Session 2aAAa

Architectural Acoustics: ETS-Lindgren Acoustic Test Laboratory and Factory Tour

Douglas Winker, Chair
ETS-Lindgren, 1301 Arrow Point Dr. Cedar Park, TX 78613

A tour of the new ETS-Lindgren Acoustic Research Laboratory in Cedar Park, Texas will be conducted. The schedule for the day includes a technical discussion on acoustic test facilities followed by a lunch break and factory tour. As a bonus, the tour will feature demonstrations of the material presented in the technical discussion.

Tour participants will see several state-of-the-art chambers for acoustic test services, including a hemi-anechoic chamber and two reverberation chambers, impedance tubes and supporting acoustic test equipment and software. The laboratory offers product noise emission testing and structural/architectural acoustic testing.

Product noise emission testing is commonly performed in the double-walled hemi-anechoic chamber that is designed to measure very low noise emissions from products and devices at 80 Hz and above Outside chamber dimensions are 8.5 m long $\times$ 8.5 m wide $\times$ 7 m high. This chamber is ideal for testing sound power and pressure levels as well as small fan noise. Products tested include Information Technology Equipment (ITE) such as laptop computers and associated printers, home appliances, garden equipment — essentially any noise emitting device may be tested in this chamber. Commonly referenced standards for testing in this chamber include ISO 3744, SO 3745, ISO 7779, ISO 11201, and ECMA 74. Structural/architectural acoustic testing is performed in the reverberation chambers. With transmission loss testing of wall samples, windows, doors, automobile panels and the like, design engineers can determine how much sound energy is transmitted through a product in the chambers. The source chamber measures 7.4 m long $\times$ 5.9 m wide $\times$ 4.8 m high; the receive chamber measures 7.4 m long $\times$ 9.2 m wide $\times$ 6 m high. ASTM E90, ASTM C423, ASTM E596, and ISO 3741 are the most commonly referenced standards for testing in these chambers.

To enhance chamber performance, the hemi-anechoic inner chamber sits on a 50 ton isolated concrete slab while the reverberation chambers sit on individual floating concrete slabs. The laboratory is ISO 17025 accredited under the US Department of Commerce NIST National Voluntary Laboratory Accreditation Program (NVLAP) Lab Code 100286–0. The tour will also feature a stop in ETS-Lindgren's ISO 9001 certified factory. Tour participants will see how acoustic chambers are constructed.

Bus loading will begin promptly at 8:00 am outside the main entrance of the Hyatt Regency Hotel, on Losoya Street. Tour participants will travel in a luxury air-conditioned motor coach for approximately 90 minute ride to Cedar Park (near Austin). The bus is equipped with bathroom facilities. Refreshments will be provided upon arrival at ETS-Lindgren. A traditional Texas Style BBQ lunch buffet will be served at noon. Snacks in the afternoon and a treat for the return bus ride to San Antonio will also be provided. The bus will depart Cedar Park by 2:30 pm for an arrival at the Hyatt Hotel before 5:00 pm, traffic permitting. Please note tour attendance is limited to 50 people and reservations will be confirmed in the order received until space is filled. There is no fee to attend, but you must have a prior reservation to board the bus. To make your reservation, please visit www.ets-lindgrenregistration.com/ASAtour. For more information, please contact Janet O'Neil, janet.oneil@ets-lindgren.com or phone +1.425.868.2558.

Session 2aAAb

Architectural Acoustics, ASA Committee on Standards, Noise, and Speech Communication: Classroom Acoustics: New Design Approaches—Both Successes and Failures

Kenneth W. Good, Jr., Cochair
Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603

Pamela J. Harght, Cochair
BAi, LLC, 4006 Speedway, Austin, TX 78751

Chair's Introduction—7:55

Invited Papers

8:00

2aAAb1. Predicting acoustic performance in reconfigurable classrooms. Kenneth P. Roy and Kenneth W. Good, Jr. (Bldg. Products Technol. Lab., Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604, kproy@armstrong.com)

Both laboratory component tests and field system evaluations were conducted to screen architectural designs relative to sound quality performance within a classroom, and noise intrusion between classrooms. Recognized standards such as ANSI S12.60 for Acoustics in Schools and other rating systems such as LEED for Schools and CHPS provide guidance on performance/design needs for K-12 schools. But what about postsecondary schools and especially those designed with reconfigurable architectural elements such as relocatable walls and raised floor system—can these be made to work? Initial results from this ongoing research will be discussed, including wall performance, ceiling/plenum effects, and sound system effects.

8:20

2aAAb2. Case study in classroom acoustics measurements. Kenneth Good and Kenneth Roy (Armstrong World Ind. 2500 Columbia Ave., Lancaster, PA 17603, kwgoodjr@armstrong.com)

Recent work was done related to exploring variations and interpretations of measurement techniques in a classroom environment. The primary focus of the work was related to quantifying room to room isolation; however, minimal background noise and reverberation time measurements were included. This presentation will discuss the differences in the measurements methods, results, and when each may be appropriate.

8:40

2aAAb3. Modular classroom acoustics: Where we are and where we should be going. 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, philipp.norman@gmail.com)

This paper presents a review of an 18-month study into the acoustical characteristics of modular (or relocatable) classrooms within the Omaha Public School District in Omaha, NE. The study included measurements of reverberation times, interior background noise levels (both occupied and unoccupied), and exterior facade sound insulation properties. From analysis of the gathered data, recommendations are made regarding sound insulation classification systems and background noise levels for the renovation of existing modular classrooms. These results may be used to inform the pending addendum to ANSI S12.60-2002, regarding acoustic guidelines for modular classrooms. Suggestions for future research are also discussed.

9:00

2aAAb4. Acoustics in a high school gymnasium. Stephanie Hoeman, Jon Birney, Hannah Schultheis, Shane Kanter, and Bob Coffeen (The Univ. of Kansas, 1465 Jayhawk Blvd., Lawrence, KS 66046, stephanie.hoeman@gmail.com)

A high school gymnasium constructed with unintentionally interesting acoustics was recently scheduled for a new sound reinforcement system. It was determined that the sound reinforcement system would be of little consequence until the architectural acoustics of the space were addressed. The gymnasium was constructed of entirely concave shapes which resulted in severe sound focusing. The space was extremely reverberant and exhibited many distinct, distracting reflections. Impulse responses were made in the space and computer models were made. The goal was to find a solution to reduce unwanted reflections and shorten the reverberation time. Since this project was for a public school with limited funds, utmost importance was placed on a solution that would be economical and uncomplicated to install. Acoustical analysis and auralizations of the space in its current condition and the projected performance of recommendations were compared to find an effective, cost-efficient solution the school board could implement to improve the architectural acoustics before the sound reinforcement system was added.

9:20

2aAAb5. Current trends in K-12 classroom design. Jessica Molter and Jonathan Hodge (Pfluger Assoc., LP 209 E Riverside Dr., Austin, TX 78704)

Presentation of current trends in K-12 Classroom Design with an emphasis on items that contribute to acoustic performance of the space.

9:40

2aAAb6. Classroom reinvented, a quantum leap in classroom design. Kenneth Good (Acoust. Privacy Enterprises, LLC, Mount Joy, PA 17552, kwgoodjr@acousticprivacy.com)

How do you design an environment to maximize your impact on a child's development when you are limited to just an hour or two a week? Traditional academic education or biblical instruction, the environment is critical to success in reaching children and youth. This case study will explore the acoustical performance and design strategies that went into LCBC's new and nontraditional building for children and students.

Contributed Paper

10:00

2aAAb7. Optimizing the signal-to-noise ratio in speech rooms using passsive acoustics. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

Adults with normal hearing require a roughly 0-dB signal-to-noise ratio for good speech intelligibility in classrooms and lecture halls. However, significantly higher values may be needed to compensate for neurological immaturity, sensorineural and conductive hearing losses, language proficiency, and excessive reverberation. ANSI 12.60 addresses ways to lower the noise interference due to background levels and reverberation time. However, it is also possible to increase the signal, by reflecting or diffusing early reflection . While speech power is delivered in the vowels which are predominately in the 250–500-Hz frequency range, speech intelligibility is delivered in the consonants, which occur in the 2–4-kHz frequency range. Therefore, effective core learning designs can incorporate scattering surfaces, rather than surfaces that absorb in the 2–4-kHz region, on the front wall, lower side walls, and central ceiling areas, to increase the speech signal. The decay time can be controlled with broadband absorption on the perimeter of the ceiling and upper wall surfaces. A computer model analysis of various speech environments will be presented.

TUESDAY MORNING, 27 OCTOBER 2009 REGENCY EAST 2, 10:20 A.M. TO 12:00 NOON

Session 2aAAc

Architectural Acoustics, Noise, ASA Committee on Standards, and Speech Communication: Classroom Acoustics: An Update

David Lubman, Cochair
DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683-4514

Kenneth W. Good, Jr., Cochair
Armstrong World Industries Inc., 2500 Columbia Ave., Lancaster, PA 17603

Invited Papers

10:20

2aAAc1. Working towards an enforceable standard for classroom acoustics. Lois Thibault (US Access Board, 1331 F St. NW, 1000, Washington, DC 20004)

The US Access Board has proposed to reference ANSI/ASA S12.60-2002 (R-2009) in the 2012 International Building Code (IBC), which will provide for local enforcement. If this proposal is adopted, the Board will then update its 2004 ADA-ABA Accessibility Guidelines to harmonize with the IBC. This paper will review the history of classroom acoustics initiatives, including the standard, and update attendees on the current process. Other classroom acoustics activity will be highlighted, particularly sustainability and green design proposals, and the 2010 re-authorization activities under the Individuals with Disabilities Education Act.

10:40

2aAAc2. The acoustic treatment in classroom refurbishment: A double blind experimental study examining the acoustic and auditory environment of the cellular classroom. David C. Canning (Div. of Psych. and Lang. Studies, Univ. College London, 26 Bedford Way, London, WC1H 0AP, United Kingdom and Hear2Learn, United Kingdom, canningd@gmail.com)

This paper will present data from the Essex Study of mainstream schools designed for inclusion of children with special hearing difficulties. It will present data from the experimental study which was set up to guide the specification of classroom acoustic performance standards. The UK Special Educational Needs Legislation requires education authorities to meet the special educational needs of all children. The major need considered in the provision of children with special hearing requirements, including deaf and hard of hearing children, is their ability to function when there is competing sound. Control of competing sound is in part the job of the teacher, as the dominant source of sound is created by the occupants, but the acoustician has a significant role to play. The acoustic performance of a classroom has considerable impact on every aspect of teaching and learning and ultimately on whether a child's special educational needs can be met within an inclusive educational setting. The findings of the study have implications for the acoustical performance of all schools, given that every school might reasonably be expected to provide for children with special hearing requirements. [The support of Sweyne Park School, Essex County Council, The National Deaf Children's Society, and the Federation of Property Services is acknowledged.]

11:00

2aAAc3. Auralizing adult-child listening differences. Peggy B. Nelson (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, peggynelson@umn.edu), Jonah Sacks, and Jennifer Hinckley (Acentech, 33 Moulton St., Cambridge, MA 02138)

Substantial data from previous research show that children and adults require different acoustical conditions for good understanding. For example, adults can understand the majority of speech when the audibility of the speech is reduced to 40% or 20%, but young children need 80% or 60% audibility for the same level of understanding. Also, while adults need 4 to 6 bands of vocoded speech to reach good performance levels, children need 8–12 bands. While adults experience a release from masking when signals and background noise arise from different angles, children do not gain the same benefit, and in fact may experience masking from background noises coming from any direction. J.H. and J.S. from Acentech prepared auralizations that demonstrate these differences between children and adults. Those auralizations will be presented and discussed as possible educational tools.

11:20

2aAAc4. Integrating acoustical issues in the design of high-performance schools. Gary W. Siebein (School of Architecture, Univ. of Florida, P.O. Box 115702, Gainesville, FL 32611), Chris Jones, Robert M. Lilkendey, Hyun Paek, and Reece Skelton (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607)

ANSI 12.60 presents acoustical performance criteria for interior finishes to control reverberation inside classrooms, provide sound isolating wall and floor/ceiling assemblies between rooms, and limit noise from building equipment to allow perception and understanding of human voices within educational occupancies by young listeners with normal hearing and some degrees of hearing impairments. In practice, many schools are built that develop in an uneasy way trying to meet or sometimes to avoid meeting ANSI criteria because of perceived difficulties and expense in achieving the required results. A method to integrate acoustical design principles with the basic architectural design scheme of a school that can be implemented early in the design process was developed so acoustical performance can be simply implemented in school projects. This paper presents a core set of acoustical planning principles that can be implemented early in the design process so that ANSI criteria for room finishes, sound isolation, and building equipment noise can be met as the design progresses.

11:40

2aAAc5. High-performance acoustic ceilings make quiet classrooms quieter. David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683, dlubman@dlacoustics.com) and Louis C. Sutherland (Consultant in Acoust., 27803 Longhill Dr., Rancho Palos Verdes, CA 90275)

Highly beneficial noise level reductions of 5–10 dB are reported in occupied classrooms equipped with highly sound absorbing ceilings. The incremental cost for such ceilings is nominal. These "Lombard effect" benefits apply to small classrooms with very low reverberation times (0.5 s or less) which is less than the 0.6-s maximum specified in ANSI S12.60-2002. The amount of noise reduction depends on teaching/learning style. In the typical noise reduction lecture classrooms (single talker), the typical noise reduction benefit is 5 dB in lecture classrooms (single talker) and group learning classrooms (multiple talkers). Unfortunately, these benefits require unoccupied background noise levels (BNLs) of about 35 dBA. They are not expected in typical noisy American classrooms (BNL ~>45 dBA). This underscores the importance of compliance with ANSI's 35-dBA BNL limit rather than the lenient and unsupported 45-dBA limit permitted in recent LEED and California CHPS guidelines. Details are reported in the outstanding study "Acoustic Ergonomics of School" by Oberdorster and Tiesler [University of Bremen, Germany (2006)]. Similar benefits were reported earlier in a case study by Sutherland ["The role of soundscape in children's learning," J. Acoust. Soc. Am. **112**, 2412–2413 (2002)].

Session 2aAOa

Acoustical Oceanography: Acoustic Inversion

Juan I. Arvelo, Chair
Johns Hopkins Univ., Applied Physics Lab., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099

Contributed Papers

8:00

2aAOa1. Seabed sound speed and attenuation from broadband acoustic measurements in the Shallow Water 2006 experiment. Lin Wan, Ji-Xun Zhou, Peter H. Rogers (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, lin.wan@gatech.edu), and David P. Knobles (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713)

In the Shallow Water 2006 experiment, a set of broadband combustive sound source (CSS) signals was measured by two L-shaped arrays separated by 20 km on the New Jersey continental shelf with a water depth of around 72 m. The measured CSS data, which exhibit modal dispersion, are utilized to infer values for the seabed sound speed and attenuation. The seabed sound speed is estimated by matching the theoretical and measured modal dispersion curves, which are extracted using an adaptive time-frequency analysis technique. The frequency dependence of the seabed attenuation is inferred by minimizing the difference between the theoretical and measured modal amplitude ratios in the 50–400-Hz band. The resultant seabed attenuation is similar to the low-frequency seabed attenuation data obtained at 20 locations in different coastal zones around the world by Zhou *et al.* [J. Acoust. Soc. Am. **125**, 2847–2866 (2009)]. The depth dependence of the seabed attenuation will be discussed. [Work supported by the Office of Naval Research.]

8:15

2aAOa2. Time domain geoacoustic inversion using back-propagation on an L-shaped array. Cheolsoo Park, Peter Gerstoft, Woojae Seong, and William S. Hodgkiss (Marine Phys. Lab., Univ. of California, San Diego, La Jolla, CA 92093-0238)

This paper presents inversion of Shallow Water 06 experimental data measured on an L-shaped array (SWAMI 32). The array was deployed in 70-m water depth and consists of an equally spaced 10-element vertical line array (VLA) and a 256-m-long bottom moored 20-element horizontal line array (HLA). A mid-frequency (1100–2900-Hz) chirp source was towed at 35-m depth along a circular track around the VLA with radius 190 m. Time domain inversions using VLA, HLA, and both arrays were carried out and results compared. For the inversions, a multi-step optimization scheme is applied to the data using very fast simulated reannealing. The objective function is defined by the power of the back-propagated signal from the array to the source. At each step, water column sound speed profile, experimental geometry, and geoacoustic parameters are inverted successively. Accurate HLA positions were essential for the HLA and the HLA+VLA inversions. Finally, the inversion results were compared with other results near the site. [Work supported by ONR.]

8:30

2aAOa3. The estimation of geoacoustic parameters via low frequencies (50–100 Hz) for selected Shallow Water 06 test data. A. Tolstoy (ATolstoy Sci., Inc., 1538 Hampton Hill Circle, McLean, VA 22101, atolstoy@ieee.org) and Yong-Min Jiang (Univ. of Victoria, Victoria, BC, V8W 3P6, Canada)

This work will demonstrate the geoacoustic inversion "success" on data of using only one or two low frequencies, multiple ranges, and multiple realizations for geoacoustic inversion of actual SW06 data. The data used are the same as those processed by Jiang and Chapman and involves three ranges (1, 3, and 5 km) and multi-tonal continuous wave data collected on a 16 phone vertical array. Multiple realizations of the data were used where each reduced the non-uniqueness a bit. The multiple ranges and frequencies also reduced the non-uniqueness of the suggested solutions. However, there still remains a significant number (hundreds) of possible "solutions," (values of ctop, cbot, hsed, and chsp) for which MFP$<$0.9 (including those suggested by Jiang and Chapman). The use of higher frequencies requires refinement of more parameters (such as the ocean sound-speed profile, source depth and range, water depth, and phone locations) but would not necessarily improve estimates of such bottom parameters as chsp and hsed. Thus, non-uniqueness of bottom parameters is an issue which may well exist for all inversion approaches.

8:45

2aAOa4. Ocean tomography with acoustic daylight: A case study. Oleg A. Godin, Nikolay A. Zabotin (NOAA/Earth System Res. Lab., CIRES, Univ. of Colorado, Boulder, CO 80305, oleg.godin@noaa.gov), and Valery V. Goncharov (P. P. Shirshov Oceanology Inst., Russian Acad. of Sci., Moscow 117997, Russia)

Ambient and shipping noise in the ocean provides acoustic illumination, which can be used, akin to daylight in the atmosphere, to visualize objects and characterize the environment [Buckingham *et al.*, Nature (London) **356**, 327–329 (1992)]. It has been shown theoretically [O. A. Godin, Phys. Rev. Lett. **97**, 054301 (2006)] that, under rather general conditions, deterministic travel times between any two points in an inhomogeneous, moving or motionless, time-independent medium can be retrieved from the cross-correlation function of non-diffused acoustic noise recorded at the two points, without a detailed knowledge of the noise field's sources or properties. Using the data obtained during the 1998–1999 Billboard Array Experiment [Worcester *et al.*, J. Acoust. Soc. Am. **117**, 1499–1510 (2005)], this paper demonstrates the feasibility of a tomographic reconstruction of the sound speed field from cross-correlation of acoustic noise recorded on a pair of vertical line arrays (VLAs) in deep water. Limitations of the noise data inversion associated with the ocean temporal variability, the VLA horizontal separation, recording bandwidth, and the noise directionality are analyzed. Prospects of long-range water column tomography with acoustic daylight are discussed. [Work supported by ONR.]

9:00

2aAOa5. Two-point coherence of acoustic noise recorded by the North Pacific Acoustic Laboratory billboard array. Oleg A. Godin and Nikolay A. Zabotin (NOAA/Earth System Res. Lab., CIRES, Univ. of Colorado, Boulder, CO 80305, oleg.godin@noaa.gov)

Acoustic noise in the ocean contains extensive information about the noise sources and the propagation environment. In particular, two-point correlation function of noise (NCCF) is known to have peaks which correspond to acoustic travel times between the two points provided the noise field is sufficiently diffuse. In this paper, measurements of acoustic noise performed during the 1998–1999 Billboard Array Experiment [Baggeroer *et al.*, J. Acoust. Soc. Am. **117**, 1643–1665 (2005)] are re-examined with the goal of extracting environmental information. NCCF is evaluated by averaging time series of noise recorded on various vertical line arrays that comprise the Billboard Array. For any two hydrophones, NCCF is found to have a number of robust peaks. Possible generation and propagation mechanism responsible for various features of the NCCF are discussed. Statistical distributions

of noise are utilized to differentiate between the NCCF peaks of different origins and to identify those peaks, which can be used to retrieve information about the sound speed without detailed knowledge of the noise sources. [Work supported by ONR.]

9:15

2aAOa6. Measurements of sound speed in bubbly liquids under high-pressure conditions. Chad A. Greene and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78712-0292)

Methane hydrates occur naturally on the ocean bottom and in the upper layers of sediment on continental shelves. Seismic surveying could be used to locate methane hydrates; however, their low-frequency acoustic properties are not well-known. In addition, these properties can vary dramatically depending on whether the methane is in a gas or solid phase. As a step toward better understanding the three-phase case of gassy sediments in water, the two-phase case of methane gas bubbles in water was investigated. Wood's equation is often used to model sound propagation in bubbly liquids and has been widely verified by experiments at atmospheric pressure. However, there is little information in the literature verifying the validity of Wood's equation at high pressures. Low-frequency (0.5–10-kHz), resonator-based sound speed measurements were obtained for air bubbles in water and methane bubbles in water under pressures ranging from 1 to 10 atm at room temperature. The results are presented and compared to the predictions of Wood's equation. [Work sponsored by ONR.]

9:30

2aAOa7. Estimating bubble density from attenuation measurements through an underwater explosion. Fred D. Holt, IV and R. Lee Culver (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804)

Underwater explosions have been studied intensively in the United States since 1941 [e.g., Cole (1945)]. Research to date primarily focuses on the initial shock and subsequent pressure waves caused by the oscillations of a "gas globe" that is the result of a charge detonation. These phenomena have relatively short timescales (typically less than 2 s). However, as the gas globe rises in the water column and breaks the surface, it leaves behind a residual bubble cloud which has been markedly less studied. A recent experiment measured the spatial and time-dependent acoustic response of the bubble cloud resulting from a charge detonated at 50-ft depth. A directional projector was used to propagate a linear FM (5–65-kHz) pulse through the bubble cloud to an array of hydrophones placed on the opposite side of the charge in order to measure attenuation. This talk will focus on the methods used to estimate bubble density size spectra from the attenuation measurements, those of Commander and McDonald (1991), Caruthers (1999), and Czerski (2009). [Work sponsored by the Office of Naval Research, Code 333.]

9:45

2aAOa8. Sediment shear as a perturbation in geoacoustic inversions and an explanation of the anomalous frequency dependence of the attenuation. Allan D. Pierce and William M. Carey (Dept. of Mech. Engr., Boston Univ., Boston, MA 02215, adp@bu.edu)

The depth dependent shear wave speed in marine sediments is much less than both the compressional and water column sound speeds. The neglect of shear in geoacoustic inversions is usually justifiable, but at frequencies less than 300 Hz, the loss of acoustic energy from the water column because of nonreturning radiation of shear waves into the bottom dominates the loss due to the intrinsic attenuation of the sediment. To account for this in a simple manner, a perturbation theory that takes advantage of the small shear speed has been devised. The canonical problem addressed takes the disturbance as having a horizontal trace velocity ω/k in the x direction, this being somewhat less than the compressional speed in the bottom but substantially larger than the depth-dependent shear wave speed. To lowest order the stress components σ_{zz} and σ_{xx} satisfy the same reduced Helmholtz–Bergmann equation as if the bottom were an inhomogeneous fluid. The shear stress σ_{xz} satisfies an approximately uncoupled equation governing a shear wave propagating vertically downward. Coupling occurs at the interface, and the power carried off by the shear wave can be approximately determined. Results succinctly and quantitatively explain why geoacoustic inversions yield attenuation frequency exponents less than 2.

TUESDAY MORNING, 27 OCTOBER 2009 RIO GRANDE EAST, 10:15 A.M. TO 12:00 NOON

Session 2aAOb

Acoustical Oceanography: Acoustics and Ocean Acidity I

Timothy F. Duda, Cochair
Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543-1053

Peter F. Worcester, Cochair
Univ. of California San Diego, Scripps Inst. of Oceanography, 9500 Gilman Dr., La Jolla, CA 92093

Chair's Introduction—10:15

Invited Paper

10:20

2aAOb1. A brief history of the discovery of the low frequency sound absorption mechanism in seawater and its *p*H dependence. David G. Browning (139 Old North Rd., Kingston, RI 02881), Vernon P. Simmons (335 Burgundy Rd., Healdsburg, CA 95448), and William H. Thorp (2 Brook St., Noank, CT 06340)

In the 1960s tests with a new surface ship sonar indicated that low-frequency (less than 10 000 Hz) propagation loss was greater than predicted by the existing $MgSO_4$ absorption model. Propagation measurements in the SOFAR channel showed that the anomalous absorption was up to ten times greater than predicted and could be fitted by a 1-kHz relaxation. This was not without controversy at the time, but interest in this fundamental parameter spurred a major measurement effort by the U.S. Navy Underwater Sound Laboratory and other laboratories in the US, Canada, New Zealand, Europe, and Australia, resulting in at-sea measurements from Hudson Bay to

the Tasman Sea. Yeager and Fisher conducted laboratory t-jump measurements that identified a boron based low-frequency relaxation mechanism in seawater. Simmons and Fisher followed with resonant sphere measurements to quantify the resulting acoustic absorption. A compilation of the at-sea results showed, somewhat surprisingly, a correlation with *p*H: the lower the *p*H, the lower the absorption. Mellen quantified the *p*H dependence by extensive laboratory resonator measurements. These results were confirmed by measurements conducted in China. The possible application of acoustically monitoring ocean *p*H was first suggested by Browning and Mellen in 1990.

Acoustical Oceanography Mini Tutorial

10:45

2aAOb2. Rapidly changing ocean *p*H and the increasing transparency of the ocean to sound. Peter G. Brewer (MBARI, 7700 Sandholdt Rd., Moss Landing, CA 95039, brpe@mbari.org)

The intrinsic sound absorption coefficient (á, dB/km) of seawater is *p*H dependent with significant effects at 10 kHz and below. Ocean *p*H is declining from fossil fuel CO_2 invasion, from excess nutrient input, and from climate change reducing ocean ventilation. These effects produce reductions in ocean borate and carbonate species such that an ~18% decrease in á in the upper ocean has occurred today. Reasonable projections based on IPCC scenarios predict changes of 40% or more by mid-century [Hester *et al.*, Geophys. Res. Lett. **35**, L19601 (2008)]; larger changes in the sound channel are very likely. The projected increased transparency of the ocean to sound has attracted strong international environmental attention, and also provides an opportunity for acoustic detection and monitoring of such changes over large ocean regions. How strong is the basis for such assertions, can they be tested, and how well can environmental effects be predicted? This tutorial shows that the rate of change in *p*H in the sound channel has been underestimated, and how the unusual 1970s era reliance on the Soviet Gorshkov atlas as a *p*H data resource came about. This now presents a challenge for convergence of modern acoustic, environmental, and global change needs.

Invited Paper

11:35

2aAOb3. Long-term trends in ambient noise levels. George V. Frisk (Dept. of Ocean Eng., Florida Atlantic Univ., 101 N. Beach Rd., Dania Beach, FL 33004, gfrisk@seatech.fau.edu)

This paper addresses the subject of long-term trends in ambient noise levels, a topic of great interest to both the scientific community and the general public. This attention stems primarily from concerns over the effects of apparently increasing sound levels on marine mammals. A growing, though limited, body of literature suggests that low-frequency noise levels increased approximately 15 dB during the period 1950–2000, an amount that corresponds to about 3 dB/decade. One hypothesis states that this increase is predominantly anthropogenic in nature and can be attributed to increased commercial shipping activity, which, in turn, can be linked to global economic growth. As a result, a direct correlation may be drawn between ambient noise levels and the behavior of the global economy. This special session addresses an additional consequence of global economic activity, namely, increased ocean acidification leading to decreased absorption and therefore to increases in ambient noise levels associated with distant shipping. This paper also suggests topics of considerable interest for future research including (1) the relative contributions of commercial shipping activity versus ocean acidification to ambient noise levels and (2) the effect of the current economic downturn on noise levels. [Work supported by ONR.]

TUESDAY MORNING, 27 OCTOBER 2009 — PECAN, 8:00 A.M. TO 12:00 NOON

Session 2aEA

Engineering Acoustics and Signal Processing in Acoustics: Acoustic Measurement and Models for Sensors and Arrays

Dehua Huang, Chair
Naval Undersea Research Center, 1176 Howell St., Newport, RI 02841-1708

Chair's Introduction—8:00

Invited Papers

8:05

2aEA1. Multidomain modeling of a variable reluctance transducer. Stephen C. Thompson (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804)

The variable reluctance magnetic transducer consists simply of a magnetic air gap and a magnetic return path that are supplied with static and dynamic sources of magnetic flux. The dynamic flux is generated by a coil current. The static flux may come either from a permanent magnet or a coil. The attractive magnetic force across the gap is modulated by the dynamic flux to provide the mechanical excitation for the device. The variable reluctance device is fundamentally nonlinear for at least three reasons: (1) the mechanical force across the gap is a quadratic function of the total magnetic flux, (2) the changing gap dimension changes the reluctance in the magnetic circuit so that the flux does not change linearly with coil current, and (3) saturation of the magnetic circuit may be an important aspect

of the design that must be included in the modeling. In practical cases, the dynamic variation in gap dimension is large enough that a linearized approximation is insufficient to predict the performance. An approximate model using a set of nonlinear differential and algebraic equations will be discussed that can predict the stability and performance of variable reluctance transducers.

8:25

2aEA2. Lead magnesium niobate-lead titanate solution based giant-piezoelectric crystals for next generation of acoustic transduction devices. Pengdi Han (H. C. Mater. Corp., 479 Quadrangle Dr., Ste.-E, Bolingbrook, IL 60440)

The PMN-PT (binary solid solution of lead magnesium niobate and lead titanate) based piezoelectric crystals have been commercialized now. The giant-piezoelectric crystal can be broadly utilized for the next generation of acoustic transduction devices. In this paper, the major concepts and current status of the crystals and products will be presented and discussed in terms of applications. Following a brief review on the history of development of crystal growth and commercialization in the past decade, the detailed physical properties of the PMN-PT and newly developed PIN-PMN-PT crystals (high depoling temperature and high driving field) are presented and discussed in terms of applications in ultrasound transducers. At present (001)-seeded single crystals of 3 and 4 in. diameters are commercially available. In order to sufficiently utilize the advantages of the crystals, discussions focus on the following: (1) The differences between single crystal and PZT ceramics. (2) The concept of so called "domain-engineering" (artificially domaining) in context of elasto-piezo-dielectric matrices. (3) The relationships between the property and ferroelectric domain structure. (4) How to select the crystal products. In summary, the achievement of the development and fabrication of the giant-piezoelectric crystals has led to a new era that there are new opportunities readily to be explored for the next generation of acoustic transduction devices.

8:45

2aEA3. Analysis models for the Underwater Sound Reference Division low-frequency acoustic calibration systems. Dehua Huang and Anthony Paolero (NUWC, Newport, RI 02841)

The state of an art for calibrating acoustic transducers at very low frequencies is by way of a confined and well understood environment. The Underwater Sound Reference Division (USRD) has three such calibration systems, called systems K, J, and L, respectively. Each system has a cylindrical tube, of certain length and diameter, that determine a cutoff calibration frequency. System K operates in a standing wave mode condition. Systems J and L both operate in traveling wave mode conditions, where plane waves propagate from one end of the tube to the other. Optimally locating a calibrated transducer and an unknown within the tube provides the proper configuration for calibration. This paper demonstrates the simulation tools to predict performance of aforementioned low-frequency calibration systems. The mathematical model, the GUI coding, simulation, predicted, and test results will be presented.

9:05

2aEA4. Electroacoustic transducer and array modeling tools. Ender Kuntsal (Int. Transducer Corp. (ITC), 869 Ward Dr., Santa Barbara, CA 93111, ekuntsal@itc-transducers.com)

The availability and the accuracy of transducer and array design tools are becoming even more important with the changing economical climate. The engineers would like to evaluate their designs more quickly in place of the prototyping, which requires additional manufacturing and testing time and expense. The software models must be user friendly, provide results that can be relied on, and also be affordable. The strength of a transducer and array design engineering team is still extensively based on experience and background, but these tools make the process more efficient. There are many transducer and design software programs used in industry and academia. Some are based on analytical solutions and some use more complicated numerical methods such as finite element models, which have the capability of modeling piezoelectric materials and acoustical parameters. Among these are ATILA, PAFEC, MAVART, NASTRAN, PZFlex, and other powerful programs such as CHIEF and TRN. Some of these programs can be used to design both resonators and arrays while taking baffling and/or acoustic interaction effects into account. In this presentation, a summary of commonly available software packages and their features will be described.

9:25

2aEA5. High-power single crystal based projectors. Richard J. Meyers, Jr., Douglas C. Markley, Charles W. Allen, and Nevin P. Sherlock (The Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804)

Significant progress has been made at integrating single crystalline relaxor ferroelectrics into many types of SONAR transducers. For high-power projectors, PMN-28PT has offered efficient, broad band-width capability. For higher duty cycles, thermal limitations are reached resulting in power and duty cycle tradeoffs. Electrical bias is also required for crystal implementation when high-power operation is needed. Continuing research in compositional tailoring has resulted in several new modified relaxor crystal systems with improved temperature stability, lower loss, and higher coercive fields. PIN-PMN-PT ternary crystals are of particular interest. This ternary system offers 25–40 °C improvement in working temperature, reduced temperature sensitivity, and approximately two times increase in coercive field without sacrifice in electromechanical coupling or piezoelectric coefficients. To demonstrate the impact of this newer ternary crystal composition on high-power transduction, planar tonpilz arrays were fabricated and tested. This presentation will highlight important material properties and compare the results of high-power measurements of single transducer elements and arrays made from PMN-28PT and PIN modified ternary crystals. Acoustic tests were conducted as a function of ambient temperature, increased drive level, and increased duty cycle operation. [Work supported by ONR.]

9:45

2aEA6. Broadband characteristics of piezoelectric transducer bonded to a thick plate resonator. Iwaki Akiyama, Natsuki Yoshizumi (Dept. of Electric and Electron. Eng., Shonan Inst. of Tech., 1-1-25 Tsujido-nishikaigan, Fujisawa 251-8511, Japan, akiyama@elec.shonan-it.ac.jp), Shigemi Saito (Tokai Univ., Orito, Shimizu-ku, Shizuoka 424-8610, Japan), Katsumi Ohira, Osamu Takahashi (Japan Probe Co., Ltd., Minami-ku, Yokohama 232-0033, Japan), and Kentaro Nakamura (Tokyo Inst. of Tech., Yokohama 226-8503, Japan)

The piezoelectric transducer bonded to a thick plate resonator wholly vibrates as a single transducer. Since the distribution of stress in the thickness direction is asymmetric, the transducer resonates at even order harmonic frequencies as well as odd order ones. The back of the resonator is bonded to a quarter-wavelength matching layer to transmit backward the fundamental frequency wave through the layer. The front of piezoelectric transducer is bonded to another matching layer to transmit forward the ultrasonic wave of higher order frequencies. If both the piezoelectric transducer and the resonator are made of low Q materials, the admittance curve including troughs among resonance peaks is flattened and shows broadband characteristics. Such broadband characteristic transducers made of one to three composite piezoelectric materials were experimentally studied. The thickness of the piezoelectric transducer and the resonator were designed for resonance frequencies of 7 and 1 MHz, respectively. The back of matching layer was bonded to the absorbing material not to reflect forward the ultrasonic waves of 1 MHz. As a result, the ultrasound of center frequency of 5 MHz and fractional bandwidth of 100% were transmitted from the transducer driven by an impulsive signal.

10:00—10:15 Break

10:15

2aEA7. The design of quarter-wavelength impedance matching layers for cylindrical transducers. Douglas R. Heyden, Preston S. Wilson (Appl. Res. Labs. and Mech. Eng. Dept., Univ. of Texas at Austin, Austin, TX 78713-8029), and Richard H. Crawford (The Univ. of Texas at Austin, Austin, TX 78712-0292)

Impedance matching layers are commonly used in piezoelectric underwater acoustic projectors. The layer maximizes transmitted power from the ceramic into the water and also increases the bandwidth of the projector. For the design of cylindrical transducers, it is common in practice to utilize the familiar plane wave formulation of the quarter-wavelength impedance matching layer as a starting point for the design of the cylindrical layer. Material properties and thickness are then modified by trial and error, heuristic, or empirical methods to optimize the design. This practice is undertaken because, apparently, the quarter-wavelength impedance matching layer formulation is not readily available in the acoustics literature for the cylindrical coordinate system. To address this deficiency, the reflection and transmission coefficients for the cylindrical three-medium problem were derived. No general zero-reflection, perfect transmission condition was found, but the equations can be used to find the material properties and layer thickness required to maximize transmission at a given frequency. The results of the derivation are shown and used in the design of a layered cylindrical piezoelectric transducer.

10:30

2aEA8. Time reversal focusing for pipeline structural health monitoring. Joel Harley, Nicholas O'Donoughue, José M.F. Moura (Elec. and Comput. Eng., Carnegie Mellon Univ., Pittsburgh, PA 15213), and Yuanwei Jin (Univ. of Maryland, Eastern Shore, Princess Anne, MD 21853)

Guided wave technologies have become popular tools for nondestructive testing due to their potential to travel long distances. Unfortunately, analyzing data from guided waves is often difficult because of the numerous multimodal and dispersive effects that distort signals in solid media. Time reversal has been shown to be robust against these unwanted and adverse effects. Time reversal techniques are commonly used to focus ultrasonic waves across a medium and have been used to perform nondestructive testing using pulse-echo techniques. This paper investigates the use of time reversal processing techniques to compensate for multimodal and dispersive effects in a low-power structural health monitoring system for pipelines. Time reversal methods are demonstrated as a pitch-catch operation between two transducer arrays to illuminate changes caused by damage on a pipe. It is then shown and discussed how differences in the location and severity of damage affect the signals recorded at the receiving transducer array and how these results can be interpreted to measure those changes. The results are demonstrated experimentally and then compared with equivalent finite element simulations. [National Energy Technology Laboratory (NETL) is the funding source for this effort with Cost Share being provided by Carnegie Mellon University (CMU). CMU is funded under a Subcontract Agreement with Concurrent Technologies Corporation. N.O. is supported by a National Defense Science and Engineering Graduate Fellowship.]

10:45

2aEA9. A radial propagator for computing axisymmetric pressure fields using the angular spectrum method. Edward H. Pees (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841)

The notion of a propagator is central to the angular spectrum of plane wave formulation of diffraction theory, which expresses the pressure field diffracted by a two-dimensional aperture as a superposition of a continuum of plane waves. In the conventional form, an exponential term, known as a propagator, is multiplied by the wavenumber spectrum obtained from a two-dimensional spatial Fourier transform of the aperture boundary condition, to obtain the wavenumber spectrum in a plane parallel to the boundary, offset by some distance specified in the propagator. By repeated use of this propagator and Fourier inversion, one can completely reconstruct the homogeneous part of the pressure field beyond the aperture boundary. In this presentation, we draw upon earlier work relating the boundary condition to the axial pressure and show that when the aperture is axially symmetric, an alternative type of propagator can be derived that propagates an axial wavenumber spectrum away from the axis of the aperture. Use of this radial propagator can be computationally advantageous since it allows for field reconstruction using one-dimensional Fourier transforms instead of Hankel transforms or two-dimensional Fourier transforms.

11:00

2aEA10. A planar acoustic array for voice collection. David J. Gonski, Duong Tran-Luu, and Stephen Tenney (Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, dgonski@arl.army.mil)

A planar 7×7 acoustic array of microphones spaced 1 in. apart has been developed and tested. This array is intended for the collection of human speech. A simple analog electronic summation of the 49 microphone signals is carried out to form a preferential collection area in front of the array. We have conducted an analysis of the directional performance of the array to model its directivity. Two configurations of the array were tested. An array was populated with omnidirectional hearing aide microphones, and a second array was configured with cardioid microphones. Performance of each of these arrays was measured in an anechoic chamber and compared with the theoretical performance. To improve the front to back rejection a sound-absorbing pad was placed on the backside of the array. Two versions of the pad were characterized in the anechoic chamber, both were found to be effective. The array was packaged into a plastic box with open cell foam on the front for wind noise suppression and the sound-absorbing pad on the back for improved front to back rejection. Detailed acoustic and electronic design characteristics are presented.

11:15

2aEA11. Microphone array techniques using cross-correlations. Matthew B. Rhudy (Univ. of Pittsburgh, 560 Benedum Hall, Pittsburgh, PA 15261, mbr5002@gmail.com) and Brian A. Bucci (Dept. of Mech. Eng., Univ. of Pittsburgh, Pittsburgh, PA 15261)

Civilian noise complaints and damage claims have created a need to establish a detailed record of impulse noise generated at military training facilities. Wind noise is causing false positive impulse detections in the cur-

rent noise monitoring systems. Multiple channel data methods were investigated in order to distinguish the characteristics of noise events. A microphone array was used to collect four simultaneous channels of military impulse and wind noise data. Cross-correlation functions were then used to characterize the input waveforms. Three different analyses of microphone array data were developed. A new value, the min peak correlation coefficient, is defined from the minimum value of peaks of the cross-correlation coefficient functions among the different channels. This value is a measure of the likelihood that a given waveform originated from a correlated noise source. The angle of incidence of the noise source is calculated using a sound source localization technique based on the geometry of the array. A weighted averaging method was also developed to synthesize multiple channels of data into one single channel. This method preserves the correlated part of the overall signal, while reducing the effects of uncorrelated noise, such as wind.

11:30

2aEA12. On the radiation and wave propagation of sound within horns.
Daniel Tengelsen (Dept. of Phys., Brig. Young Univ., N-283 ESC, Provo, UT 84602, danieltengelsen@gmail.com), Vianey Villamizar, Brian E. Anderson, and Timothy W. Leishman (Brigham Young Univ., Provo, UT 84602)

The horns used in loudspeaker systems are well known for their ability to increase radiation efficiency and control directivity. Because of the horn's ubiquitous nature, significant research efforts have been undertaken to explore both its directional and frequency response characteristics. The following research reported in this presentation incorporates several numerical methods (including finite difference and boundary element analysis) to better understand the nature of wave propagation through horns and the effects that geometrical and configurational changes have in better controlling directivity and improving frequency response.

11:45

2aEA13. On the acoustic impedance of a sealed loudspeaker enclosure.
Timothy W. Leishman and Xi Chen (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602)

The acoustic impedance of a sealed loudspeaker enclosure is often oversimplified in loudspeaker models, which typically include only rough acoustical characteristics of the enclosed air volume and absorptive fill materials when no driver is present. The acoustical effects produced by sound transmission through and around the rear driver elements (e.g., the frame, magnet, voice-coil former, spider, pole-piece vent, etc.) may also be important for certain loudspeakers and applications. This presentation explores these effects through a discussion of enhanced modeling possibilities and the introduction of measurement techniques that may be used to assess the acoustic impedance of a loudspeaker enclosure when the driver is in place. The impedance more closely represents that actually seen by the rear portion of the driver diaphragm.

TUESDAY MORNING, 27 OCTOBER 2009 REGENCY EAST 1, 10:00 A.M. TO 12:00 NOON

Session 2aED

Education in Acoustics: Hands-on Experiments for High School Students

Uwe J. Hansen, Chair
Indiana Univ., Dept. of Chemistry and Physics, Terre Haute, IN 47809

Approximately 20 acoustics demonstrations will be set up, ranging in complexity from simple resonance on a string to ultrasonic levitation. Around 40 local high school students will perform these experiments with help from Acoustical Society of America (ASA) scientists and student members of ASA. Regular ASA conference participants are welcome as long as they do not interfere with student experimentation.

TUESDAY MORNING, 27 OCTOBER 2009 LIVE OAK, 9:00 TO 10:15 A.M.

Session 2aMUa

Musical Acoustics: Acoustics of Free-Reed Instruments I

James P. Cottingham, Chair
Coe College, Physics Dept., Cedar Rapids, IA 52402

Invited Papers

9:00

2aMUa1. Blown-closed free reeds with and without pipe resonators. James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402, jcotting@coe.edu)

The Asian mouth-blown free reed instruments are of ancient origin and use a symmetric free reed coupled to a pipe resonator. The reed behaves as a blown-open or outward striking reed, with playing frequency below both the resonant frequency of the pipe and the natural frequency of the reed. Although these instruments were known in Europe when the Western free reed family originated about 200 years ago, it does not appear that the mechanism used in the Western instruments was copied from them. In Western free reed instruments the reed tongue is offset from the opening in the frame, permitting operation on only one direction of air flow. Pipe resonators are not required and generally not used. If one of these reeds is coupled with a pipe, the sounding frequency can, within certain limits, be pulled to match the pipe frequency, and it behaves as a blown-closed or inward striking reed, with playing frequency below both the resonant frequency of the pipe and the natural frequency of the reed. This paper summarizes recent experimental research on the blown-closed free reed, with or without coupling to a pipe resonator, with emphasis on significant differences between the two situations.

9:30

2aMUa2. Free reeds coupling with either a vocal tract or rather small pipe. Laurent Millot (IDEAT UMR 8153, CNRS, Univ. Paris 1, ENS Louis-Lumière, 7 allée du Promontoire, BP 22, F-93161 Noisy-le-Grand Cedex, France, l.millot@ens-louis-lumiere.fr)

Free reed model used is based on a precise description of flow passing throw, the section defined between the shallot and the bended reed, and on the modelling of reed as a clamped-free cantilever beam. Classical description of loading pipe is based on wave reflection function, not designed to be excited by a flow but backward- and forward-traveling plane waves, so the free reed coupling with such a pipe model is physically irrelevant. Then, it is shown that an acoustical flow description is needed, and simple, for at least rather short pipe, giving more realistic sound simulations. It is then explained why a vocal tract description, physically relevant, is possible using only acoustical flow modeling for each element constituting the vocal tract model. In fact, flow acoustical modeling for elements is linked to electroacoustical analogies derived not as a low-pass approximations of input impedance but using the approximation of flow behavior for each element, valid for the whole audio range rather than only the low-pass register. Theoretical explanations should be given to explain why wave paradigma may be irrelevant and sound numerical simulations would be given for each coupling situation considered.

Contributed Paper

10:00

2aMUa3. Acoustical curiosities of an American reed organ. Thomas G. Muir (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, One Coliseum Dr., Univ., MS 38677)

Free reed organs of the 19th and early 20th centuries have long provided interesting pursuits involving their acquisition, restoration, history and musicology, as well as performance, and for some—acoustical studies of the instruments. There is a preservation society with a journal, and the literature includes a treatise, books, and learned articles. Some acoustical curiosities of one such instrument are described, including stop types, reed spectra, waveforms, loudness, and intonation.

TUESDAY MORNING, 27 OCTOBER 2009 LIVE OAK, 10:30 TO 11:15 A.M.

Session 2aMUb

Musical Acoustics: Acoustical Measurements on Musical Instruments

Daniel O. Ludwigsen, Chair
Kettering Univ., Physics Dept., 1700 W. Third Ave., Flint, MI 48504

Contributed Papers

10:30

2aMUb1. Physics of water crotales. Randy Worland (Dept. of Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416-1031, worland@ups .edu)

Contemporary composers writing for percussion often incorporate unconventional playing techniques with the use of traditional instruments. Among these extended techniques is the lowering of gongs and crotales into water as they are being struck, resulting in a unique glissando effect that involves changes in both pitch and timbre. The orchestral crotale has a relatively simple geometry and overtone structure, making it an appropriate starting point for the study of this performance technique. Building on previously published work on the physics of crotales [B. M. Deutsch, C. L. Ramirez, and T. R. Moore, J. Acoust. Soc. Am. **116**, 2427–2433 (2004)], an experimental investigation of the overtone frequencies and frequency ratios of a crotale as a function of water depth is presented. Mode images created using electronic speckle-pattern interferometry are also shown.

10:45

2aMUb2. The auger shell whistle. Daniel Zietlow and Thomas Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, dzietlow@rollins.edu)

We report on investigations of a musical instrument made from the shell *Terebra Turritella*, or auger shell. To our knowledge, this instrument has been developed only recently. The instrument is played from the large end using an edge tone, and a diatonic scale is achievable merely by drilling five tone holes in a straight line down the length of the spiral shell. It is surprising that a naturally occurring cavity can produce a diatonic scale, and implies that there is something special about the shape of the auger shell. However, experimental and theoretical investigations reveal that the auger shell does not actually have such a resonance structure.

11:00

2aMUb3. Why using complex stimuli for acoustical measurements may be necessary to get physically relevant analysis results. Laurent Millot (IDEAT UMR 8153, CNRS, Univ. Paris 1, ENS Louis-Lumière, 7 allée du Promontoire, BP 22, F-93161 Noisy-le-Grand Cedex, France, l.millot@ens-louis-lumiere.fr)

Classical methods for measurements and analysis in Musical Acoustics (and also Acoustics) rely on Fourier transforms, linear excitation superposition assumption, and monochromatic excitation or noise equivalents. It is shown that classical methods using measurements at a single point (input impedance measurement, for instance) may not be physically relevant. Indeed, only a time sampling and related time Fourier transforms are performed while physical phenomena are spatio-temporal. Even if one considers a monochromatic forward traveling wave excitation, the use of the

dispersion relation is not valid in spatio-temporal or spatio-frequency parameter space. It is shown that measurements need to be performed using also a spatial mapping to be physically relevant which implies at least Fourier bi-transforms (plane case). Then, it may be more simple and physically relevant to perform measurements with the whole complex excitation and use a frequency subband analysis tool, with a perceptive frequency subband mapping. Such a tool is described and it is shown that is permits the real-time switch between listening of partial to total resynthesis for any studied measurement signal, providing great surprises for the physical relevant subband(s) and the perceptive relevant ones. Audio performance will be given.

TUESDAY MORNING, 27 OCTOBER 2009 REGENCY EAST 3, 8:15 TO 11:50 A.M.

Session 2aPA

Physical Acoustics and Biomedical Ultrasound/Bioresponse to Vibration: 40th Anniversary of the Khokhlov-Zabolotskaya (KZ) Equation

Vera A. Khokhlova, Cochair

Moscow State Univ., Acoustics Dept., Leninskie Gory, 119992, Moscow, Russia

Mark F. Hamilton, Cochair

Univ. of Texas at Austin, Dept. of Mechanical Engineering, 1 University Station, Austin, TX 78712

Chair's Introduction—8:15

Invited Papers

8:20

2aPA1. Historical aspects of the Khokhlov–Zabolotskaya equation and its generalizations. Oleg V. Rudenko, Vera A. Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow 119991, Russia, rudenko@acs366.phys.msu.ru), and Mark F. Hamilton (The Univ. of Texas at Austin, Austin, TX 78712-0292)

Derivation of the Khokhlov–Zabolotskaya (KZ) equation provided a new approach to describing the combined effects of nonlinear propagation and diffraction in sound beams. In this paper, historical aspects of the KZ equation and its generalizations are presented. The interest in nonlinear acoustic beams of Academician Khokhlov and his colleagues at Moscow State University was inspired in the 1960s by emerging developments in laser physics and the corresponding models of nonlinear optical beams. The two cases, acoustical and optical, represent two limiting cases of nonlinear beams in weakly and strongly dispersive media, respectively, which required different theoretical approaches. The KZ equation and analogous nonlinear evolution equations of nonlinear wave physics are reviewed. It is illustrated how theoretical studies combined with numerical modeling resulted in predictions of new physical phenomena in nonlinear acoustic beams. Concurrently, newer applications of nonlinear acoustics such as parametric arrays, sonic booms, and medical acoustics stimulated the derivation of generalized KZ-type equations together with analytical and numerical methods to solve them. Modern applications and corresponding generalized KZ-type models that include effects such as frequency-dependent absorption, weak dispersion, scalar and vectorial inhomogeneities of the propagation medium, different orders of nonlinearity, and more accurate description of diffraction are presented.

8:40

2aPA2. Variations of the nonlinear equation for diffracting beams in fluids to study different modes of propagation in elastic media. Evgenia A. Zabolotskaya (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

Four variations of the equation for nonlinear acoustic beam propagation derived originally for a fluid will be reviewed. These variations describe finite amplitude beam propagation in isotropic elastic solids and in crystals, influence of diffraction on nonlinear Rayleigh waves, and nonlinear shear wave beams. The distinguishing features of each case will be discussed. The equation for nonlinear longitudinal wave beams in isotropic solids is based on nonlinear theory of elasticity and is close to the original equation for fluids. The equation for sound beams in crystals takes into account that the direction of energy propagation does not coincide with the direction normal to wavefronts. Nevertheless, the general form of the equation is similar to that of the original equation for fluids. Distinguishing features of Rayleigh waves are that they are two-dimensional and their nonlinearity is nonlocal. Because of the nonlocal nonlinearity, the evolution equation for this case was derived in the frequency domain using Hamiltonian formalism. The equation for nonlinear shear wave beams was derived in the cubic approximation in terms of particle displacement and it accounts for any polarization. Solutions of these evolution equations will be presented to illustrate phenomena specific to each case.

9:00

2aPA3. Group analysis of the Khokhlov–Zabolotskaya type equations. Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russia, and Ctr. for Industrial and Medical Ultrasound, APL, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, olegs@apl.washington.edu)

Khokhlov–Zabolotskaya (KZ) equation is the basic equation of the theory of acoustic beams propagating in quadratically nonlinear media. Being generalized for the case of a medium with general-type nonlinearity and expressed in dimensionless notation, it has the form $[u_z+P(u)u_\tau]_\tau=u_{xx}+u_{yy}$, where u represents waveform, τ is retarded time, z is axial coordinate, x and y are lateral coordinates, and $P(u)$ characterizes a nonlinear addition to the linear wave velocity: quadratic nonlinearity corresponds to $P(u)=u$, cubic one to $P(u)=u^2$, etc. No general analytical solution has been obtained for this nonlinear KZ-type equation, so numerical methods or asymptotic approximations are usually employed. Additional powerful mathematical tool is Lie group analysis that enables to find general symmetries of differential equations. These symmetries help to generalize known analytical and numerical solutions, derive new solutions, and obtain conservation laws. Results of group analysis of KZ and KZK equations are discussed for various nonlinearities $P(u)$. Examples of obtaining of new solutions and deriving of reduced equations are presented. It is also shown that the generalized KZ equation can be written in Euler–Lagrange form, which makes it possible to apply Nöther theorem and derive new conservation laws for nonlinear acoustic beams.

9:20

2aPA4. Nonlinear acoustic wave propagation in inhomogeneous moving media. Philippe Blanc-Benon (LMFA, UMR CNRS 5509, Ecole Centrale de Lyon, 69134 Ecully Cedex, France, philippe.blanc-benon@ec-lyon.fr), Mikhail V. Averiyanov (Moscow State Univ., Moscow 119991, Russia), Robin O. Cleveland (Boston Univ., Boston, MA 02215), and Vera A. Khokhlova (Moscow State Univ., Moscow 119991, Russia)

Extensive theoretical analysis, numerical studies, and both large-scale and laboratory-scale experiments have been dedicated to the problem of shock wave propagation in air during recent years. The current interest is motivated by supersonic civil transport which is necessarily affected by problems of sonic boom propagation in the atmosphere. The high-amplitude shock wave generated by a supersonic aircraft propagates through the atmosphere toward the ground and generates an acoustic field with non-uniform pressure distribution. Temporal characteristics and spatial structure of the sonic boom are influenced by aircraft trajectory, nonlinear effects, and diffraction and scattering by inhomogeneities. We review recent results from various teams based on a generalized KZK-type equation that includes the effects of a moving inhomogeneous media. Statistical analysis of the numerical solutions is performed, and the results are compared to experimental data obtained in the controlled laboratory-scale experiments conducted in the Ecole Centrale de Lyon anechoic wind tunnel.

9:40

2aPA5. Acoustic dissipation and finite-amplitude sound propagation in two-phase porous media. N. I. Pushkina, J. I. Osypik, and Ya. M. Zhileikin (Sci. Res. Comput. Ctr., Moscow State Univ., Vorobyovy Gory, Moscow 119992, Russia, n.pushkina@mererand.com)

Acoustic spectroscopy is known to be an important tool for studying various media. In the presented work, specific features of acoustic dissipation in two-phase porous media are theoretically studied and the propagation of a finite-amplitude attenuating sound beam described by the KZK-type equation is analyzed numerically for the case of a marine sediment. The KZK-type equation has been derived from the classical Biot equations, and it contains a dissipative operator corresponding to the frequency correction function $F(\kappa)$ in the Biot equations. In the present work, the properties of the correction function are studied and its well applicable representations are obtained. It is shown that the expansion of the correction function over κ^2 converges at $\kappa<5$. An asymptotic expansion of this function is obtained at large κ values. For high frequencies, simple dependence of viscous attenuation and phase velocity on parameters of a medium, useful for diagnostics of these parameters, has been found. The propagation of a finite-amplitude acoustic beam in a dissipative marine sediment has been numerically analyzed which showed that intense acoustic beam propagation can be accompanied by considerable non-linear phenomena while diffraction only weakly affects the process.

10:00—10:15 Break

10:15

2aPA6. Toward a better understanding of high intensity focused ultrasound therapy using the Khokhlov–Zabolotskaya–Kuznetsov equation. L. A. Crum, M. S. Canney, M. R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), O. V. Bessonova (Moscow State Univ., Moscow 119991, Russia), and V. A. Khokhlova (Univ. of Washington, Seattle, WA 98105)

High intensity focused ultrasound (HIFU) therapy is an emerging medical technology in which acoustic pressure amplitudes of up to 100 MPa are used to induce tissue ablation, often in combination with real-time imaging. The ultrasound energy is typically focused into a millimeter-size volume and used to thermally coagulate the tissue of interest while ideally sparing surrounding tissue. Nonlinear effects are important in HIFU as *in situ* intensities for clinical applications of up to 30 000 W/cm^2 have been reported. Since controlled experiments are often difficult to perform, especially *in vivo*, modeling can aid in understanding the physical phenomena involved in HIFU-induced tissue ablation. The Khokhlov–Zabolotskaya–Kuznetsov (KZK) equation is applicable to HIFU because it includes all of the basic physical phenomena that are relevant to HIFU including acoustic beams, diffraction, focusing, nonlinear propagation, shock formation, and dissipation. In this paper, an overview of several recent advances in KZK modeling for HIFU applications are described. It is shown that shock-induced heating in tissue can cause localized boiling in milliseconds; furthermore, the bubbles associated with boiling can significantly alter HIFU treatments. [Work supported in part by NSBRI SMST01601, NIH EB007643, and RFBR 09-02-01530.]

10:35

2aPA7. Shocking stones with the Khokhlov–Zabolotskaya–Kuznetsov equation. Robin O. Cleveland (Dept. of Mech. Engin., Boston Univ., 110 Cummington St, Boston, MA 02215, robinc@bu.edu)

In shock wave lithotripsy (SWL) high-amplitude acoustic waves generated outside the body are focused onto kidney stones in order to fragment them into pieces that are small enough to pass naturally. In order to investigate mechanisms of fragmentation and collateral damage to the soft tissue it is necessary to understand the acoustic field delivered to the tissue in and around the kidney. This is not easily accomplished experimentally and motivates the development of numerical models. The application of the Khokhlov–Zabolotskaya–Kuznetsov (KZK) equation to the SWL problem is discussed as there are two underlying assumptions that are challenged: the peak pressures are on the order of 50 MPa resulting in an appreciable acoustic Mach number of 0.02, and large focusing gains are employed which violate the paraxial approximation. Predictions using the KZK equation, with a layered model of the tissue path to the kidney, demonstrate that the waveform shape, in particular, the risetime, is strongly affected by the tissue path. However, once the waveform enters the urine in the collecting space of the kidney, the waveform heals and a sharp shock results within about 5 mm of propagation. [Work supported in part by NIH DK-43881.]

10:55

2aPA8. Nonlinear pulsing schemes for diagnostic ultrasound. Michalakis Averkiou (Dept. of Mech. Eng., Univ. of Cyprus, 75 Kallipoleos Str., 1678 Nicosia, Cyprus, maverk@ucy.ac.cy)

With the introduction of ultrasound contrast agents and the development of tissue harmonic imaging (THI), nonlinear acoustics has become a major research direction in diagnostic ultrasound. In THI an image is formed from the nonlinear components (due to nonlinear propagation) of the backscattered signal. Pulsing schemes have been invented to specifically detect the nonlinear components. These schemes are pulse inversion (PI), power modulation (PM), and their combinations (PMPI2, PMPI3, and PMPI4). The KZK equation is used to investigate these pulsing schemes in conditions that closely resemble ultrasound imaging in order to fully understand the properties of these schemes. Measurements of nonlinear propagation of such pulses were compared with the KZK predictions and were found in good agreement. Pulse inversion isolates the even harmonic components only and extracts the *total* amount of nonlinearity in those components. With pulsing schemes with different amplitudes (PM, PMPI2, and PMPI3), the *differential* nonlinearity between pulses is detected, with PMPI2 extracting the largest total amount. In addition, the schemes with amplitude modulation have a *nonlinear fundamental* component that suffers less than the higher harmonics from thermoviscous absorption and thus offers more signal in penetration limited cases.

11:15

2aPA9. Statistical solutions of the Khokhlov–Zabolotskaya–Kuznetsov equation for analyzing mechanisms of image quality enhancement in tissue harmonic imaging. Xiang Yan (Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78712-0292) and Mark F. Hamilton (The Univ. of Texas at Austin, Austin, TX 78713-8029)

Statistical solutions of the Khokhlov–Zabolotskaya–Kuznetsov equation were obtained to analyze mechanisms contributing to image quality enhancement in tissue harmonic imaging (THI). The focus is on suppression of image clutter due to phase aberration and reverberation. Tissue heterogeneity is modeled with a random phase screen characterized by its variance and spatial correlation length. Solutions for the mean intensities of the linear (fundamental mode) and second-harmonic fields were derived from a focused Gaussian beam that is transmitted through a phase screen located an arbitrary distance from the source. The random phase variations of the screen are assumed to be small and described by a Gaussian autocorrelation function. The solutions are validated by comparison with ensemble averages of direct numerical simulations. A benefit of the analytical approach is separation of the different contributions to deformation of the beam by the phase screen, and the statistical approach is convenient for quantifying the merits of THI. The degree to which THI reduces beam deformation is assessed using a measure based on signal-to-clutter ratios introduced previously for this purpose by C. E. Bradley [Proceedings of 17th International Symposium on Nonlinear Acoustics (AIP, New York, 2006)]. Statistical solutions will also be presented for backscattering. [Work supported by NIH DK070618.]

Contributed Paper

11:35

2aPA10. Nonlinear acoustical beam formation and beam profiles in fluids. Cristian Pantea and Dipen N. Sinha (Mater. Phys. and Applications, MPA-11, MS D429, Los Alamos Natl. Lab., Los Alamos, NM 87545)

Nonlinear acoustical beam formation in fluids is investigated in several different configurations: (i) collinear mixing due to simultaneous excitation of a single transducer by two different high frequency signals, (ii) noncollinear mixing due to excitation from two separate transducers, (iii) nonlinear de-modulation of an amplitude modulated fixed frequency, and (iv) nonlinear mixing due to excitation of a single transducer by a fixed frequency and a chirped frequency signal. In all cases, the difference frequency acoustic beam generated by nonlinear mixing in the fluid is investigated. In contrast to using solids as the nonlinear medium, fluids have the advantage that the beam can be scanned in all directions allowing the determination of three-dimensional beam profile. The experiments were performed mainly in water and Fluorinert FC43. In the collinear measurement configuration, where the nonlinear down-converted beam is produced from a single transducer, the experimental results are compared with the predictions from the KZK equation. Electronics and transducer nonlinearity and the role of transducer "effective diameter" will also be discussed.

Session 2aPP

Psychological and Physiological Acoustics: Cognitive Aspects of Hearing

Craig A. Champlin, Chair

Univ. of Texas at Austin, Communication Sciences and Disorders, 1 University Station, Austin,TX 78712-0114

Contributed Papers

8:15

2aPP1. Signal-to-noise-ratio loss in hearing-impairment and feature enhancement. Woojae Han (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, whan5@illinois.edu), Riya Singh, and Jont Allen (Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801)

The primary purpose of this study is to investigate detailed consequences of hearing loss and to scrutinize if individual hearing impaired (HI) listeners exhibit individual patterns in their consonant confusion scores. 16 English nonsense CV syllables, by 18 various talkers in five speech-weighted noise (−12, −6, 0, +6, and +12-dB signal-to-noise-ratios) and quiet conditions, were randomly presented to 11 HI subjects (18 HI ears) having sensorineural hearing losses at two different amplified levels: with and without NAL-R (similar to fitting hearing aids). The HI ears have sensorineural hearing loss with flat (mild-to-moderate-to-severe), ski-slop at low- and/or high-frequency regions, and cookie-bite configurations. To characterize the consonant loss, the performance of each HI ear was compared to average normal-hearing listeners in masking noise. Preliminary analysis shows that each HI ear has an individual profile, which is a characteristic of the impaired ear. Once the noise-sensitive consonants are identified, the aim is to use signal processing techniques to selectively enhance features to make them audible in noise. [Work supported by the NIH Grant RDC009277A.]

8:30

2aPP2. Comodulation masking release to detect cochlear dead regions in hearing impaired ears. Riya Singh (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Everitt Lab, 1406 W. Green St., Urbana, IL 61801), Woojae Han, and Jont B. Allen (Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801)

The purpose of this study is to investigate if comodulation masking release experiments with hearing impaired (HI) individuals can be used as a reliable method to detect dead regions in the cochlea, if any. The experiment involves detection of a pure tone in a narrow band of noise (the on-signal band) in presence of four other narrow bands of noise that fall outside the auditory bandwidth at the center frequency of the on-signal band. These flanking bands are either all co-modulated so as to have the same amplitude envelope as the on-signal band or are randomly modulated and are presented monaurally to an impaired ear. Each band is level adjusted to be audible to the HI ear. Data are being collected for both these cases from normal-hearing listeners and HI listeners with no dead regions. Every HI individual also undergoes psychophysical tuning curve and threshold-equalizing noise tests which are currently used tests to detect cochlear dead regions. Each HI individual also takes speech perception tests with CV syllables under various noise conditions. The consonant loss profiles from the perception experiments help make a reasonable estimate of the possibility of a dead region. [Work supported by NIH.]

8:45

2aPP3. Infants' vowel discrimination in modulated and unmodulated noise. Lynne A. Werner (Dept. Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105-6246)

Adults detect and discriminate masked sounds better in temporally modulated maskers than in unmodulated maskers. It is not known whether the same is true of infants. In this study, 7–9-month-olds learned to respond when a repeated vowel changed from /a/ to /i/ or from /i/ to /a/ in speech spectrum noise. Discrimination was assessed using an observer-based procedure. In one condition the noise was unmodulated. In the other, the noise was modulated with the envelope of single talker speech. The level of the noise was 60-dB SPL. For infants in group 1, the level of the vowel in both conditions was set so that $d'=1$ was achieved in the unmodulated noise; discrimination in modulated noise was the second condition tested. For infants in group 2, the level of the vowel was fixed at the average level used to test the infants in group 1; discrimination in modulated noise was tested first. In both groups, d' in the modulated noise condition averaged only 0.5. Thus, infants were unable to take advantage of masker modulation to improve speech discrimination. Their speech discrimination was poorer in modulated than in unmodulated noise. [Work supported by NIH R01 DC00396 and P30 DC04661.]

9:00

2aPP4. Enduring non-verbal memory for spectral timbre. Denis McKeown and Tom Mercer (Dept. of Psych., Univ. of Leeds, Leeds, United Kingdom)

It is often assumed that auditory sensory memory disappears within a few seconds or is so smudged as not to permit fine discriminations between successive sounds beyond a few seconds. But studies have been confounded by proactive interference, task difficulty, and verbal labeling. In Experiment 1, participants heard two complex tones separated by a silent retention interval of 2, 4, 8, 16, or 32 s. Tones varied widely in pitch from trial to trial (so no standard memory could be formed) and were non-speech-like (difficult to verbalize), and very long inter-trial intervals were used (32 s) to reduce interference from prior sounds. After the tone-pair presentation, listeners made a same-different response based on slight differences in spectral timbre. Discrimination performance expressed as d', at each retention interval revealed that the persistence of memory for spectral timbre surpassed the usually quoted lifetime of some few seconds and for certain participants it endured for 32 s. In Experiment 2, overall stimulus levels were roved within trials to rule out loudness cues in the discrimination. It is proposed that reasonably robust non-verbal memory traces for spectral timbre persist for several seconds allowing listeners to make fine discriminations of changes to a spectrum.

9:15

2aPP5. Efficient coding of correlated complex acoustic dimensions through active listening. Christian E. Stilp, Timothy T. Rogers, and Keith R. Kluender (Dept. of Psych., Univ. of Wisconsin, 1202 W. Johnson St., Madison, WI 53706, cestilp@wisc.edu)

Efficient coding extracts and exploits redundancy to optimize information processing in sensorineural systems. Stilp *et al.* [J. Acoust. Soc. Am. **124**, 2496 (2008)] reported evidence for rapid and efficient adaptation to correlation among complex acoustic attributes. In the study, following passive exposure to highly correlated stimulus features, discriminability of sound pairs violating the correlation is temporarily lost before subsequent recovery via active testing with stimuli whose features were poorly correlated. The present study examines listeners' ability to extract and exploit correlation between stimulus attributes exclusively through active testing. Listeners discriminated stimuli (AXB) for which two complex, independent dimensions, attack/decay (AD) and spectral shape (SS), were

highly correlated (r^2=0.96). In the first testing block of 128 trials, discrimination of sound pairs respecting the correlation is superior to sound pairs violating the correlation. Only through successive testing blocks, listeners discover variance orthogonal to the otherwise perfect correlation between AD and SS, and discrimination recovers to baseline levels. Listener performance will be discussed within the context of maximum likelihood and connectionist models. [Work supported by NIDCD.]

9:30

2aPP6. Neural representation of speech sounds: Study using frequency following response. Radhika Aravamudhan (George S. Osborne College of Audiol., Salus Univ., 8360 Old York Rd., Elkins Park, PA 19027, raravamudhan@salus.edu)

The neural encoding of an acoustic signal begins in the auditory nerve and travels to the auditory brainstem and further to the auditory cortex. Previous studies have used nonspeech signals like tones and clicks to evaluate the integrity and synchrony of the auditory pathway. Most of the previous research has focused on how any acoustic signal is perceived by using behavioral methods. One of the main areas in understanding how we hear the signal depends on how this acoustic signal is represented in the auditory pathway. A number of studies have studied the acoustic parameters that influence the perception of speech, but very few of them have focused on how these acoustic changes are represented in the auditory pathway. Since the signal representation in the auditory pathway is very crucial to how the signal is perceived, studies that focus on the relationship between the changes in the input acoustics and its influence on neural representation become very essential. In the current project the neural representation of isolated vowels and vowels in CV context was recorded using human frequency following response. The results will be discussed in the paper.

9:45

2aPP7. How can a video game cause panic attacks? I. Effects of an auditory stressor on the human brainstem. Judith L. Lauter, Elizabeth Mathukutty, and Brandon Scott (Dept. of Human Services, Stephen F. Austin State Univ., Box 13019 SFA Station, Nacogdoches, TX 75962, jlauter @sfasu.edu)

The auditory brainstem response (ABR) was recorded during simultaneous binaural presentation of two types of sounds: (1) condensation clicks presented through in-the-ear earphones at 43.1/s, 60 dB nHL; and (2) recordings of breathing sounds, presented through supra-aural headphones, at levels adjusted by participants to be equivalent to the clicks. In alternate blocks, the breathing sounds were either: (1) a recording of quiet breathing (blocks 1, 3, and 5) or (2) a recording of erratic (stressed) breathing (blocks 2 and 4). The erratic breathing was modeled on a video game soundtrack in which the character was represented as running, wounded, and frightened. Four 2048-sweep ABR waveforms were collected in each of the five blocks, and the mean amplitude of ABR peak V was calculated over each set of four waveforms. Results indicate a significant decrease in the amplitude of ABR peak V during erratic breathing versus quiet breathing. Implications include (1) possible new evidence of the effect of selective attention on the ABR, (2) the potential for using auditory stressors to study the central physiology of emotional responses in humans, and (3) clues to physiological correlates of the effects of certain video games known to evoke panic attacks in susceptible players.

TUESDAY MORNING, 27 OCTOBER 2009 — REGENCY WEST 1 & 2, 9:00 TO 11:50 A.M.

Session 2aSC

Speech Communication: Advances in Speech Synthesis

Norma S. Antonanzas-Barroso, Chair
UCLA School of Medicine, Head and Neck Surgery, 31-24 Rehab Center, Los Angeles, CA 90095-1794

Invited Papers

9:00

2aSC1. Talking heads: Speech synthesis and embodied cognition. Philip Rubin, Gordon Ramsay (Haskins Labs., 300 George St., New Haven, CT 06511 and Yale School of Medicine, 333 Cedar St., New Haven, CT 06510, rubin@haskins.yale.edu), and Eric Vatikiotis-Bateson (Univ. of British Columbia, Vancouver, BC V6T 1Z4, Canada, evb@interchange.ubc.ca)

This presentation provides a brief overview of 300 years of effort toward the creation of talking heads: mechanical, electronic, and/or computational models of human speech. Speech, language, communication, and cognition are fundamentally shaped, in part, by both biological and physical factors. To understand this grounding and how to effectively replicate its most salient aspects in synthesis systems requires us to pay serious attention to the structure, kinematics, and dynamics of the articulators; the organization and characterization of complex, emergent behavior in multimodal systems; and the consideration of how events within such systems unfold over multiple time scales. New approaches that will help advance our knowledge and improve our synthesis tools and techniques will be discussed.

9:20

2aSC2. Challenges in evaluating the intelligibility of text-to-speech. Ann Syrdal (AT&T Res., 180 Park Ave., Rm. D159, Florham Park, NJ 07932-0971, syrdal@research.att.com), Murray Spiegel (Telcordia Technologies, Piscataway, NJ 08854-4151), Deborah Rekart (AT&T Services, Dallas, TX 75287), Susan R. Hertz (Nova Speech LLC and Cornell Univ., Ithaca, NY 14850), Tom Carrell (Univ. of Nebraska, Lincoln, NE 68583-0738), H. Timothy Bunnell (Alfred I duPont Hospital for Children, Wilmington, DE 19803), and Corine Bickley (Gallaudet Univ., Washington, DC 20002)

Text-to-speech (TTS) technology imposes different constraints on intelligibility than those sufficient for the evaluation of other speech communication systems. For example, the newly revised standard S.2-2009 explicitly excludes TTS from the speech communication systems it covers. Since there is no current standard appropriate for evaluating TTS intelligibility, the ASA Standards Bioacoustics (S3) working group on Text-to-Speech Technology (WG91) was formed with the initial goal of developing such standard. We describe several ways in which standard methods of testing speech intelligibility are unsuitable for TTS technology and outline our approach to overcoming these limitations. We present an overview of our proposed standard, which is currently nearing its final draft stages.

9:40

2aSC3. Recombinant speech synthesis: Natural text-to-speech synthesis with prosodic control. Esther A. Klabbers, Taniya Mishra, and Jan P.H. van Santen (Div. of Biomedical Comput. Sci., Oregon Health & Sci. Univ., 20000 NW Walker Rd., Beaverton, OR 97006, klabbers@cslu.ogi.edu)

Unit selection text-to-speech synthesis methods rely on large corpora to cover all phoneme sequences in as many prosodic contexts as possible. This coverage is rarely complete except in limited domains. This becomes particularly salient when using prosodic markup to generate specific prosodic patterns (e.g., emphatic stress). An architecture is proposed combining the naturalness of unit-selection synthesis with the requirement of prosodic control. The speech corpus consists of multiple sub-corpora, each optimized to cover a "linguistic subspace"; subspaces include phoneme sequences, left-headed feet, sentence structures, and paralinguistic categories. The system relies on the superpositional model of intonation to decompose natural pitch contours into component contours, e.g., phrase curves (corresponding to phrases) and accent curves (corresponding to left-headed feet); on analogous methods for timing; and on hybridization methods to implement paralinguistic features. During synthesis, phoneme sequences, curves, and parameters are generated from the sub-corpora, optionally modified as per prosodic control tags, and "re-combined." The explicit representation in terms of component curves allows for complete prosodic control, while the naturalness of the prosodic patterns is guaranteed by extracting these curves from natural speech and smoothly modifying them, thereby preserving important natural detail. [Research supported by NSF grant 0205731, "Prosody generation in child-oriented speech."]

10:00—10:30 Break

10:30

2aSC4. Speech transformation: Increasing intelligibility and changing speakers. Alexander Kain (Div. of Biomedical Comput. Sci. Oregon Health & Sci. Univ., 20000 NW Walker Rd., Beaverton, OR 97206, kaina@ohsu.edu)

Speech transformation changes an aspect of speech without changing its message, typically using a process of analysis, feature modification, and synthesis. The feature modification step can consist of a (trainable) mapping function or a hybridization of several feature sets for the purposes of perceptual experimentation. First, speech transformation approaches are presented that aim to increase the intelligibility of speech. One approach is used in the context of increasing the intelligibility of conversationally spoken speech for hearing-impaired listeners. Another approach aims to increase the intelligibility of speaking-impaired individuals by the general population. Second, transformation approaches are presented which aim to change a source speaker's speech (natural or TTS-generated) to sound as if a specific target speaker had spoken it (also known as voice transformation).

10:50

2aSC5. Quasi-articulatory synthesis as a tool for basic science and education. Helen M. Hanson (ECE Dept., Union College, 807 Union St., Schenectady, NY 12308)

HLsyn is a quasi-articulatory speech synthesis system based on Klatt's formant synthesis model [D. K. Klatt and L. C. Klatt, J. Acoust. Soc. Am. **87**, 820–857 (1990)]. The elements of a circuit model of the vocal tract are derived from cross-sectional areas of constrictions and subglottal pressure. Solution of the circuit results in pressure drops across constrictions, which in turn lead to source characteristics (both periodic and noise). The circuit model imposes constraints on the relations between the pressures and flows in the circuit, resulting in more natural variation in the amplitudes of two simultaneous sources, e.g., a periodic source and a noise source. While the virtues of HLsyn as a speech synthesis system have often been sung [e.g., Hanson and Stevens, J. Acoust. Soc. Am. **112**, 1158–1182 (2002)], HLsyn has other uses that may go unnoticed. Using HLsyn to explore or simulate the production of certain sounds often leads to insights about how speech is produced. That is, this synthesizer can be used to explore the basic science of speech production and perception. Likewise, HLsyn as a tool in basic speech science courses brings equations and circuits to life for students. Examples of insights gained through HLsyn will be described. [Work supported by NIH.]

11:10

2aSC6. Advances in simulation of sentence-level speech production with kinematic models of the vocal tract and vocal folds. Brad H. Story (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721)

Speech simulation refers to a system in which computational models are used to simulate the physical processes of human sound production. These processes consist primarily of vocal fold vibration and acoustic wave propagation in the tracheal, nasal, vocal tract systems, and radiated acoustic output. Although simulations of various aspects of speech production are typically limited to short time scales, this presentation will focus on using kinematic models of the vocal folds and vocal tract shape to generate speech at the level of words and sentences. Specifically, the vibrating medial surface of the vocal folds is represented by a kinematic model which allows for direct time-dependent specification of fundamental frequency, amplitude of vibration, glottal geometry, and glottal symmetry. The vocal tract shape is also governed by a kinematic model, based on data collected with MRI and x-ray microbeam techniques, that allows for specification of time-dependent cross-sectional area changes in an acoustic waveguide. When these models are coupled, the result is an acoustically-interactive simulation of the sound production process from which pressures and airflows are generated. The components of the system will be presented and then used to demonstrate samples of simulated speech. [Work supported by NIH R01-DC04789.]

11:30—11:50 Panel-Discussion

Session 2aSP

Signal Processing in Acoustics: Time-Frequency Theory and Applications

Edmund J. Sullivan, Chair
EJS Consultants, 46 Lawton Brook Ln., Portsmouth, RI 02871-1032

Chair's Introduction—7:55

Invited Papers

8:00

2aSP1. Wave propagation in phase-space. Leon Cohen (Dept. of Phyiscs, City Univ.-Hunter College, 695 Park Ave., New York, NY 10021)

We derive exact expressions for the spreading of a propagating pulse in a dispersive medium. We address both the deterministic and the random case, and also allow for the possibility of frequency-dependent attenuation. The conditions for contraction of a pulse before it eventually spreads to infinity are derived. In the interpretation of the expressions, we show that considerable insight may be achieved if the pulse is analyzed in phase-space where the phase-space distribution may be time-frequency or position-wavenumber. Applications to a train of pulses relevant to reverberation and clutter is also discussed. [Work supported by ONR 321US.]

8:20

2aSP2. Time-frequency and position-wavenumber acoustic signal analysis. Patrick J. Loughlin (745 Benedum Hall, Univ. of Pittsburgh, Pittsburgh, PA 15261, loughlin@pitt.edu)

The spectrogram, which is a plot of the spectral intensity of a signal over time, has been widely used in acoustic signal processing because many signals, such as speech, animal vocalizations, music, and the sonar backscatter from elastic objects, have frequencies that change over time, which convey important information about the signal or source from which it originated. Since the development of the spectrograph at Bell Laboratories in the 1940s, more modern methods for time-frequency analysis, such as the Choi–Williams and Zhao–Atlas–Marks distributions, have been developed, which overcome some of the limitations of the spectrogram and, in particular can show time-frequency detail in the signal that is obscured by the spectrogram. We will discuss these methods and the general area of time-frequency acoustic signal analysis with examples drawn from a variety of applications. A particular focus will be made on showing how time-frequency analysis, and also position-wavenumber analysis, can be used to formulate and gain physical insights into dispersive pulse propagation. We will also comment on and illustrate the use of wavelets for time-frequency analysis. [Work supported by ONR 321US.]

8:40

2aSP3. Gibbs sampling for modal arrival time and amplitude estimation from time-frequency representations of acoustic signals. Zoi-Heleni Michalopoulou (Dept. of Math. Sci., New Jersey Inst. of Technol., Newark, NJ 07102, michalop@njit.edu)

A Gibbs sampling-maximum *a posteriori* approach is developed for modal arrival time and amplitude estimation from time-frequency representations of broadband acoustic signals propagating in underwater media. The goal is to obtain accurate estimates of arrival times of propagating modes and corresponding amplitudes, which can then be employed for source localization and geoacoustic inversion. The method provides uncertainty information on both modal arrival time and amplitude estimates, typically unavailable when traditional methods are used. Estimates and uncertainty are propagated through an inversion process, generating posterior probability distributions of source range and geoacoustic properties. [Work supported by ONR.]

9:00

2aSP4. Adaptive noise and interference removal from speech based on a modified short time Fourier transform. Douglas J. Nelson (Natl. Security Agency, 9800 Savage Rd., Fort Meade, MD 20755-6214)

A simple linear adaptive method for removing noise and interference from speech is presented. The method is based on a linear time-frequency representation in which the short time Fourier transform (STFT) is concentrated (reassigned) in frequency. The concentration process results in a complex-valued surface in which speech components and narrowband AM/FM interference components are essentially time-varying spectral impulses. Unwanted components may be removed from the surface, and the clean signal with these components removed may be reconstructed as a linear time marginal, computed by integrating the surface with respect to frequency. This process is linear and results in a nearly distortion-free reconstructed signal. Moreover, unlike processes like methods such as spectral subtraction, the process requires no Fourier inversion. This process is computationally efficient since the STFT is implemented as a bank of recursive two tap filters.

9:20

2aSP5. Estimation of seismic interface wave dispersion in geoacoustic inversion. Hefeng Dong (Dept. of Elec. & Telec., Norwegian Univ. of Sci. & Techn., NO-7491 Trondheim, Norway) and Lanbo Liu (Univ. of Connecticut, Storrs, CT 06269-2037)

Shear wave velocity variation as depth in underwater environment is closely related to dispersion of seismic interface waves traveling along the water/sediment boundary. An estimate of the dispersion of seismic interface waves can be obtained by time-frequency analysis. There are different methods for the time-frequency analysis to estimate dispersion of the interface waves. Phase velocity dispersion can be estimated by multi-sensor method, while group velocity dispersion can be estimated by single-sensor method. Multi-sensor method uses array data by knowing sensor-spacing, which gives the average estimate of the phase velocity within the ranges covered by the array. Single-sensor method uses one trace at a time by knowing the distance between source and the sensor, and gives the local group velocity estimate. In this paper, time-frequency analysis methods for both phase- and group-velocity estimations are presented. An experimental example for estimation of shear wave velocity variation as depth in the upper layers of the sediment is presented by inverting the estimated dispersion relations of seismic interface waves [Work supported by NFR under No. 186923/I30].

9:40

2aSP6. Space-time-frequency analysis of the scattered acoustic wave field from elastic shells recorded by bistatic sonar systems. Karim G. Sabra and Shaun D. Anderson (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Graduate Box 1000, Atlanta, GA 30332, karim.sabra@me.gatech.edu)

An ongoing challenge for underwater sonar systems is to discriminate a man made target (shell) from surrounding clutter returns and to provide robust classification features for the estimation of the physical target characteristics (e.g., shell thickness and material properties). To this end, time-frequency analysis, and, in particular, Wigner–Ville analysis, has been shown to provide a robust processing tool for interpreting the evolutional time dependent aspect of the scattered acoustic wave field from elastic shells. The design of a robust space-time-frequency bistatic sonar system to enhance the target detection of shells with the use of a sensor array will be presented. Practical implications for bistatic mine countermeasure sonar systems, using a network of autonomous underwater vehicles, will be discussed.

10:00—10:15 Break

Contributed Papers

10:15

2aSP7. Application of compressive sampling to passive sonar signals. R. Lee. Culver (Grad. Prog. Acoust. & Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804) and N. K. Bose (Penn State Univ., State College, PA 16804)

Recently the compressive sampling (CS) paradigm has generated considerable interest in the signal processing community because it offers the potential to fully characterize signals without satisfying the Nyquist requirement (sampling frequency must be more than twice the highest frequency in the signal). Signal compression itself is not new; it is used in all file compression algorithms. However, it generally requires that all coefficients be generated, many or most of which are discarded and only a few are transmitted. The theoretical basis for CS has been presented in a number of signal processing and statistics journal articles [e.g., Candès and Wakin, IEEE SP (March 2008)]. CS is fixed (non-adaptive) and is efficient in that only the coefficients required for signal characterization are calculated. The key requirement of CS is to find a basis in which the signal representation is sparse and thus can be represented with a minimum number of coefficients. Here we explore possible benefits of applying CS to sonar signals, including signal compression, bandwidth reduction, and exploitation of sparse sampling geometries. [Work sponsored by ONR Undersea Signal Processing.]

10:30

2aSP8. Time-frequency analysis techniques for long range sediment tomography. Gopu R. Potty and James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02879)

Long range sediment tomography technique uses travel-time of acoustic normal modes at different frequencies to invert for the geoacoustic properties. The modal travel-times are calculated from the time-frequency analysis of the acoustic time series. Travel times in the frequency range 1 Hz to less than 1 kHz modes 1–6 obtained using broadband sources are typically used as data for the inversion. Various time-frequency analysis techniques used in the past in the context of long range sediment tomography will be presented. These techniques include short time Fourier transform, wavelet transform, matching pursuit algorithms, and dispersion based short time Fourier transforms. Performance of these techniques will be compared using data from various types of broadband sources deployed during different field experiments. [Work supported by Office of Naval Research.]

10:45

2aSP9. Using time corrected instantaneous frequency to detect source motion of a towed projector. Jack A Shooter (Appl. Res. Labs, Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, shooter@arlut.utexas.edu)

The TCIF algorithm [S. Fulop and K. Fitz, J. Acoust. Soc. Am. 119, 360–371 (2006)] was applied to a narrow band projector from the Shallow Water 2006 Hudson Canyon experiment. To observe Doppler shifts near CPA, short overlapping FFTs were formed from the measured time series data. The appearance of a secondary Doppler component indicates that the source was wobbling or fishtailing during the tow event. Analysis indicated the source was moving back and forth with amplitudes of about 6 in., at a rate of 1–2 cycles/s. Detecting these source dynamics required a basic FFT length less than 18% of the variable period and signal-to-noise ratio greater than 18 dB. Narrowband filtering the data significantly reduced interference from outside the band of interest. The FFT lengths were so short the frequency bin spacing was much greater than the wobble frequency variation, yet TCIF permitted the source wobble period to be observed. [Work supported by IR&D.]

11:00

2aSP10. Temporal analysis of human motion signatures. Alexander Ekimov, Marshall Bradley, and James Sabatier (NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, aekimov@olemiss.edu)

Passive and active signals characterize human motion. Passive signals produced by footsteps are recorded with microphones and geophones. Active signals are measured with Doppler techniques and characterize the oscillatory motion of human body parts with different acoustic cross-sections and velocities. Simultaneous measurements of human passive and active signatures show temporal synchronization. This synchronization results in equal cadence frequencies for human motion signatures. These signatures are compared to a mathematical model of human motion [Boulic *et al.*, "A global human walking model with real-time kinematic personification," Vi-

sual Comput. (1990)]. This empirical model predicts the motion of key body parts including shoulders, elbows, fingers, hips, knees, and toes for an "average" human walking at a constant velocity. Speeds of the body parts depend on the walking velocity and the size of the individual as parametrized by thigh height. Comparison of experimental data with the predictions of the mathematical simulations enhances the capability to characterize humans. Measurements of the motion signatures of walking humans using a multimodal sensor suite (seismic, passive, and active ultrasonics, and radar sensors) are presented and discussed. [Work supported by Department of the Army, Army Research Office Contracts W911NF-04-1-0190 and W911NF-08-1-0389.]

11:15

2aSP11. Blind deconvolution of quadratic time-frequency representations of musical signals for reverberation feature extraction. Gang Ren (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, garen@ece.rochester.edu), Mark F. Bocko, and Dave Headlam (Univ. of Rochester, Rochester, NY 14627)

An acoustic space is uniquely characterized by its reverberant behavior. Due to the complexity of the multiple reflections and diffraction of sound in an enclosure, currently available system identification algorithms cannot effectively estimate the impulse response of a concert hall simply from musical recordings in that space. This paper reports the use of blind image deconvolution methods to construct echo patterns from quadratic time-frequency representations of reverberant recordings of music. First, a quadratic time-frequency analysis is performed to decompose the musical signal into its constituent harmonic components. Quadratic time-frequency analysis methods are known to give enhanced resolution in the time-frequency plane in comparison to conventional Fourier analysis. Reverberant features then appear as blur in the time-frequency plane, which can be estimated by employing the methods of blind deconvolution of this "image" of the sound. The proposed algorithm retrieves both the blur pattern, which corresponds to the reverberation echogram, and the direct acoustic signal. By choosing the time-frequency frame scale and smoothing window, a multi-resolution analysis of the underlying acoustic impulse response is obtained. Various quadratic time-frequency analysis methods are evaluated and their relative performance is reported. The proposed methods are also compared to existing dereverberation algorithms.

11:30

2aSP12. Speech localization in any direction using power and frequency signatures, gradients, and differences. Colin L. Barnhill and James West (ECE Dept., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218, cb@jhu.edu)

Speech localization is a relatively simple task for a human but, often, a difficult task for acoustic arrays. Although arrays can localize impulsive and narrow-band sources through the use of cross-correlation and subspace methods (like MUSIC), these methods break down when there is low SNR, reverberant conditions, or the source is neither impulsive nor narrow-band. Interesting speech sources (for teleconferencing or surveillance) are most often in environments where many of the breakdown conditions exist. New or multiple methods are necessary to reliably localize speech. A new algorithm will be presented that makes use of multiple localization methods to locate the desired speech signal. The localization methods are based on the time and frequency power signatures, gradients, and differences of third order supercardioid beams. Multiple beams are used to spatially segregate power signatures, and the results are clustered to determine position. Using a spherical array, it is possible to localize speech in any direction using these beams. Experimental acoustic and mathematical results in real room situations will be presented.

TUESDAY MORNING, 27 OCTOBER 2009 RIO GRANDE CENTER, 10:20 TO 11:45 A.M.

Session 2aUW

Underwater Acoustics: Reverberation Measurements and Modeling I

Eric I. Thorsos, Cochair
Univ. of Washington, Applied Research Lab., 1013 NE 40th St., Seattle, WA 98105

John S. Perkins, Cochair
Naval Research Lab., Code 7140, Washington, D.C. 20375

Jason D. Summers, Cochair
SAIC, Acoustic and Marine Systems Operation, 10401 Fernwood Rd., Bethesda, MD 20817

Chair's Introduction—10:20

Invited Paper

10:25

2aUW1. Update on the reverberation modeling workshops. John S. Perkins (Naval Res. Lab., Washington, DC 20375, john.perkins@nrl.navy.mil) and Eric I. Thorsos (Univ. of Washington, Seattle, WA 98105-6698)

In order to investigate the status of reverberation modeling and establish well-defined benchmark problems, two reverberation modeling workshops have been jointly sponsored by the Office of Naval Research and the Navy Program Executive Office C4I (PMW 120). The first workshop was held 7–9 November 2006, and the second during 13–15 May 2008. This paper presents the approach used in formulating the reverberation problems posed to workshop participants and shows examples of the modeling agreement obtained for some of the workshop problems. Issues that have arisen as workshop participants have collaborated in their attempts to produce consensus solutions will be pointed out. Finally, some of the problems to be addressed at a proposed third workshop will be discussed.

Contributed Papers

10:45

2aUW2. Ray versus mode differences in reverberation modeling solutions for environments with high boundary scattering loss. Eric I. Thorsos, Frank S. Henyey, Jie Yang, and Stephen A. Reynolds (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu)

Several of the problems for the first Reverberation Modeling Workshop yielded interesting differences between solutions obtained with ray and normal mode methods. These particular problems were defined with high boundary scattering loss. A bottom reverberation case at 3.5 kHz with a down-refracting sound speed profile (Problem VI) will be considered as a case in point. The ray solutions show a "direct path" contribution unaffected by the bottom scattering loss as long as a direct path can reach the bottom, while the mode solutions obtained to date show a lower reverberation level during this period due to modal attenuation. These differences occur in both incoherent and coherent reverberation solutions for both rays and modes. Arguments will be presented that indicate the correctness of the ray solutions for this case. Suggestions will also be made on how the mode approach can be used to obtain solutions in agreement with the ray method. [Work supported by ONR and the Navy Program Executive Office C4I (PMW 120).]

11:00

2aUW3. Reverberation versus time or reverberation versus range? A definitive relationship. Chris Harrison (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, Italy)

Strictly reverberation consists of all the back-scattered contributions from many locations that arrive at one absolute time regardless of their respective travel times. However, for calculation purposes, particularly at long range, it is often more convenient to assume the scatterers to be close together at a single location with corresponding propagation loss for the outward and return paths. It would be useful to know whether the latter approximation is a safe one or not. An analytical method of calculating reverberation either at a fixed time or at a fixed range will be presented. The ratio of these quantities is found to depend on the (vertical) angular spread of the contributing paths. In isovelocity water at long ranges, mode-stripping reduces the angular spread to the point where fixed-range reverberation is indistinguishable from fixed-time. The greatest effect is at shorter ranges where there is no significant mode-stripping, and the ratio rises to roughly 1 plus 0.855 times the square of the critical angle (in radians). Thus the effect is typically less than 1 dB at any range.

11:15

2aUW4. Modeling acoustic propagation in shallow water using finite elements. Marcia J. Isakson and Preston S. Wilson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713, misakson@arlut.utexas.edu)

Finite element models approach an exact solution of the Helmholtz equation as discretization density increases. Therefore, these solutions are an excellent benchmark model for reverberation studies. This study will present finite element reverberation solutions for two-dimensional shallow water waveguides for times up to 4 s. Five waveguides are considered: rough bottom only, rough surface only, rough surface and rough bottom, rough interfaces with summer (downward refracting) profile, and rough interfaces with winter (upward) refracting profile. [Work sponsored by Office of Naval Research, Ocean Acoustics.]

11:30

2aUW5. High-frequency broadband coherent reverberation predictions for the reverberation modeling workshop. Kevin D. LePage (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, Italy)

The reverberation modeling workshop included cases with strong attenuation to the forward propagation caused by multiple forward scattering. The effects of this attenuation were consolidated into effective reflection coefficients using the small slope approximation and provided to workshop participants. In this talk the coherent modeling of reverberation for these strongly attenuated cases using complex modes as computed by KRAKENC is described, and the results of these coherent modal predictions are compared to results obtained by other participants using ray theory. The importance of including both the complex mode shapes and eigenvalues, as well as the coherent interactions between modes, to obtain accurate predictions is discussed.

TUESDAY AFTERNOON, 27 OCTOBER 2009 — RIO GRANDE WEST, 2:00 TO 4:00 P.M.

Session 2pAB

Animal Bioacoustics: Emotion-Related Mechanisms of Mammalian Vocalizations

Michael J. Owren, Chair
Georgia State Univ., Dept. of Psychology, P.O. Box 5010, Atlanta, GA 30302-5010

Chair's Introduction—2:00

Invited Papers

2:05

2pAB1. Emotion-related acoustic communication in bats. Sabine Schmidt (Inst. of Zoology, Univ. of Veterinary Medicine Hanover, Buenteweg 17, 30559 Hanover, Germany, sabine.schmidt@tiho-hannover.de)

Some features of emotional prosody in human speech may be rooted in mechanisms common to mammals. The role of vocal communication in social interactions was studied in bats, a highly vocal group evolutionarily remote from primates. The present paper focuses on communication during agonistic encounters in the Indian False Vampire bat. Three call types with distinct time-frequency contours occurred; aggression calls, whistles, and response calls. In a first experiment, agonistic approach situations were analyzed to assess the extent to which these call types reflected the specific part of the caller in the interaction and the intensity of the agonistic display. A frame-by-frame video analysis followed by a sound analysis revealed that call type indicated the part of the respective caller while interaction intensity was encoded in similar parameter changes across call types. The systematic change in vocal parameters with