst. of Tech., Chicago, IL 60616)

In sonar and ultrasonic nondestructive evaluation applications, backscattered signal contains information pertaining to size, shape, and orientation of reflectors. The echoes from target reflectors are often overlapping and masked in the presence of high clutter. Therefore, it is a challenging problem for target localization and object recognition. In this paper, a chirplet model for an adaptive signal decomposition and parameter estimation algorithm is investigated. The proposed chirplet method is capable of representing a broad range of echo shapes, including narrow-band, broad-band, symmetric, skewed, and nondispersive/dispersive. Chirplet transform is used not only as a means for time-frequency representation but also to estimate the echo parameters, including amplitude, time-of-arrival, center frequency, bandwidth, phase, and chirp rate. These parameters embody the dispersion effect, frequency dependent absorption, and scattering in complex and inhomogeneous environments. Experimental data, including ultrasonic backscattered signal and bat chirp signal, have been analyzed to evaluate the algorithm. It has been shown that the adaptive algorithm performs robustly, yields accurate echo estimations, and results in considerable SNR enhancements. Furthermore, system-on-a-chip implementation based on FPGA has been developed to demonstrate the feasibility of real-time applications. This type of study addresses a broad range of applications, including pattern recognition, deconvolution, target sizing, and characterization.

10:45

5aSP7. An image processing based neural network method of wave form classification. Jeffrey Vipperman and Brian Bucci (648 Benedum Hall, 3700 O'Hara St., Pittsburgh, PA 15261, jsv@pitt.edu)

In an effort to identify military impulse noise, as it relates to civilian damage and disturbance claims, several metric based approaches utilizing artificial neural networks and Bayesian classifiers have proven successful in addressing this issue. However, in the course of research, it became apparent that the noise sources to be classified, namely, various types of military impulse noise, wind noise, and aircraft noise, could be easily identified by a minimally trained observer by way of a simple visual inspection of the wave form. Additionally, since the noise classification algorithm is desired to be implemented on DSP boards with possibly limited computational resources, it may prove beneficial to avoid the computation of metrics which involve complex mathematical operations. Borrowing from proven artificial neural network techniques already proven in the field of optical character recognition, this proposed noise classifier views a captured wave form as a number of points located on a spatial grid. The density of points within each grid sector is then used as input to an artificial neural network. The resulting classifiers performed with accuracies up to 0.997 on testing data. [This research was supported by the U.S. Department of Defense, through the Strategic Environmental Research and Development Program (SERDP).]

11:00

5aSP8. A data fusion and multiple ping method for improving the resolution of low-power acoustic and seismic sensing. Alexander Apartsin and Nathan Intrator (School of Comput. Sci., Tel-Aviv Univ., Ramat Aviv 69978, Israel)

Biological research has shown that some mammals use low-power seismic pulses for underground communications and navigation. For instance, the blind mole rat bangs its head against the walls of its underground tunnel to produce seismic signals. It has been demonstrated that this animal builds a three-dimensional map of the environment by collecting and analyzing

returned responses. This behavior enables the mole rat to avoid obstacles and voids while it digs its tunnels. Being able to use low-power signals for accurate remote sensing under low signal to noise ratio (15–25 dB) conditions has some important applications (e.g., underground installment detection). A practically used method for coping with noise is based on repeated transmission of identical pulses and averaging the peak value of autocorrelation function from multiple measurements. Our research suggests that by carefully selecting a pulse family and a fusion method of multiple measurements, one can significantly improve sensing resolution. In particular, our simulation results show that employing robust statistics fusion method and by producing sonar pings with a shifting phase, the average error rate of travel time estimates can be reduced by about 30%.

FRIDAY MORNING, 14 NOVEMBER 2008 LEGENDS 8, 9:00 TO 11:15 A.M.

Session 5aUW

Underwater Acoustics: Deep Water Propagation

Michael G. Brown, Cochair
Univ. of Miami, RSMAS-AMP, 4600 Rickenbacker Cswy., Miami, FL 33149-1098

Wendy Saintval, Cochair
Rensselaer Polytechnic Inst., Math Sci., 110 8th St., Troy, NY 12180

Contributed Papers

9:00

5aUW1. Mode coupling induced by random ocean sound speed structure: Mean intensity and revisiting the evolution equations for the cross-mode coherences. John Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943) and Andrey Morozov (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

The seminal papers of Dozier and Tappert (1978), Dozier (1983), and Creamer (1996) have analyzed mode coupling induced by a random ocean sound speed structure to derive transport equations for the range evolution of mean mode energy. A major assumption in this work is that the cross-mode coherences rapidly decay with range and can thus be ignored. Theoretical predictions of mode energy have been compared to a direct Monte Carlo simulation, yielding favorable results [Dozier and Tappert (1978) and Creamer (1996)], so the general accuracy of the transport equations cannot be questioned. However, if the observable of interest is the mean acoustic intensity, one will quickly realize that the cross-mode coherences do not decay rapidly with range and thus need to be estimated for accurate acoustic predictions; an example from deep ocean propagation will be provided. This paper presents a theoretical framework for computing cross-mode coherences and demonstrates why the coherences are not important for the evolution of mode energy.

9:15

5aUW2. Perth-Bermuda (1960) revisited global scale acoustic propagation. Kevin D. Heaney and James J. Murray (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039)

A recent paper in Geophysical Research Letter [Dushaw, Geophys. Res. Lett. **35**, L08601 (2008)] commented on the Perth-Bermuda Sound Propagation Experiment (1960): An Adiabatic Mode Interpretation JASA 95, November 1991, by Kevin D. Heaney, William S. Kuperman, and B. E. McDonald. Dushaw raises questions about the validity of the approach and conclusions of Heaney *et al.* In particular, the validity of the adiabatic mode assumption for long-range propagation in the presence of internal wave scattering and seafloor interaction is challenged. Dushaw finds no other explanation for the observed double pulse arrival in the original 1960 experiment. The approach and conclusions of Heaney *et al.* will be presented. The range of validity of the adiabatic mode approximation in global scale acoustics will be addressed. The original computations of Heaney *et al.* were for a single mode at 10 Hz due to the computational power available at the time. Broadband results (10–50 Hz) using multiple modes summed coherently will be presented.

9:30

5aUW3. Resonant forward scattering of sound in deep ocean environments. Irina Rypina (PO Dept., WHOI, Clark Bldg., Woods Hole, MA 02543), Ilya Udovydchenkov (WHOI, Woods Hole, MA, 02543), and Michael Brown (Univ. of Miami, Miami, FL, 33149)

Resonant forward scattering of sound in deep ocean environments occurs when a ray cycle (double loop) distance is equal to the wavelength of a spectral component of the sound speed perturbation field or, more generally, when these quantities are rationally related. In the presence of a sound speed perturbation field with a broad spectrum (e.g., an internal-wave-induced perturbation field), many resonances are excited and these can be expected to overlap everywhere. In spite of this complexity, there are theoretical reasons that lead to the expectation that forward scattering, in general, and mode coupling, in particular, are largely controlled by resonant scattering. Relevant theory will be briefly described and numerical simulations (ray, mode, and PE) will be presented to illustrate important concepts. [Work supported by ONR.]

9:45

5aUW4. Low mode-number scattering of transient signals in deep ocean. Ilya Udovydchenkov (AOPE Dept., WHOI, 98 Water St., Woods Hole, MA 02543, ilya@whoi.edu), Michael Brown (Univ. of Miami, Miami, FL 33149), and Timothy Duda (WHOI, Woods Hole, MA 02543)

A modal description of the propagation of transient signals in deep ocean environments is considered. The environment is assumed to consist of a range-independent background sound channel with a single sound speed minimum on which a structured range-dependent perturbation, due, for example, to internal waves, is superimposed. Mode coupling in such environments is predominantly local in mode number, leading, to an excellent approximation, to the diffusive spreading of energy in mode number with increasing range. Because mode number is non-negative, however, the simple diffusive spreading of energy in mode number requires modification for low mode numbers (corresponding to the near axial sound energy). Simulations of distributions of energy in (low) mode number and time are presented and compared to estimates based on the theoretical work of Virovlyansky *et al.* [J. Acoust. Soc. Am. **121**, 2542–2552 (2007)]. The results presented are relevant to the analysis of measurements made during the recent long-range acoustic propagation experiment. [Work supported by ONR.]

10:00—10:15 Break

10:15

5aUW5. Vertical profiling of ambient noise with Deep Sound. David Barclay, Fernando Simonet, and Michael Buckingham (Marine Physical Lab, Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, dbarclay@mpl.ucsd.edu)

Deep Sound is a free-falling high-bandwidth acoustic recording system designed to profile ambient noise from the surface to depths of 9 km. The recording platform is autonomous and descends under gravity to its preprogramed maximum depth, where a burn-wire releases weight, permitting the system to return to the surface under its own buoyancy. Two hydrophones are mounted at half meter vertical spacing allowing the noise spectrum and vertical coherence (directionality) to be obtained over four decades of frequency (10 Hz–100 kHz). The acoustic recordings are made continuously as the instrument descends and ascends along with measurements of sound speed and depth. The system's low power and large data storage capabilities allow round trips to the deepest trenches of the ocean. Alternative modes of operation include (1) synthetic aperture signal detection and (2) residence on the seabed with return to the surface at a later time. Deep Sound's design and acoustic characteristics will be described and data from initial shallow water deployments around San Diego will be reported. [Work supported by the Office of Naval Research.]

10:30

5aUW6. Deep shadow zone arrivals on the ocean bottom. Ralph Stephen (WHOI, 360 Woods Hole Rd., Woods Hole, MA 02543), Matthew Dzieciuch, Peter Worcester (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0238), Rex Andrew, Bruce Howe, James Mercer (Univ. of Washington, Seattle, WA 98105-6698), and John Colosi (Naval Postgrad. School, Monterey, CA 93943)

To gain insight into the physical mechanisms responsible for deep shadow zone arrivals in long range ocean acoustic propagation we have characterized some key features of the replica correlated signals observed on an ocean bottom seismometer (OBS) deployed on the 2004 long-range ocean acoustic propagation experiment. We compare the vertical seismometer data from an OBS at 4973-m depth beneath the deep vertical line array (DVLA) with the deepest hydrophone on the DVLA at 4270 m depth. The results of the preliminary analysis show (1) that the vertical geophone channel has more arrivals than the deep DVLA hydrophone channels, (2) that the first arrivals on the OBS geophone and DVLA deep hydrophones correspond to energy in the first deep arriving path predicted by the parabolic equation solution, (3) that the later arrivals which are much larger in amplitude are not explained by the parabolic equation method, and (4) that it is the later arrivals that contribute to the long range (up to 3200 km) shadow zone "receptions." Possible mechanisms for the later arrivals and some consequences of the observation are discussed. [This work was supported by the Office of Naval Research.]

10:45

5aUW7. Modeling the three-dimensional field of a seismic airgun array and comparison to 2003 measured data. Arslan M. Tashmukhambetov (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, atashmuk@uno.edu), Natalia A. Sidorovskaia (Univ. of Louisiana at Lafayette, Lafayette, LA), George E. Ioup, and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA 70148)

The full three-dimensional field of a seismic airgun array is modeled using an enhanced parabolic equation run on a parallel cluster computer system, which is part of the Louisiana Optical Network Initiative network cluster. Source signatures are calculated using GUNDALF and NUCLEUS softwares. The calculated results are compared to available measurements collected in Green Canyon in the northern Gulf of Mexico in 2003. Three-dimensional maps showing angular variation (both emission and azimuthal angles) and range dependence are generated, which show peak pressures, sound exposure levels, total shot energy spectra, and one-third octave band analyses. [Research supported by the International Association of Oil and Gas Producers and the International Association of Geophysical Contractors.]

11:00

5aUW8. The source characterization study 2007: A detailed three dimensional acoustic field measurement of a seismic airgun array. Arslan M. Tashmukhambetov, George E. Ioup, Juliette W. Ioup (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, atashmuk@uno.edu), Natalia A. Sidorovskaia (Univ. of Louisiana at Lafayette, Lafayette, LA), Joal J. Newcomb (NAVOCEANO, Stennis Space Ctr., MS), James M. Stephens, and Grayson H. Rayborn (Univ. of Southern Mississippi, Hattiesburg, MS)

In September 2007 the Littoral Acoustic Demonstration Center (LADC) collected acoustic and related data from three moored arrays and ship-deployed hydrophones spanning the full water column to measure the three-dimensional acoustic field of a seismic airgun array. The seismic source vessel shot a series of lines to give a detailed angle and range information concerning the field. The data were collected in the western Gulf of Mexico between the East Break and Alamos Canyon regions. Peak pressures, sound exposure levels, total shot energy spectra, and one-third octave band analyses are measures used to characterize the field. Three dimensional maps of these quantities are generated to show dependence on emission and azimuthal angles and range. Both the direct and indirect fields are characterized. Moveout analysis is done to delineate arrivals and to detect ducted and interface waves. [Research supported by the International Association of Oil and Gas Producers.]