Technical University of Denmark

The specificity of sensorimotor learning: Generalization in auditory feedback adaptation

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Session 4aAA

Architectural Acoustics: Computational Room Acoustics and Room Acoustics Measurements

David S. Woolworth, Chair
Oxford Acoustics Inc., 356 CR 102, Oxford, MS 38655

Contributed Papers

8:30

4aAA1. *In situ* estimation of acoustic impedance on the surfaces of realistic interiors: An inverse approach. Gabriel Pablo Nava, Yoichi Sato, Shinichi Sakamoto (Inst. of Industrial Sci., The Univ. of Tokyo, 4-6-1 Komaba, Meguro-ku, Tokyo 153-8505, Japan), and Yosuke Yasuda (The Univ. of Tokyo, Kashiwashi, Chiba 277-8563, Japan)

In situ measurement of acoustic impedance is traditionally performed using pairs of microphones located close to the test surface. However, this method becomes troublesome if inaccessible complex-shaped surfaces, such as those in a real room, are considered. To overcome this problem, a method to estimate the normal acoustic impedance on the interior surfaces of a room is proposed. As input data, the algorithm takes: (1) The 3-D shape of the room; (2) the strength of the sound source; and (3) a set of sound pressures measured at random locations in the interior sound field. The estimation of the acoustic impedance at each surface is achieved via the solution of an inverse problem, which arises from the boundary-element method applied to the discretized interior boundaries of the room. Unfortunately, the solutions of this kind of problems are known to be unstable and sensitive to noise, due to a rank-deficient linear system. Dealing with such a system is avoided in the proposed method by formulating an iterative optimization approach, which is shown to be more robust to noise. Previous work has reported examples with numerical simulations. This paper goes further and presents results obtained with real data from experiments.

8:45

4aAA2. Acoustical renovation of rooms for music instruction in schools to improve teaching and listening conditions. Hyun Paek (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607) and Gary W. Siebein (Univ. of Florida, Gainesville, FL 32611-5702)

Several music classrooms were renovated because music instructors complained of excessive noise levels during classes. Instructors also could not hear students well and commented on harsh reflected sounds. The rooms were trapezoidal shaped with relatively low, angled gypsum board ceilings with few sound absorbing materials installed. Acoustical measurements of reverberation times, overall sound levels, and reflected sounds were used to identify the causes of the perceived problems. Renovation schemes included floating planes of sound-diffusing panels or prefabricated-diffusing panels suspended in a ceiling grid to allow students to hear each other; raising the ceiling height to reduce excessive sound levels; and strategically placing sound-absorbent materials to reduce reverberation and acoustic defects. Post-construction acoustical measurements, as well as discussions with music instructors, revealed that while both reverberation times and loudness levels decreased, the early sound energy reflected across the room increased and resulted in improved musical communication among students and instructors.

9:00

4aAA3. Perception and performance of hearing-impaired versus normal-hearing persons under noise. Lauren M. Ronsse and Lily M. Wang (Architectural Engr. Prog., Univ. of Nebraska Lincoln, Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0681, lronsse@mail.unomaha.edu)

Examining the effects of mechanical system noise on worker performance and perception for normal-hearing persons has been an area of previous research; this study extends this research area to include hearing-impaired subjects. The goal is to determine if persons with hearing impairments respond similarly to seven different mechanical system background noise conditions as persons with normal hearing. The seven noise signals vary in terms of level and spectral quality, but are all within the range of background noise conditions found in commercial offices. Performance is gauged on three types of tests (math, verbal and typing), while subjective perception is measured via a subjective questionnaire. The results of this investigation will show if significant differences are present between the two groups, and if so, they may lead to the development of different standards for noise criteria levels in spaces designed for the hearing-impaired. [Work supported by a Univ. of Nebraska Layman Award and an ASHRAE Graduate Student Grant-in-Aid.]

9:15

4aAA4. Residential wall construction transmission losses and home office productivity. Alicia J. Wagner and Lily M. Wang (Architectural Engr. Prog., Univ. of Nebraska Lincoln, Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0681, ajwagner@mail.unomaha.edu)

With advances in technology, the office worker finds herself in a changing work environment; there are now more opportunities to work from home. While recent research has addressed variations in worker productivity due to existing sound conditions within a space, home office performance has a greater dependence on isolation from exterior sound sources than conventional office work. This project investigates how sound levels produced by walls with different levels of transmission loss (TL) affect an individual's performance. Performance on math, verbal, and typing tests was assessed for sound signals played across four residential wall constructions ranging from a sound transmission class of 35 to 55. Correlations between the TLs and math, verbal, and typing productivity will be shown and used to determine if there is a minimum acceptable TL level.

9:30

4aAA5. Evaluation of accoustic comfort in classrooms of the Federal University of Parana. Andressa M. C. Ferreira and Paulo H. T. Zannin (Departamento de Engenharia Mecanica, Universidade Federal Do Parana, arqferreira@yahoo.com.br)

This research is a case study developed to evaluate and compare the classroom acoustics of two different buildings from the Federal University of Paran, built in different times: Building CP (from the 1960s) and building JB (built in 2000). Acoustical parameters have been measured in eight classrooms of the CP building, and seven classrooms of the JB building: Sound pressure level, reverberation time, and sound insulation. The methodology employed followed standard procedures for acoustical evaluation

of classrooms. Results showed that the classrooms of the CP building, from the 1960s, display satisfactory acoustical conditions, especially in their reverberation times. On the contrary, classrooms of the more recently built JB building yielded poor results with respect to reverberation times, either in empty classrooms, or in partially or even totally occupied classrooms, not meeting the values set by the standards. Thus, despite the evolution of the studies and research on room acoustics, older classrooms of the Federal University of Paran have better acoustics than more recently built classrooms.

10:00–10:30 Break

9:45

4aAA6. Use of 2-D and 3-D visualization and auralization tools to assess the acoustics of unusual shaped spaces. Joshua M. Cushner and Ryan B. Biziorek (Arup Acoust., 155 Ave. of the Americas, New York, NY 10013, joshua.cushner@arup.com)

The assessment of room acoustics commonly includes the calculation of relevant metrics by standard empirical equations or computational ray-tracing. These methods are not always sufficient for unusual shaped rooms, and can be supplemented with visualization and auralization techniques that improve our understanding of the behavior of sound in these spaces. Utilizing two- and three-dimensional room models with proprietary developed software, visualization of 360-deg particle emissions from a point source can provide useful information about sound propagation, spatial characteristics, and interactions with room boundaries. Following an iterative visualization process, design verification can be established using auralization techniques in a controlled listening environment. Such techniques facilitate better understanding of the acoustic phenomenon within the room and can help to refine acoustic solutions for room shapes or more efficient use of room surfaces for sound-absorbing finishes. Results from recent project studies show this approach to be useful to enhance communication with the architect and provides the acoustician with another useful analysis tool.

10:30

4aAA7. An exploration of virtual 3-D sound. Scott McDermott and Cheehung Henry Chu (Ctr. for Adv. Comput. Studies, The Univ. of Louisiana at Lafayette, Lafayette, LA 70503, sdm1718@cacs.louisiana.edu)

Many of the applications, virtual environments, and video games available to average computer users embrace stunning 3-D graphics and real-world visualizations. Developers spend an extraordinary amount of time and effort creating these immersive, realistic virtual environments, primarily focusing on the graphics components. Within these virtual realities, the user should be able to easily and accurately perceive the locations of sound sources, as well as the acoustic nature of the environment. However, for reasons of economy and simplicity, most developers apply readily available industry standards for generating pseudo-3-D sounds in their applications. This research explores the shortcomings of these standards and proposes an effective alternative. This project includes a number of computationally efficient, physics-based 3-D acoustics simulations, each of which will produce realistic aural reproductions. The goal is to evaluate and compare these algorithms against each other, non-3-D sound reproduction, and the current industry standards (e.g. Microsoft's DirectX's pseudo-3-D algorithm). Two hypotheses will be tested. First, users will find true, physics-based 3-D algorithms to render improved auralization reproductions compared against DirectX and/or OpenAL. Second, localization and spatialization improve with user training when using these algorithms.

10:45

4aAA8. An automated high spatial-resolution scanning system for room-acoustic scale models. Rolando de la Cruz, Gordon Rubin, and Ning Xiang (Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY 12180)

Experimental study of acoustically complex environments is a lengthy and troublesome process, in part due to the great deal of effort in acquiring data. A custom Cartesian scanning system has been developed to perform automated room impulse response measurements in scale models. High spatial-resolution measurements are performed within a planar grid and mapped to observe the spatial distribution of the sound field. Phase relationship across the scanning grid is maintained between measurements using continuous pseudo-random signals. The efficiency and repeatability of the high spatial-resolution scanning system is demonstrated in acquiring a large number of room impulse responses over a predefined Cartesian grid, while maintaining a wide bandwidth in accordance with the spatial Nyquist theorem. This paper discusses some preliminary results obtained in an eighth scale model of a coupled-volume system.

11:00

4aAA9. Prediction of sound around a corner in long corridors. William W. S. Fung, Sai Tung So, Chun Wah Leung (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong), and Kai Ming Li (Purdue Univ., West Lafayette, IN 47907-2031, mmkmli@purdue.edu)

An L-shape section is a prime consideration in the design of corridors in modern buildings. Similar to a straight corridor, the direct sound field dominates the sight-line region in the near field. However, there is no contribution from the direct sound field when the receiver locates around the corner at the shadow zone. The reflected sound field becomes more important as the receiver moves further from the bend. The present study examines the propagation of sound along a long space with an L-shape section. A numerical model using an image-source method is developed for the prediction of sound transmission along the L-shape section of a long corridor. The reverberation times are predicted by the proposed numerical model. In particular, the respective sound fields at two octave bands (500 Hz and 2,000 Hz) are obtained. The noise levels at these two octave bands are needed for the computation of the rapid speech transmission index. This information is often used to conveniently assess the speech intelligibility of a communications channel. The proposed image source model is validated by comparing with field measurements conducted in an L-shape corridor and also conducted in a scaled model in an anechoic chamber.

11:15

4aAA10. Laser Doppler vibrometer-based acoustical measurement of reflection coefficient up to 20 kHz. Gordon Rubin and Ning Xiang (Grad. Prog. in Architectural Acoust., Rensselaer Polytech. Inst., Troy, NY 12180)

The well-established two-microphone technique for determining the acoustic reflection of a material in an impedance tube has physical limitations that hinder the measurement of this quantity at high frequencies. Tube geometry, microphone spacing, and perturbation of the sound field by measurement instruments make the determination of normal incidence reflection coefficients above 10 kHz difficult, if not impossible, using microphones and the traditional impedance tube methods. In a recent work, an extended impedance tube technique utilizing a scanning laser doppler vibrometer (SLDV) was presented [Vanlanduit et al., J. Acoust. Soc. Am. **3** (2005)]. In this study, a method using a single point laser Doppler vibrometer (LDV) is used to study the limitations and merits of acousto−optic sensing in impedance tubes. Foams of various thicknesses are measured using the transfer function method in a transparent impedance tube. Results are presented that convey the potential usage of these acousto−optic techniques in the measurement of acoustic reflection properties of materials up to 20 kHz and beyond.

Session 4aBB

Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Biological Effects and Medical Applications of Stable Cavitation I

R. Glynn Holt, Chair

Boston Univ., Dept. of Aerospace and Mechanical Engineering, Boston, MA 02215

Chair's Introduction—8:35

Invited Papers

8:40

4aBB1. Stable cavitation description and characterization. Charles C. Church (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

In his now-famous work, ''Physics of Acoustic Cavitation in Liquids,'' [Physical Acoustics, Vol. 1-B, edited by W.P. Mason (Academic Press, New York, 1964), Chap. 9, pp. 57–172], Hugh Flynn coined two terms that have become commonplace in the lexicon of bubbologists, ''transient cavitation'' and ''stable cavitation.'' Although in routine use in the field of acoustic cavitation research, the original, rather precise meanings of these phrases have since been obscured. For example, ''inertial cavitation'' is now preferred owing to the perception that the former, ''transient cavitation,'' refers to motion that must inevitably result in the destruction of the bubble upon collapse, while in fact the original criterion was simply that the contraction must approximate that of a Rayleigh cavity until arrested by the internal pressure. Similarly, ''stable cavitation'' is often considered to be low-amplitude motion, but Flynn's criterion was simply that the bubble oscillate nonlinearly about its equilibrium radius. This talk will describe the original and modern meanings of these terms, discuss various cavitational phenomena, and differentiate the effects likely to be induced by each type of bubble motion. [Work supported by DAMD17-02-2-0014.]

9:00

4aBB2. Cavitation microstreaming patterns in single and multiple bubble systems. Andrew Ooi, Paul Tho (The Univ. of Melbourne, Parkville, Victoria 3010, Australia), and Richard Manasseh (CSIRO Manufacturing and Infrastructure Technol., Highett, Victoria 3190, Australia)

Cavitation microstreaming is a well-known phenomenon, yet there are very few flow visualizations or measurements of the velocity fields. In this talk, results from micro-particle image velocimetry measurements and streak photography, illustrating the flow field around a single and multiple oscillating bubbles resting on a solid boundary, will be presented. The different modes of bubble oscillations were also measured in terms of the variation in the radius of the bubble and the movement of the bubbles centroid so that the streaming flow field could be accurately related to the bubble motion. Thus, the resulting flow field can be correlated with the different vibration (volumetric, translating, and orbiting) modes. The flow field resulting from these oscillation modes contains closed streamlines, representing vortical regions in the vicinity of the bubble. Despite some inconsistencies, there is general agreement between these streaming patterns and those found in existing theoretical models. In addition, shape mode oscillations of single bubbles, as well as several different cases of multiple bubbles simultaneously oscillating with the same frequency and phase, were also investigated and show a rich variety of streaming patterns.

9:20

4aBB3. Subharmonic generation mechanisms in stable cavitation. R. Glynn Holt (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215) and Joachim Holzfuss (Technische Universität Darmstadt, 64289 Darmstadt, Germany)

The observation of subharmonic emissions during an ultrasound experiment has often been taken as the sine qua non of stable cavitation. While the appearance of subharmonics is very likely the indication of the presence of bubbles, it leads to a thorny question: What do subharmonic emissions tell us about the nature of the cavitation in the experiment? The answer to that question will naturally be of pragmatic importance to the experimenter seeking to enhance, inhibit, or otherwise control the resulting bioeffect. In this talk, we review the physical mechanisms for bubble generation of subharmonics, giving where possible, examples of each mechanism and eliciting the consequences for biomedical experiments.

9:40

4aBB4. Ultrasound contrast agent microbubbles: Stable and transient subharmonic emissions. Leonardo Masotti, Elena Biagi, Luca Breschi, and Enrico Vannacci (Ultrasonic and Non Destructive Testing Lab., Electrons and Telecommunications Dept. via S Marta 3, 50139 Firense, Italy)

The paper describes a method able to perform an ultrasonic study of a single UCA bubble immersed in a tank filled with a liquid (water) by means of ultrasonic equipment able to insonify and acquire RF echo signals coming from an isolated bubble with a high frame rate. The work was focused on the study of subharmonic emission from an isolated bubble of SonoVue. In particular, the acoustic pressure threshold for a subharmonic stable emission was evaluated for a subset of 50 microbubbles at 3.3 MHz and at 5 MHz of insonation frequencies. An unexpected second pressure threshold, which caused the standstill of the subharmonic emission, was detected at 3.3 MHz and 5 MHz. A transient subharmonic emission, which is hypothesized as being related to the formation of new free gas bubbles, was detected during the ultrasound-induced destruction of microbubbles. Several sequences of RF echo signals and the related spectra, acquired from a set of 400 isolated microbubbles, is investigated at different acoustic driving pressures and excitation frequencies. Specific bubble behaviors concerning the stable subharmonic emission, the inhibition of subharmonic generation, and the transient subharmonic emission were detected during experimentation and discussed in this paper.

10:00–10:15 Break

Contributed Papers

10:15

4aBB5. Monitoring and simulating stable cavitation during ultrasound-enhanced thrombolysis. Saurabh Datta, Azzdine Y. Ammi (Dept. of Biomed. Eng., Univ. of Cincinnati, 231 Albert Sabin Way, MSB 6152, Cincinnati, OH 45267), Constantin C. Coussios (Univ. of Oxford, Oxford, UK), and Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH 45267-0586)

The presence of stable cavitation has been shown to be highly correlated with thrombolytic efficacy for recombinant tissue plasminogen activator (rt-PA) mediated thrombolysis. A commercial contrast agent, Definity, was used with 120 kHz pulsed ultrasound to nucleate, promote, and sustain stable cavitation. The effect of stable cavitation on increased penetration of rt-PA into the clots is discussed. Also, the possibility of lowering the rt-PA dose using sustained stable cavitation adjuvant to thrombolytics is presented. To understand the bubble dynamics involved, the bubble response was studied using the Keller-Miksis model and stable and inertial cavitation thresholds were studied as a function of bubble radius. The largest mass loss (26.2%) was observed for clots treated with 120 kHz ultrasound (0.32 MPa peak-to-peak pressure amplitude, 80% duty cycle), rt-PA (96 μg/ml) and stable cavitation nucleated by Definity. A comparable mass loss of 22% was observed at a much lower concentration of 11 μg/ml in the presence of stable cavitation. In addition, the simulation results provide insight into the nuclei sizes relevant for this therapeutic application and possible mechanisms involved.

10:30

4aBB6. 120 kHz pulsed ultrasound enhanced thrombolysis with tissue plasminogen activator-loaded echogenic liposomes. Jason M. Meunier (Dept. of Emergency Medicine, The Univ. of Cincinnati, 231 Albert Sabin Way, PO Box 670769, Cincinnati, OH 45267-0769, meuniejn@uc.edu), Denise A. B. Smith, Christy K. Holland (The Univ. of Cincinnati, Cincinnati, OH 45221-0761), Shao-Ling Huang, David D. McPherson (The Univ. of Texas Health Sci. Ctr., Houston, TX 77030), and George J. Shaw (The Univ. of Cincinnati, Cincinnati, OH 45267-0769)

Echogenic liposomes (ELIP), phospholipid vesicles filled with gas and fluid, can be manufactured to incorporate the thrombolytic drug tissue plasminogen activator (tPA). Real-time thrombolysis of blood clots exposed to tPA-incorporating ELIP (t-ELIP) was monitored using video microscopy with an inverted optical microscope. Human whole blood clots on silk sutures were exposed to tPA alone (3.15 micrograms/ml), t-ELIP alone (3.15 micrograms/ml), t-ELIP and 120 kHz ultrasound (0.18 MPa peak negative pressure, 1.667 kHz pulse repetition frequency, 50% duty cycle), or tPA and ultrasound, for 30 min. The extent of thrombolysis was determined by assessing clot width as a function of time, using a time-lapse microscopic imaging technique. The average percent change in clot width at 30 min for clots treated with t-ELIP alone exceeded tPA alone (22.8% vs. 15.6%, respectively). Thrombolytic efficacy was similar for either tPA or t-ELIP exposed to 120 kHz ultrasound. Thus, the thrombolytic drug could be effectively released by exposure to 120 kHz ultrasound. [This work was supported by The Distinguished Chair for Clinical Research in Emergency Medicine Foundation Award, K02-NS056253, NIH 1RO1 NS047603, and NIH 1R01 HL074002.]

10:45

4aBB7. Effect of 810 kHz cw ultrasound on bacterial biofilms. Kofi Anguah, Roby Velez, Amy C. Vollmer, and E. Carr Everbach (Eng. and Biol. Dept., Swarthmore College, Swarthmore, PA 19081, ceverba1@swarthmore.edu)

Biofilms of florescent E. coli bacteria were grown in protein-rich media on coverslips that served as the floor of an ultrasound exposure chamber. Two opposite walls of the chamber were strips of lead zirconate titanate (PZT-4) driven at their resonance frequency of 810 kHz in continuous wave mode, thereby setting up acoustic standing waves within the 12 mm by 16 mm by 0.5 mm chamber. A convolution confocal florescence microscope was used to visualize the biofilm before, during, and after 10-minute exposures at various acoustic pressure amplitudes, with and without Optison microbubbles present. Quantifiable changes to the biofilm structure suggest that stable cavitation is a mechanism of interaction that may provide a method of enhancing antibiotic action in medical applications. [Work supported by HHMI grant #52005202.]

11:00

4aBB8. Simulation of diagnostic ultrasound image pulse sequences in cavitation bioeffects research. Douglas L. Miller, Chunyan Dou (Dept. of Radiol., Univ. of Michigan, Ann Arbor, MI 48109), and Roger C. Wiggins (Univ. of Michigan, Ann Arbor, MI 48109)

Research on cavitational bioeffects of diagnostic ultrasound typically involves a diagnostic scanner as the exposure source. However, this can limit the ranges of exposure parameters for experimentation. A single-element laboratory exposure system was used to simulate 1.5 MHz contrast-aided diagnostic ultrasound of rat kidney. Amplitude modulation of pulses (1.7 microsecond duration, 0.43 ms repetition period and 2.3 MPa amplitude) with a Gaussian envelope of 4.6 ms duration reproduced the image pulse sequence (IPS) of 1 s intermittent beam scanning. A 10 microliter/kg/min IV dose of Definity contrast agent was given during 1–5 min exposures. Glomerular capillary hemorrhage was assessed by histology. A fixed beam induced hemorrhage within the exposed area comparable to the diagnostic imager, but the use of 5 exposures at closely spaced positions more faithfully reproduced the effect produced within a scan plane. IPSs consisting of only one or two pulses approximated the effect of the simulated IPS. Use of a 100 ms duration envelope confirmed a previous observation that slow Doppler imaging mitigated the bioeffect. Thus, relatively simple laboratory exposure systems can simulate diagnostic ultrasound scanning in cavitation bioeffects research, allowing exploration of parameter ranges beyond those available on present clinical systems. [Work supported by NIH grant EB00338.]

11:15

4aBB9. Growth of gas bubbles by rectified diffusion at high supersaturation levels. Yurii A. Ilinskii, Preston S. Wilson, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

There is continued interest in possible effects of under-water sound on the growth of gas bubbles in capillaries of marine mammals and humans. [Houser et al., J. Theor. Biol. **213**, 183 (2001)] showed that following a series of dives, gas supersaturation in capillaries of marine mammals can reach 300%. For such high supersaturation levels, standard mathematical

models of both static and rectified diffusion underestimate the rate of bubble growth by 10%–20%. The discrepancy is demonstrated by comparing predictions based on existing mathematical models with direct numerical solutions of the differential equations for gas diffusion in the liquid and heat conditions in the bubble. The Rayleigh-Plesset equation is used to describe the bubble dynamics. Underestimation of bubble growth by existing mathematical models is due to the underlying assumption that the gas concentration in the liquid is given by its equilibrium state for a bubble of constant radius. This assumption is violated when high supersaturation causes the bubble to grow too fast in relation to the time scale associated with diffusion. Rapid bubble growth results in an increased concentration gradient at the bubble wall, and therefore a growth rate in excess of predictions based on quasistatic gas concentrations. [Work supported by ONR.]

FRIDAY MORNING, 30 NOVEMBER 2007 NAPOLEON B1, 9:00 TO 10:00 A.M.

Session 4aEA

Engineering Acoustics: Topics in Engineering Acoustics

Daniel M. Warren, Chair
Knowles Electronics, 1151 Maplewood Drive, Itasca, IL 60134

Contributed Papers

9:00

4aEA1. Effect of leakage in Helmholtz resonators. Iljae Lee, Ahmet Selamet, and H. S. Kim (The Ohio State Univ., 930 Kinnear Rd., Columbus, OH 43212, Selamet.1@osu.edu), and Norman T. Huff (Owens Corning, Novi, MI 48377)

The effect of leakage in Helmholtz resonators has been experimentally and numerically investigated. Transmission loss of a Helmholtz resonator having a gap between the cavity and main duct is measured using an impedance tube setup. The effect of leakage on the transmission loss is examined using different amounts of gap openings. Experimental results are then compared with the predictions from the boundary-element method. The study shows that the leakage increases the resonance frequency substantially and widens the transmission loss. Hence, the leakage needs to be taken into account for accurate predictions of Helmholtz resonators.

9:15

4aEA2. Optimization of baffle configurations to prevent aeroacoustic instabilities in heat exchangers: Preliminary experiments. A. Miguel Moreira (College of Technol., Setubal Polytechnic Inst., Campus do Instituto Politcnico de Setubal, Estefanilha 2910-76, Portugal, mmoreira@est.ips.pt), Jose Antunes, Vincent Debut, Martins Paulino (Inst. of Nuclear Technol., Portugal), and Heitor Pina (Tech. Inst., 1049-001 Lisboa, Portugal)

Gas heat exchangers are prone to aeroacoustic instabilities, which often lead to severe noise levels, structural vibrations, and fatigue. Actually, this problem is solved by placing rigid baffles inside the container, which modify the acoustic modal fields and eventually inhibit the instability. For realistic industrial components using a restricted number of acoustical baffles, their optimal location is a challenging problem, as trial and error experimentation is often a costly and frustrating procedure. Recently, some strategies were proposed for the optimal location of a single baffle in a typical re-heater from a power station boiler, based on simulated annealing as well as genetic algorithm approaches. In this paper and using the above-mentioned optimization strategies, a more complex case of the problem—the optimal location of a given number (two or more) of baffles in a typical re-heater, was addressed. Some preliminary experiments were performed and compared with the simulation results. From the discussion, improvements in the developed optimization functional were proposed. [This work has been endorsed by the Portuguese FCT and POCI 2010, with funding participation through the EC programme FEDER (Project No. POCI/EME/5728/2004).]

9:30

4aEA3. Digital reconstruction of sound using array of microspeakers. Kassiani Kotsidou and Charles Thompson (ECE Dept., Univ. of Massachusetts Lowell, Lowell, MA 01854)

This work examines the digital reconstruction of sound using the method of matched asymptotic expansions. We will study an array of microspeakers and in particular, the role the cut off frequency and the damping ratio have in the successful reconstruction of sound. The transducers are fed by pulses whose duty cycle will be studied and its ramifications will be reported. We will consider how the coupling between the transducers affects the performance of the overall system.

9:45

4aEA4. Personal sound system design for mobile phone, monitor, and television set; Feasibility study. Chan-Hui Lee, Ji-Ho Chang, Jin Young Park, and Yang-Hann Kim (Ctr. for Noise and Vib. Cotrol, KAIST, 373-1 Guseong-dong, Yuseong-gu, Daejeon 305-701, Republic of Korea, chance99@kaist.ac.kr)

Having a personal sound system that does not need to use an earphone or any wire connected microphone would have great interest and potential impact on the associated industries. A device that uses nonlinear characteristics of high intensity ultrasonic sound, such as a parametric array, enables us to localize the sound on any position we want. However, it requires a fairly significant amount of power consumption. Therefore, the mobile phone or note book computer system cannot accept this type of device. Instead, we have attempted to use a line array speaker system to localize the sound in our listening zone. It is known that sound pressure or intensity can be well focused in the zone of interest: For example, see [J.-W. Choi, Y.-H. Kim, J. Acoust. Soc. Am. **111** (2002)]. We applied the method to see the possibility to make a good bright zone around the human head or ears. Depending on the size of the zone and array parameters, for example, array size, speaker spacing, frequency characteristics of the speakers, the quality of the bright zone is changed. The quality means the ratio of sound energy between the bright and dark zone, regarded as acoustic contrast, analogous with what we use for an optical device.

Session 4aED

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

Uwe J. Hansen, Chair
Indiana State Univ., Physics Dept., Terre Haute, IN 47809

Approximately 20 acoustics experiments will be set up, ranging in complexity from simple resonance on a string to ultrasonic levitation. Students from local area high schools will perform the experiments with help from ASA scientists and students. Regular ASA conference participants are welcome to the session, as long as they do not interfere with student experimentation.

FRIDAY MORNING, 30 NOVEMBER 2007 GRAND COUTEAU, 8:00 TO 11:15 A.M.

Session 4aMU

Musical Acoustics: Musical Acoustics and Instrumental Acoustics

Scott McDermott, Chair
1431 St. Claude Avenue, New Orleans, LA 70116

Contributed Papers

8:00

4aMU1. A theoretical dissipative analysis of optimized multimodal resonators coupled to room acoustics. José Antunes (Instituto Tecnologico Nuclear, Appl. Dynam. Lab., Estrada Nacional 10, 2686 Sacavem, Portugal, jantunes@itn.pl), Octávio Inácio (ESMAE-Inst. Politecnodo Porto, Porto, Portugal), François Axisa (Com à l'Energie Atomique, Gif-sur-yvette, France), and Vincent Debut (Inst. Tech. Nuclear, Sacavem, Portugal)

Helmholtz resonators are often applied for the sound equalization of control rooms in recording studios, through adequate leveling of the low frequency acoustic modal room responses. The number of controlled acoustic modes depends on the central frequency and damping of resonators, as well as on the modal density of the controlled system within the resonator's frequency range. In a recent paper, we proposed to improve the efficiency of such devices by, instead of using basic Helmholtz resonators, develop shape optimized multimodal resonators in order to cope with a larger number of intrusive room modes. In spite of the promising results thus obtained, further work is needed to demonstrate the feasibility of such an approach. The present paper is a further step in that direction by analyzing the acoustics of the fully coupled room/resonators system including dissapative effects. More specifically, using an acoustical substructure computational approach, we theoretically derive the coupled modes of rooms fitted with several optimized multimode resonators.

8:15

4aMU2. Auditory resolution in auditory virtual environments. Georgios Marentakis, Nils Peters, and Stephen McAdams (CIRMMT, McGill Univ., 555 Sherbrooke St. W., Montreal, QC, Canada H3A 1E3)

Minimum audible angles (MAA) were estimated for listeners in the sweet spot of four- and eight-loudspeaker arrays in the studio as a function of angle of incidence (0, 60, 90) and source position within the array (on the loudspeaker, midway between or one-third of the way between loudspeakers). Vector-based amplitude panning (VBAP) was used with a 250-ms broadband stimulus. MAAs increased with angle of incidence in direct comparison to real sources. For frontal incidence, MAA was 7.4, 2.9 for the four- and eight-speaker VBAP system (compared to 1 with real sources). A systematic effect of source position within the speaker array was found with MAAs being smallest for sources halfway between the speakers. The experiment was replicated in a concert hall for listeners at eight positions away from the sweet spot for eight- and 16-loudspeaker arrays and for angle of incidence of 0, 45, and 90 relative to a listener in the sweet spot, facing forward. Participants were able to perceive change in accordance to their relative angle to the stimulus. For frontal incidence, MAA at threshold was 3.24 and 2.0 for VBAP for eight and 16 speakers, respectively. The results are investigated using binaural localization models.

8:30

4aMU3. The enigma of Vitruvian resonating vases and the relevance of the concept for today. Rob Godman (Univ. of Hertfordshire, College Ln., Hatfield, Herts AL10, UK)

Anyone who has visited a Greek or Roman theatre cannot fail to be impressed by the overall clarity of sound without any form of enhancement. The seats arranged in curved rows around the circular orchestra form large horizontal reflecting surfaces. This ensures that the path of the sound waves travel from the source (the actor or singer) to each of the listeners in a direct path (i.e., without reflection). Vitruvius, however, claimed further enhancements could be made. "In theatres, also, are copper vases and these are placed in chambers under the rows of seats in accordance with mathematical reckoning. The Greeks call them Echeia. The differences of the sounds which arise are combined into musical symphonies or concords: The circle of seats being divided into fourths and fifths and the octave." (Vitruvius, on Architecture, Book I,—on training of architects, Loeb). With the aid of modern-effect systems that attempt to mimic real and imaginary spaces, it may be difficult to imagine the importance of the Vitruvian idea. This paper demonstrates a number of digital reconstructions, examines the issue of intent, and questions an arguably more complex issue, that of a potential fusion between archaeology, science, and music.

8:45

4aMU4. Quantal computation of tonality in music. Alpar Sevgen (Dept. of Phys., Bogazici Univ., Bebek 34342, Istanbul, Turkey)

Each independent scale interval structure with N semitones and M notes contains a multiplet of N scales (for example, the major scale has the interval structure 2212221; and its twelve multiplet members are: C-major, D flat-major, . . . , B-major scales). Operators that step through these multiplet members (by sharping or flatting the notes of the multiplet members) are identical with those of N-level finite state quantum mechanics. It is possible to construct orthonormal basis states which represent the multiplet members of a given interval structure. A musical score can then be expressed as a mixture of various tonalities, using the density matrix formalism of quantum mechanics. Several examples help illustrate this quantitative approach.

9:00

4aMU5. Time series analysis of the Kurdish long necked lute, tanbour. Hedayat Alghassi (Dept. of Elec. and Comput. Eng., Univ. of British Columbia, Vancouver, BC, V6T 1Z4, Canada), Sohrab Ferdowsi, Roozbeh Alghassi (Tehran Univ., Tehran, Iran), and Babak Khademi (Stanford Univ., Stanford, CA 94305)

There are a variety of long necked, plucked string lutes in the East with the universal name tanbour/tanbur. This research concentrates on the three-stringed fretted tanbour of Yarsan Kurds of Gouran, which is the prevailing derivative of ancient tanbour originated from Persia. Contrary to most of Western stringed instruments, the vibration in tanbour is not limited to soundboard, especially at higher frequencies. This fact, along with string tension modulation, creates a nonlinear vibration of the body and neck. Therefore, the single channel sound pickup and linear modeling is not representing tanbour's complex sound field. Utilizing an array of four microphones, three spreading in front of soundboard and one in back of body, the system transfer function for tanbour's main plectrum was derived via suitably optimizing a nonlinear-auto-regressive-exogenous model with the envelope of signals extracted from five tanbours with different known body sizes. Besides neck length and body width, length, and depth, the location of bridge is incorporated in the model, since it has a major role in the vibration, quality, and final setup of tanbour. Employing this model and tanbour's physical sizes, one can estimate the optimal bridge location on the sound board to achieve finest vibration in any tanbour.

9:15

4aMU6. Implementation of non-stringed instruments without bodies. Sangjin Cho and Uipil Chong (Univ. of Ulsan, San29, Mugeo-Dong, Nam-Gu, Ulsan, Korea)

This paper describes the implementation of the non-stringed instruments using the TMS320F2812 DSP, which is possible to produce the sounds of musical notes. This basic concept of the instrument comes from the non-stringed instruments, which is not necessary to tune the string of the instruments before starting the play. In this paper, this system chose the guitar as an example for implementation. The system does not need a guitar body, but the pickup part is needed. The pickup of the system plays an important part in representing the up/down strokes and arpeggios. Therefore, the proposed system consists of three parts: Chord glove, pickup, and processor. Mapping between guitar chord and the glove takes place on the values of the force sensing resister and FLEX sensors. The extension of this system can be applied to stringed instruments in the future. [Work supported by KOSEF, R01-2005-000-10671-0.]

9:30

4aMU7. Experimental investigation of the effects of extra mass on a piano soundboard's vibration property. Hao Xing (Inst. of Acoust., Chinese Acad. of Sci., 21 Bei Si Huan Xi St., Beijing 100080, P. R. China), Bonan Zhao, and Haiyan Zhao (Tsinghua Univ., Beijing 100084, P. R. China)

A soundboard, as an important resonance component of a piano has been continuously studied in the past several decades. It is known that the variation of mass distribution on a soundboard could affect its vibration properties and the sound quality of a piano. However, those effects have been seldom investigated until now. In this paper, an experiment is carried out to study the effects of additional mass-blocks on a soundboard. Small pieces of metal blocks are pasted on several different locations on a soundboard, and a laser vibrometer is used to extract its mode shapes and spectrum under the excitation of an acoustic source with frequency ranges from 10 Hz to 1 kHz. The results show a notable difference of spectrum between mass-added cases and original in low frequencies, but little difference in high frequencies. It could be concluded that additional mass-block could greatly affect the energy of the low-mode of a soundboard.

9:45

4aMU8. Reed vibration, pressure, and airflow in Western free-reed instruments. Edward L. Toussaint (Lawrence Univ., Appleton, WI 54912, edward.l.toussaint@lawrence.edu) and James P. Cottingham (Coe College, Cedar Rapids, IA 52402)

Western free reed instruments, including the reed organ, harmonium, harmonica, and accordion, use asymmetric free reeds that are able to maintain sustained oscillation in the absence of a pipe resonator. Studies of the motion of air-driven American reed organ reeds and accordion reeds have been made that yield the volume air flow through the reed opening as a function of time. Reed vibration waveforms are obtained using a variable impedance transducer, and corresponding sound pressure waveforms are obtained from a probe microphone at positions near the reed opening on both sides of the reed tongue. The airflow waveform is obtained by integrating the pressure waveform and using a calculated area function and is calibrated using the measured average and minimum airflow rates. The results are compared with earlier preliminary studies as well as recent theoretical work. For pressure and airflow waveforms among different types of reeds or for the same reed at different blowing pressures, differences can be understood in terms of the configuration of the reed and reed frame system, as well as the amplitude of reed vibration. [Research supported by NSF REU Grant No. PHY-0354058.]

10:00–10:15 Break

10:15

4aMU9. Empirical physical modeling based music synthesis and representation. Xiaoxiao Dong, Mark Sterling, and Mark Bocko (Univ. of Rochester, 405 Comput. Study Bldg., 500 Joseph C. Wilson Blvd., Rochester, NY 14627, xidong@ece.rochester.edu)

We describe a method in which empirically-based musical instrument physical models are employed both to synthesize musical sounds and to form the basis of a compact representation of mono-timbral musical sound. Results are presented for a B^b clarinet. The physical model incorporates measured acoustic impedance spectra of a clarinet air column for all playable notes. Low bandwidth control parameter time histories, serving as inputs to the physical model, are inferred from audio recordings of actual clarinet music. The control parameters represent the fingerings, the blowing pressure of the player, and the mouthpiece clamping pressure of the player's embouchure. It is shown that, given an appropriate physical model, the control parameter time histories can serve as a highly compact representation (compression by a factor of several hundred) of a source recording. To serve as an assessment tool, we also describe a distance metric emphasizing timbral aspects of musical audio. The metric provides a method to quantitatively evaluate the parameter extraction routines, to

test the effect of minor alterations to the model, and also to compare the present physical modeling approach with other forms of musical sound synthesis. The framework described may be extended to other musical instruments and instrument families.

10:30

4aMU10. The influence of the acoustic feedback on the fluid-structure interaction within single-reed mouthpieces: A numerical investigation. Andrey Ricardo da Silva and Gary Scavone (Computational Acoust. Modeling Lab. and CIRMMT, McGill Univ., 555 Sherbrooke St. W., Montreal, Canada H3A 1E3, andrey.dasilva@mail.mcgill.ca)

The aim of this article is to complement a previous numerical study conducted to investigate the flow behavior in single-reed woodwind instruments during dynamic regimes with a moving reed [J. Acoust. Soc. Am. **120**, 3362 (2006)]. The code uses a fully-coupled scheme based on the lattice Boltzmann method and on a finite-difference scheme to represent the interaction between the reed and the fluid field. The previous study had suggested that, during a dynamic regime and in the absence of acoustic feedback, the flow behavior diverges significantly from what is predicted by the current quasi-stationary models based on the Bernoulli obstruction theory. The present work provides a further investigation on the same topic by taking into account the influence of the acoustic feedback from the bores open end and its interaction with the mean flow within the mouthpiece-reed system. This is achieved by coupling the previous mouthpiece-reed model to a digital waveguide whose reflection function is described by a low-order digital filter to represent the open-end boundary condition. [The first author would like to thank CAPES for supporting his doctoral research.]

10:45

4aMU11. Vibration modes of the snare drum batter head. Barry Larkin (Dept. of Music, Iowa State Univ., Music Hall, Ames, IA 50011) and Andrew Morrison (Illinois Wesleyan Univ., Bloomington, IL 61701)

Percussionists have always had to contend with an undesirable ringing sound while performing on the snare drum. It is usually referred to as the edge ring. A common method to eliminate this sound comes from placing some type of dampening material on the edge of the drum head. Manufacturers of drums have provided many ways of dealing with this problem, including internal dampening devices, customized drums heads, or material designed to be placed on the drum head. Using electronic television holography, was revealed the source of this ring to be the third mode of vibration that produces a pitch approximately one octave and a half-step above the fundamental. During this presentation, holographic images from the fundamental to the eighteenth mode will be displayed. Methods to dampen this mode will be displayed and demonstrated as part of this paper.

11:00

4aMU12. Acoustics of the mridangam: Study of a new design of a South Indian drum. Rohan Krishnamurthy (Kalamazoo College, Kalamazoo, MI 49006, rohan_krishnamurthy@rohanrhythm.com), Ian K. Hempe, and James P. Cottingham (Coe College, Cedar Rapids, IA 52402)

The acoustical properties of the ancient Carnatic drum, the mridangam, have been studied, including instruments of traditional design as well as instruments with a newly designed mounting method for the drumheads. The mridangam is comprised of three primary parts: The tonal head (valanthalai), the bass head (thoppi), and the central shell (katta), to which the two heads are traditionally fastened by leather rope. The new design is proposed as a convenient and practical way to mount and tune the drumheads. Measurements were made on drums with the drumheads mounted in the traditional way, and then repeated with the heads remounted using the new design. For measurements of the modes and mode frequencies of the drumheads, the modes were excited by a mechanical vibrator and vibration of the drumhead was measured using a laser vibrometer system. The frequencies of the first few modes of the tonal head were found, as expected, to be tuned to be approximately harmonic. Measurements of drumhead vibration and sound spectra were also made when the drumhead was excited by a skilled player using standard strokes. Some effects of coupling between the two drumheads were studied. [Work supported by the Kalamazoo College HHMI Undergraduate Research Program.]

Session 4aNS

Noise and Animal Bioacoustics: Advances in Measurement of Noise and Noise Effects on Animals and Humans in the Environment I

Ann E. Bowles, Cochair
Hubbs Sea World Research Inst., 2595 Ingraham St., San Diego, CA 92109

Brigitte Schulte-Fortkamp, Cochair
Technical Univ. Berlin, Inst. of Fluid Mechanics and Engineering, Secr TA 7, 10587 Berlin, Germany

Chair's Introduction—8:00

Invited Papers

8:05

4aNS1. Animals, ordinary humans, and scientific experts: Noise(s) or sounds? How semantics may help understanding animal behavior and human cognition as well. Daniele S. Dubois (LCPE/LAM/IJLRDA, 11 rue de Lourmel, 75015 Paris, France, ddubois@ccr.jussieu.fr)

If an anthropomorphic conception of animal behavior is inescapable, cross cultural studies, situated cognition, as well as research in ethology, have recently pointed that contemporary cognitive models of information processing that pretend to be universal of any living system, could be misleading as relying on a specific conception of cognition. Actually, such models can account for the cognition of experts trained in an analytic processing of the world, grounded in contemporary scientific (physical) knowledge, but not for the ways ordinary humans perceive and react to environmental noise for behaving and acting in the world. In every day life, humans process multimodal incoming stimulations in a holistic manner that gives meaning to their ways of being. Within this frame, acoustic stimulations are not processed per se, but as cues pointing to categorical knowledge already stored in memory and structured according to adaptation purposes. Such a situated model of cognition has been fruitfully integrated in soundscape research and in environmental acoustics. It now challenges the traditional psychophysics paradigms in developing an ecological exploration of categorization for sounds. We would like to present such methodological issues, expecting to provide new insights for human as well as for animal behavior research.

8:30

4aNS2. Noise and noise effects on humans and its meaning for measurement decisions. Brigitte Schulte-Fortkamp (TU-Berlin ISTA, Einsteinufer 25, D-10587 Berlin, Germany, brigitte.schulte-fortkamp@tu-berlin.de)

To discuss advances in measurements of noise and noise effects on animals and humans is challenging. Usually, it is physics that dominates the approach in the studies about noise and noise effects on humans; indicators in use are based on A-weighted sound pressure levels. By extension, psycho-acoustics has contributed to understanding the limitations of the A-weighted sound level in dB as a metric criterion or a criterion for measurements. In addition, noise annoyance appears to be differently perceived, expressed, and experienced in different places. For this reason, supplementing the acoustics and psycho-acoustics approaches by a more qualitative approach, as in soundscape research, seems to be appropriate in order to consider more the perception, the way people experience noise, and relate this to their well-being. Research shows that perceived and expressed annoyance are explained not only by acoustics, but also by other factors, and their relationships can explain the links between populations and their environment. Therefore, the way people react and evaluate noise as the real experts has to be taken into consideration with respect to conceptualization of measurement approaches and measures. The diverse approaches will be introduced with respect to decisions for measurements in defined environments.

8:55

4aNS3. Documenting noise exposures for wildlife studies: Review of large studies over the past 18 years. Robert Kull (Parsons, 5800 Lake Wright Dr., Ste. 101, Norfolk, VA 23502, bob.kull@parsons.com)

A couple of decades ago, researchers recognized the importance of improving understanding of noise exposure to wildlife for bioacoustic studies. Due to the differences in the ecology of species of concern, various approaches took place over these past 20 years to address this issue. The metrics used to determine effects and how the metrics were applied to observed behavioral and physiological responses also varied, based on several factors. This presentation will review methods used on several large projects, including big horn sheep, caribou, kit fox, Mexican spotted owls, peregrine falcons, and red cockaded woodpeckers. This review may help provide the direction for future bioacoustic studies.

9:20

4aNS4. The design of experimental studies of noise impacts on wildlife: Issues and outcomes. Ann E. Bowles, Donald Hunsaker II, Samuel L. Denes (Hubbs-SeaWorld Res. Inst., 2595 Ingraham St., San Diego, CA 92109), Robert Kull (Parsons Corp., Norfolk, VA 23502), Jeffrey R. Dunk (USDA Forest Service, Arcata, CA 95521), Lisa Hayward, and Samuel L. Wasser (Univ. of Washington, Seattle, WA 98195)

Historically, most studies exposing terrestrial wildlife to anthropogenic noise experimentally have failed to prove biologically-significant impact. These counter-intuitive outcomes have given rise to political debate, fueled by marginal effects or weak trends in some studies that might have proved to be significant with better sampling procedures or controls. A number of recent studies have attempted to address these weaknesses by developing adequate sampling designs and balancing exposures with respect to important environmental factors (e.g., habitat type). Examples include studies of: (1) Effects of low-flying NATO aircraft on Mexican spotted owl territory occupancy and reproduction; (2) effects of U.S. Marine Corps. helicopters on passerine reproductive success; (3) effects of operations on a U.S. Air Force bombing range on small mammal abundance; (4) impact of off-highway vehicles on reproduction of northern goshawks in the Plumas National Forest; and (5) impact of Enduro motorcycle races on northern spotted owl physiology in the Mendocino National Forest. To date, preliminary outcomes of the studies suggest that more sophisticated models of effect should be developed, and additional research should focus on long-term cumulative impact.

9:45

4aNS5. Response of gopher tortoises to military training operations: Preliminary data. David Delaney, Larry Pater (USACERL, P.O. Box 9005, Champaign, IL 61826), Tom Radzio, Joseph Hackler, Andrew Walde (ITS, Victorville, CA 92395), and Matthew Hinderliter (The Nature Conservancy, Hattiesburg, MS 39407)

The Department of Defense (DOD) needs defensible data to understand how gopher tortoises utilize military lands to address the question of potential military effects on tortoise populations, while also maintaining its ability to continue to train to standard. The objective of the research is to examine if there are differences in tortoise activity and behavior due to variation in military training activity. Automated radio-telemetry equipment is being used to monitor the activity patterns of gopher tortoises on Camp Shelby, MS. The automated radio-telemetry equipment enables one person to monitor the activity of many transmittered tortoises continuously over an extended time, regardless of weather, light level or terrain. Video surveillance and sound recording equipment are also being used to monitor tortoise behavior in proximity to burrows and quantify and characterize potential anthropogenic disturbances within study locations. The goal of this study is to benefit the recovery and management of gopher tortoise populations on DoD and non-DoD lands by providing natural resource managers detailed temporal and spatial data on gopher tortoise activity, behavior, and movement patterns under varying conditions. Funding was provided by the Engineering Research and Development Center. Preliminary examples of field data and database tables will be presented.

10:10–10:25 Break

10:25

4aNS6. The effects of helicopter noise on the reproductive success of the coastal California gnatcatcher. Don Hunsaker II (Hubbs-SeaWorld Res. Inst., 2595 Ingraham St., San Diego, CA 92109, dhunsaker@hswri.org), Jacque Rice (US Naval Facilities Eng. Command, San Diego, CA 92132), and John Kern (Kern Statistical Services, Sauk Rapids, MN 56379)

Our laboratory conducted a five-year study on the potential effects of helicopter noise on the reproductive success of the coastal California gnatcatcher (*Polioptila californica californica*) on Marine Corps Air Station Miramar (MCAS Miramar) in Southern California. Seven-hundred twenty-one nests were monitored for reproductive success, predation, noise levels, and habitat quality. An array of Larson-Davis sound level meters was used to monitor habitat on MCAS Miramar for a total of 6,176 days during 620 runs at 328 locations. Most sites were exposed to noise in excess of 60 dBA SPL for less than 5% of the monitoring period, but some nests experienced levels in excess of 70 dBA for more than 20% of the time. Statistical models of nest success, nest site selection, and number of fledges per pair showed that the factors best predicting reproductive success were measures of suitable nesting habitat, not noise levels. Helicopter and other noise sources did not affect the reproductive success of gnatcatchers. [Supported by the Marine Corps Air Bases Western Area and the Naval Facilities Engineering Command, SW.]

10:50

4aNS7. Effects of the sounds from an artificial oil production island on bowhead whale calling behavior. Susanna B. Blackwell, Charles R. Greene, Jr. (Greeneridge Sci., Inc., 1411 Firestone Rd., Goleta, CA 93117), Trent L. McDonald, Ryan M. Nielson, Christopher S. Nations (WEST, Inc., Cheyenne, WY 82001), W. John Richardson (LGL, Ltd., King City, Ont. L7B 1A6, Canada), and Bill Streever (BP Exploration (AK) Inc., Anchorage, AK 99519-6612)

The westward migration of bowhead whales (Balaena mysticetus) was studied during autumn 2001–2004 to examine the effects of sounds from an artificial oil production island (Northstar) on whale calling behavior. Whale calls were recorded by an array of directional autonomous seafloor acoustic recorders (DASARs) located 6.4–21.5 km northeast of Northstar in the Beaufort Sea. Simultaneously, a continuous record of sounds produced by Northstar was obtained ~450 m north of the island. More than 130,000 bowhead calls were detected, and were classified as to their type, including five types of simple calls and one complex call category. In addition, the directional capability of DASARs allowed triangulation of an estimated whale position for a majority of the calls. The primary objectives of the study were to assess any effects of Northstar sounds and environmental-physical covariates on the duration, midfrequency, frequency range, type of call, and call detection rates. The analyses showed that an increase in transient sounds from

Northstar (i.e., boats) resulted in significantly shorter calls. In addition, call detection rates were significantly higher for whales upstream of the array, and the use of complex calls increased significantly as the whales swam westward past Northstar. [Study funded by BP Exploration, Alaska.]

Contributed Papers

11:15

4aNS8. Marine wildlife behavior database for estimating environmental impacts. Kathleen J. Vigness Raposa, Adam S. Frankel, Geoff Sisson, William T. Ellison (Marine Acoust., Inc., 809 Aquidneck Ave., Middletown, RI 02842, kathleen.vigness@marineacoustics.com), and Christopher Damon (Univ. of Rhode Island, Kingston, RI 02881)

Assessing the potential effects of underwater sound on the environment is an increasing concern. In order to predict an animal's exposure from an acoustic source, it is necessary to know how it moves through the sound field. While a number of tools have been developed in recent years to estimate environmental impacts, they require an estimate of the species present at the time of the operation, their abundance and distribution, as well as their diving, movement, and acoustic behaviors. To determine whether a species might be susceptible to certain sound exposures, it is also necessary to consider the probable hearing range and sensitivity of the species. The marine wildlife behavior database (MWBD) is creating a worldwide standard for measuring and characterizing behavior. It includes geo-referenced baseline data on diving, movement, and acoustic characteristics of fishs, marine mammals, and sea turtles. The database is designed to capture the distribution of the values of a behavior in a way that is amenable to models that replicate the movement or other behaviors of an animal. The MWBD is publicly available over the Internet through a registered node on the National Spatial Data Infrastructure at http://www.edc.uri.edu/MWBD.

11:30

4aNS9. Update on exposure metrics for evaluation of effects of sound on fish. Mardi C. Hastings (Appl. Res. Lab., Penn State Univ., State College, PA 16804) and Arthur N. Popper (Univ. of Maryland, College Park, MD 20742)

Several studies [A. R. Scholik and H. Y. Yan, Hear. Res. **152**, 17–24 (2001); A. N. Popper et al., J. Acoust. Soc. Am. **117**, 3958–3971 (2005); M. E. Smith et al., J. Exp. Biol. **209**, 4193–4202 (2006)] indicate that the equal energy hypothesis does not apply when evaluating auditory effects in fish. This is not surprising, as similar results have been reported for mammals [R. P. Hamernik and W. Qiu, J. Acoust. Soc. Am. **110**, 3163–3168 (2001)]. New data show that fish recover relatively quickly from large temporary threshold shifts (TTS) after exposure to impulsive sounds, but take several days to recover from smaller amounts of TTS due to exposure to band-limited noise. Recent studies [e.g., J. J. Govoni et al., J. Aquatic Anim. Health. **15**, 111–119 (2003)] also validate that peak sound pressure is not an appropriate metric for effects on nonauditory tissues, even if the peak level is negative. Large negative pressures have been thought to be more harmful due to swim bladder expansion, but there are no data to support this hypothesis. Physical injury correlates with a cumulative energy index, as has been found in terrestrial mammals [J. H. Stuhmiller et al., J. Biomechanics **29**, 227–234 (1996)].

11:45–12:00
Panel Discussion

FRIDAY MORNING, 30 NOVEMBER 2007 BORGNE, 8:00 TO 10:30 A.M.

Session 4aPA

Physical Acoustics: Acoustics of Soils and Porous Media

Wheeler B. Howard, Cochair
National Center for Physical Acoustics, Univ. of Mississippi, University, MS 38677

Alexander Ekimov, Cochair
National Center for Physical Acoustics, Univ. of Mississippi, University, MS 38677

4a FRI. AM

Contributed Papers

8:00

4aPA1. An experimental study for the penetration of sound into a rigid porous medium. Hoyt Chang and Kai Ming Li (Ray W. Herrick Labs, School of Mech. Eng., Purdue Univ., 140 S. Intramural Dr., W. Lafayette, IN 47907-2031, mmkmli@purdue.edu)

In a recent study, a new analytic solution for computing the penetration of airborne sound into a rigid porous ground has been derived. Modeling the lower medium as a modified fluid, a simple analytic solution can be obtained that can be cast in a form analogous to the Weyl-van der Pol formula for the sound field in the upper medium. The current study is devoted to an experimental validation of the analytical model. A layer of glass beads is used in the experimental study for modeling the rigid porous ground. To test the analytical formulation, measurements of excess attenuation spectra above and within the layer of glass beads were conducted. The impedance, propagation constant, and the index of refraction of the glass bead aggregate were deduced from the excess attenuation spectra. The experimental results are compared with other methods for finding the propagation of the rigid porous medium. The proposed method suggests a new way of finding the propagation constant and the index of refraction of a rigid porous medium.

8:15

4aPA2. Refraction and attenuation of sound in a rigid porous medium. Kai Ming Li (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 140 S. Intramural Dr., West Lafayette, IN 47907-2031)

An investigation was conducted to predict the penetration of sound into a ground surface irradiated from a point source located in the air above the air/ground interface. To simplify the analysis, the outdoor ground surface is treated as a modified fluid in which its acoustical properties are characterized by using a complex density and a propagation constant of the porous medium. By modeling the porous ground as a

modified fluid, the physical phenomenon for the refraction of sound from the upper medium (air) to the lower medium (porous ground) has been explored. It has been shown from the mathematical analysis that the refracted path can be determined by Snells Law. The sound field can be formulated in an integral form for subsequent evaluations. Two approaches are used to calculate the integral, numerically predicting the total attenuation of sound in the porous medium. A numerically intensive scheme, known as the fast field formulation, and an analytical method known as the steepest-descents integration, are used to calculate the sound fields in the lower medium. Numerical results computed from both schemes agree well with each other over the practical range of interest.

8:30

4aPA3. Acoustic to seismic signatures of layered near-surface soil. Wheeler Howard (MIL-TEC, Oxford Enterprise Ctr., 9 Industrial Park Dr., Oxford, MS 38655) and Craig J. Hickey (NCPA, Univ. of Mississippi, University, MS 38677)

The near-surface soil structure, which strongly influences agricultural productivity and soil erosion, is amenable to study by acoustic to seismic techniques. These techniques utilize the coupling between airborne sound and seismic vibrations of the soil to investigate the distribution of soil's mechanical properties. Two field sites were chosen over which the acoustic to seismic coupling signature of the ground, was measured. A predictive model, assuming acoustic plane waves incident on a vertically stratified ground was developed to predict the acoustic to seismic response. In this presentation, the acoustic to seismic signatures from these sites are compared to model results using the ground truth from seismic refraction and cone penetrometer surveys as input to the model.

8:45

4aPA4. The influence of water potential, moisture, and temperature on sound speed and the hysteretic nonlinearity parameter of granular materials. Zhiqu Lu (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, University, MS 38677)

In the 153rd ASA meeting (Salt Lake City, Utah), the results of a long-term survey of a field soil by using both linear and nonlinear acoustic techniques were reported. The study showed that soil water potential is the predominant factor governing the sound speed and the hysteretic nonlinearity parameter while moisture plays a minor role. It was also found that, unlike sound speed that is insensitive to temperature, the nonlinear parameter shows strong temperature-dependent behavior. The question remains: Is this temperature effect an intrinsic property of the hysteretic nonlinearity parameter, or is it caused by the degradation of the transducers since the measurement has been running for two years? To answer this question, the influence of water potential, moisture, and temperature on sound speed and the hysteretic nonlinearity parameter of granular materials (unconsolidated sands) has been studied in the lab with temperature-controlled conditions.

9:00

4aPA5. Theoretical modeling of active seismic detection of shallow buried tunnels. Elizabeth T. Küsel, Mark Andrews, and Purnima Ratilal (Northeastern Univ., 440 Dana Res. Ctr., Boston, MA 02115, kusele@ece.neu.edu)

Low-frequency seismic waves are widely used to detect oil/gas reservoirs, and study the properties of Earth's interior. Here, we investigate the application of low-frequency active seismic waves (tens of Hz to a couple of hundred Hz) for long-range wide-area detection of shallow underground tunnels. The main objective is to image wide areas of the ground at the longest possible ranges to identify potential underground tunnel hotspots. The tunnel is modeled as a circular air-filled cylinder. The incident field at the tunnel is calculated in terms of the horizontal (u) and vertical (w) displacements by the elastic parabolic equation. The scattered field from the tunnel is modeled by application of Green's theorem, approximating the radial dependence of the scattered field by the Hankel function, and satisfying the boundary conditions. Compressional and shear waves scattered from the tunnel will be calculated to determine which wave type is most suitable for detection purposes. Scattered field levels from the tunnel will be compared with seismic ambient noise and reverberation from surrounding medium heterogeneities. Tunnel detection ranges will be quantified as a function of tunnel diameter and length, seismic wave frequency, receiving array length and orientation, and medium properties.

9:15

4aPA6. Varied amplitude nonlinear acoustic method of landmine detection. Laurent Fillinger (Stevens Inst. of Technol., Hoboken, NJ 07030, laurent.fillinger@artannlabs.com), Brad Libbey (RDECOM CERDEC NVESD, Fort Belvoir, VA 22060), Alexander Sutin, and Armen Sarvazyan (Artann Labs., West Trenton, NJ 08618)

Acoustic methods of land mine detection are being developed for nonmetal mines where conventional electromagnetic methods fail. Nonlinear acoustic methods have shown higher sensitivity than linear methods. In previous studies, the nonlinear methods, based on interaction of two frequency waves and phase inversion of two pulses, were investigated. We propose an alternative nonlinear method based on the use of wide-frequency pulses with different amplitude and subtraction of the normalized received signals. This method evaluates the nonlinear effects in a wide-frequency band similar to the phase inversion method and it detects nonlinear components that are lost in the phase inversion method. In our tests, the suggested method was combined with time reversal acoustic focusing to concentrate seismic wave energy in proximity to the mine, which increased nonlinear effects. Experiments with various types of mines were conducted at Fort Belvoir using six loudspeakers in a box for excitation in the frequency band 50–500 Hz. The nonlinear mine response measured by the varied amplitude method provides the highest mine/no mine contrast among all the experiments performed using linear and nonlinear techniques. [Work was supported by the U.S. Army RDECOM CERDEC Night Vision and Electronic Sensors Directorate.]

9:30

4aPA7. Evaluation of parametric acoustic array technology in application to the acoustic landmine and improvised explosive device (IED) detection problems. Murray S. Korman (Dept. of Phys., U. S. Naval Acad., Annapolis, MD 21402), Antal A. Sarkady (U. S. Naval Acad., Annapolis, MD 21402), and James M. Sabatier (Univ. of Mississippi, University, MS 38677)

There has been recent interest in using the parametric acoustic array for obtaining highly directional low frequency sources in acoustic landmine and IED detection. Experiments (using a commercial parametric array (0.5 m diam) located 1.5 m directly over the target and driven by a swept sine audio modulated 60 kHz tone) were performed to compare the "on" to "off" target soil surface particle velocity of an inert VS 1.6 (plastic) anti-tank landmine buried 2.5 cm deep in masonry sand in a concrete box located in the anechoic chamber facility. The laser Doppler vibrometer response (normalized to the microphone) was measured from 100–1,100 Hz using a spectrum analyzer. A SPL of 40 dB *re* 20 $\mu\, rmPa$ at 150 Hz was insufficient, where the mine was known to have its largest resonance. Resonances between 300–600 Hz also went undetected, but amplitude contrast ratios of 20 and 3 were measured at 850 and 1,050 Hz, respectively, upon signal averaging. Limitations in carrier SPL, auditory damage, saturation effects, distortion, and low efficiency need to be addressed to achieve the required 60–80 dB SPL needed near 100 Hz. Mines or IEDs buried at 15 cm require more SPL. [Work supported by ARL.]

9:45

4aPA8. Seismic excitation using a high power CO_2 laser. Brad Libbey, Jamie Krissoff, James Perea, and Gwen Newsome (Army RDECOM CERDEC NVESD, 10221 Burbeck Rd., Fort Belvoir, VA 22060)

Lasers provide one method of remotely exciting propagating waves in soil. The absorption of optical energy results in plasma formation, a dynamic change near the soil surface, and excitation of propagating waves. Experiments were performed using a high power CO_2 pulsed laser to

determine the pressure waveforms generated in desert sand and caliche clay. The laser energy ranged from 400–750 J with a pulse duration of 35 μs. The resulting pressure pulse was observed for duration of at least 400 μs and had a peak pressure of 100 kPa. The duration increased at greater depths, while the peak pressure and energy in a pulse decreased rapidly at these depths. Surface velocity reached a peak of 35 mm/s as acquired on geophones placed in proximity to the laser plasma event. This system, in combination with a laser Doppler vibrometer, has the potential to be a useful component of non-invasive mine detection equipment.

10:00

4aPA9. Directivity pattern of footstep sound at ultrasonic frequencies. Alexander Ekimov and James M. Sabatier (Univ. of Mississippi, NCPA, 1 Coliseum Dr., University, MS 38677)

The human footstep generates broadband frequency sound in air due to friction forces between the foot and the ground surface. An ultrasonic (25–26 kHz) footstep signal was measured on the ground at a distance of 17 m from the walker. Another aspect of footstep sound characterization is a directivity pattern. The sound wavelength at those frequencies corresponds to 13–14 mm, which is much less than the source size (the area of sound generation is approximately the size of a foot, 300 mm$\times$120 mm). The results of the experimental investigations are presented and discussed. An analysis of test results shows that the directivity pattern of footstep sound has inhomogeneous dependence on the azimuthal angle. A maximum value of the pattern corresponds to the direction of walking, while the minimum value is in the opposite direction. [Work supported by Department of the Army, Army Research Office Contract No. W911NF-04-1-0190.]

10:15

4aPA10. Seismic refraction tomography of a small earth dam. Craig J. Hickey (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38655) and Lakshmikanth Duddu (Univ. of Mississippi, University, MS 38655)

Over the past 50 years there were nearly 11,000 flood control dams constructed by the United States Department of Agriculture (USDA) nationwide in 2,000 watersheds (USBR, 1987). The failure of these earth dams is associated with erosion, seepage, piping, slope failures, and slides. Precursory evidence for certain failures can be identified by visual inspection; however, others are not so easily detected. Seismic imaging may provide unique and valuable information about the subsurface to assist in the evaluation of the integrity of an earth dam. In this presentation, results from a field study carried out on Drewery Lake Dam in the Upper Yocona River watershed to determine the feasibility of using seismic refraction tomography for imaging earthen dams is discussed. Attributes in the refraction tomograms are correlated with the location of the drainage pipe and the excavated surface given in the as-built plans indicating that seismic refraction tomography does have sensitivity to the internal structure of a dam. This study also suggests that the construction practices for earthen dams result in seismically uniform structures. A uniform background is very advantageous if seismic tomography is to be used to detect zones of internal weakness.

FRIDAY MORNING, 30 NOVEMBER 2007 NAPOLEON D2/D3, 8:00 A.M. TO 12:00 NOON

Session 4aPP

Psychological and Physiological Acoustics: More Potpourri (Poster Session)

Edward Goshorn, Chair
Univ. of Southern Mississippi, 118 College Dr., #5092, Hattiesburg, MS 39406-0001

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

4aPP1. Are there systematic movements of poles/zeros with sound source location for pole/zero models of acoustic directional transfer functions? Bahaa Al-Sheikh, Matin Matin (Dept. of Elec. Eng., Denver Univ., 2390 S. York St., CMK 206, Denver, CO 80208), and Daniel Tollin (UCHSC, Aurora, CO 80045, daniel.tollin@uchsc.edu)

Humans and animals locate sound sources using three main acoustical cues: Interaural time (ITD) and level (ILD) differences and monaural spectral shape cues. The cues are generated by the spatial- and frequency-dependent filtering of the propagating sound waves by the torso, head, and external ears. These acoustic transformations can be captured in measurements of the directional transfer functions (DTFs), the directional components of the head related transfer functions (HRTFs). A set of DTFs can be used to create a virtual auditory environment by presenting over headphones arbitrary sounds filtered with DTFs. DTFs are generally measured at finite locations in azimuth and elevation, so models are needed to synthesize DTFs for source locations not measured. Here, we investigated the use of different pole/zero models with different orders to model DTFs measured in humans and cats. With models that successfully characterized the DTFs with little error, we examined the systematic movements of the poles/zeros in the unit circle with changes in source azimuth and elevation. The ultimate goal of this research is to find a simple relationship between the movements of the poles/zeros with source location. [NIH Grant No. R01-DC6865.]

4aPP2. A study on the input-output function of a nonlinear cochlear transmission-line model with the function of an outer hair cell model. Yasuki Murakami and Masashi Unoki (School of Information Sci., JAIST, 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan, y-muraka@jaist.ac.jp)

A nonlinear cochlear transmission-line model with the physiological function of an outer hair cell (OHC) model is presented to investigate how OHCs produce the nonlinearity in the input-output (IO) functions of cochlear filtering. As the physiological function of the OHC model, the pressure produced by the somatic motility of the OHCs was studied. The somatic motility is induced by the transducer currents that change with the displacement of the hair bundle. The pressure produced by the OHC

model can be linearly approximated by the amount of current. The proposed model is comprised of a middle ear model and a cochlear transmission-line model, which includes a basilar membrane and lymph fluid model, a tectorial membrane model, hair bundles model, and the OHC model. The model's parameters were set as estimates for a human obtained from animal data. In simulation results, this model could account for the IO functions of cochlear filtering obtained from both physiological and psychological experiments. These results suggest that the somatic motility of the OHCs, which depends on the transducer currents, produces the nonlinearity in the IO functions of cochlear filtering.

4aPP3. Does Norwich's entropy theory of perception produce equal-loudness contours? Lance Nizami[a] (1312 Grayson Pl., Decatur, GA 30030, nizamii2@aol.com)

Norwich & Wong [Percept. Psychophys. 59, 1997; derived backwards in JASA, 97, 1995] modeled loudness L versus intensity I by $L(I,k,\gamma,I_{th},n)=(k/2)ln(1+\gamma(I/I_{th})^n)$. They assumed $L_{th}=L(I_{th})$ and $L=L-L_{th}$, producing *two* loudnesses, L or $L=L-L_{th}$, giving $L(I_{th})=L_{th}$ or 0, and $L(0)=0$ or $(-k/2)ln(1+\gamma)$, a *negative*. They took dL/dI, called it $\Delta L/\Delta I$, and rearranged it, giving $\Delta I/I(\Delta L,k,\gamma,I_{th},n)$. New terms $S_\infty(\Delta L,n,k)$ and $S_0(\Delta L,n,\gamma,k)$ were introduced, thus $\gamma=S_\infty/(S_0-S_\infty)$. The result was $\Delta I/I$ in S_∞, S_0, and n, identical to an equation of Riesz (Phys. Rev., 31, 1928). Riesz's empirical S_∞, S_0, and n provided γ's and n's for $L(I)=L(I,k,\gamma,I_{th},n)-(k/2)ln(1+\gamma)$. *Assuming k was constant*, Norwich & Wong equated $L(I)$'s for comparison and reference tones, generating equal-loudness contours. The contours contradicted empirical ones; to adjust, they replaced Riesz's n by Stevens' Index, x. The present author curvefitted the Entropy Equation and Stevens' Law to 37 loudness-growth plots. k depended on maximum loudness as $k=0.173L_{max}^{1.41}$ $n\neq x$, rather, $n=0.123+1.215x$. Altogether, the derived contours are invalid. [a]Research done at Univ. of Toronto in Mississauga, 3359 Mississauga Rd. N., Mississauga, ON, Canada.

4aPP4. Tonotopic map reorganization and spectral weights in high-frequency sensorineural hearing loss. Blas Espinoza-Varas (Dept. Commun. Sci. and Disord., OU Health Sci. Ctr., 825 NE 14th St., Oklahoma City, OK 73190), Hammad Akram, Titus Oleyadun (College of Public Health, Oklahoma City, OK 73190), Merlyn George (Univ. of Oklahoma, Norman, OK 73019), and Hyunsook Jang (Hallym Univ., Chuncheon 200-702, Korea)

Owing to reduced afferent input from high characteristic-frequency hair cells, high-frequency sensorineural hearing loss (SNHL) reorganizes the tonotopic maps in the central auditory nervous system: Map expansion is observed in the midfrequencies and map contraction in the high frequencies. Does the relative weighting of mid and high frequency information depend on this reorganization? With complex tones consisting of a 1,000- and a 3,127-Hz sinusoid, 80 and 60 ms each, the resolution of simultaneous increments in frequency (IF) at 1,000 Hz and in duration (IT) at 3,127 Hz was studied in participants having either normal hearing or moderate-to-severe SNHL above 2,000 Hz. In both groups, the sensation level at 1,000 Hz (1.5–8 dB) was 17–38 dB lower than at 3,127 Hz (25–39 dB). In normal-hearing participants, IF resolution was inferior to that of IT, suggesting stronger weighting for the more audible component. In hearing-impaired participants, IF resolution was superior or equal to that of IT, suggesting stronger weighting for the lower sensation level 1,000-Hz component impinging on the expanded tonotopic map region; in addition, IT resolution improved significantly following extensive training. [Work supported by PHF and NIH grants.]

4aPP5. Independence in the perception of complex non-speech sounds. Noah Silbert (Dept. of Linguist., Indiana Univ., Memorial Hall 322, 1021 E. Third St. Bloomington, IN 47405), James Townsend, and Jennifer Lentz (Hearing Sci., Indiana Univ., Bloomington, IN 47405)

Little, if any, work has explicitly addressed independence in the perception of complex sounds. General recognition theory provides a powerful framework in which to address such issues. Two experiments were carried out to test within-stimulus, between-stimulus, and decision-related notions of independence in each of two stimulus sets. One set consisted of broadband noise stimuli varying in frequency and duration, the other set consisted of harmonic tone complexes varying in fundamental frequency and spectral shape. Model fitting analyses indicate that decision-related independence (decisional separability) holds for all participants with the noise stimuli and the majority of participants with the harmonic stimuli, that within-stimulus independence (perceptual independence) holds for all participants with the noise stimuli and between-stimuli independence (perceptual separability) holds for all but two participants with the noise stimuli, whereas considerable individual differences are exhibited in a variety of failures of either perceptual independence or perceptual separability or, occasionally, both, for participants with the harmonic stimuli. The noise stimulus results are consistent with expectations based on models of frequency and duration encoding, whereas the harmonic stimulus results present a challenge to a number of models of pitch and timbre encoding. [This work was supported by NIH Grant No. T32 MH019879-13.]

4aPP6. Development and achievement of hearing-aid fitting strategy based on loudness model. Ying-Hui Lai, Yang-Ming Jhou, and Sheunn-Tsong Young (Inst. of Biomed. Eng., Natl. Yang-Ming Univ. No. 155, Sec. 2, Linong St., Taipei City 112, Taiwan, young@bme.ym.edu.tw)

The purpose of this study was to develop a hearing-aid fitting strategy (named NYMU strategy), which aimed to compensate the loudness perception of hearing-impaired individuals as that of normal-hearing individuals. The loudness growth curve of normal-hearing individuals was derived by the equal loudness contour. The loudness growth curve of hearing-impaired individuals was constructed by fitting the curve through three points, which were the uncomfortable loudness level, the most comfortable loudness level, and the pure-tone threshold of hearing-impaired individuals for each frequency band. The NYMU strategy primarily calculated the hearing-aid gain for each frequency band according to the differences between the loudness growth curve of normal-hearing individuals and that of hearing-impaired individuals; the gain was set to guarantee the audibility of the lowest boundary of speech intensity with normal vocal effort. Furthermore, the NYMU strategy adopted an innovative approach, which adjusted the point of the most comfortable loudness level to fine-tuning the loudness growth curve, and it resulted in the fine-tuning gain of each band. The preliminary results reveal that the NYMU strategy provides more gain above 4,000 Hz and under 500 Hz, compared with the NAL-NL1 strategy. In addition, three of five subjects get higher word recognition scores when they were fitted by the NYUM strategy rather than the NAL-NL1 strategy.

4aPP7. Off-frequency listening: A population study. Veronica Eckstein and Bruce Berg (Dept. of Cognit. Sci., Univ. of California Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697)

A pitch-matching paradigm was used to investigate the pitch of three-tone complexes in a college-age population. For different conditions, the central tone was 1,000 Hz with the side tones spaced by either 15 Hz or 45 Hz. The tones were equal intensity or a signal was added in phase to the central tone with a level of 5, 0, or 5 dB relative to the side tones. Listeners adjusted the frequency of a pure tone to match the pitch of the complex. According to a model known as the envelope-weighted average of the instantaneous frequency (EWAIF) [L. Feth, Percept. Psychopys. **15**, 375–378 (1974)], the pitch should be approximately equal to the center frequency for all conditions. Contrary to this prediction, increasing the bandwidth or the decreasing the signal level systematically lowered the

pitch. However, the results are consistent with EWAIF calculations if the stimuli are first passed through the skirt of an auditory filter. Results for the population were consistent with pitch matches for expert listeners. [Work supported by the Center for Hearing Research, UCI.]

4aPP8. Predicting the ability to learn complex auditory tasks. Charles S. Watson and Gary R. Kidd (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405)

The goal was to determine whether the ability to learn complex auditory discrimination and identification tasks could be predicted from performance on a screening test battery. The screening battery included auditory tasks chosen on the basis of earlier factor-analytic studies of individual differences [Kidd et al. JASA **122**, 418–435 (2007)], visual acuity tasks, and a test of working memory. These tests were administered to 1,000 college students with normal audiograms. Based on screening test performance, 25 exceptional performers and 25 average performers were selected for prolonged training on three new auditory tasks. The training tasks included multi-tone pattern discrimination, identification of a vocabulary of sequences of three-tone syllables, and binary categorization of multi-dimensional profile-like stimuli. A broad range of individual differences in post-training performance was observed on each of these tasks, but differences were not well predicted by screening task performance or estimated general intelligence. Analysis of the screening task performance supported the previously reported independence of individual differences in spectral-temporal acuity and the recognition of familiar sounds (both speech and nonspeech). Intellectual abilities and visual processing abilities were also independent of each other and of auditory abilities. [Work supported by Grant No. N000140310644 from the Office of Naval Research.]

4aPP9. SPATS: Speech perception assessment and training system. James D. Miller, Charles S. Watson, Diane Kewley-Port, Roy Sillings, William B. Mills, and Deborah F. Burleson (Commun. Disord. Technol., Inc., 501 N. Morton St., Sta 215, Bloomington, IN 47404)

A software system (SPATS) to test and train important bottom-up and combined bottom-up/top-down speech-perception skills is described. Bottom-up skills are the abilities to identify the constituents of syllables: Onsets, nuclei, and codas in quiet and noise as produced by eight talkers. Top-down skills are the abilities to use knowledge of linguistic context to identify the words in spoken sentences. The sentence module in SPATS emphasizes combined bottom-up/top-down abilities in recognizing sentences in noise. The word-initial onsets, stressed nuclei, and word-final codas are ranked in importance and grouped into subsets, based on their importance. Testing utilizes random presentation of all the items included in a subset. Training in Quiet (SNR=40 dB) or in Noise (SNR=5 dB), is adaptively focused on individual listeners learnable items of intermediate difficulty. Alternatively, SNR-adaption training uses Kaerenbachs algorithm to find the SNR required for a target percent correct. The unique sentence module not only assesses and trains a listener's ability to infer words in sentences based on linguistic cues, but also trains the ability to combine bottom-up with top-down abilities to identify words in spoken sentences in noise. Scoring in the sentence module is objective and automatic. SPATS demonstrations are included.

4aPP10. Preliminary evaluation of the speech perception assessment and training system (SPATS) with hearing-aid and cochlear-implant users. James D. Miller, Charles S. Watson (Commun. Disord. Technol., Inc., 501 N. Morton St., Sta 215, Bloomington, IN 47404), Doris J. Kistler, Frederic L. Wightman (Heuser Hearing Inst., Louisville, KY 40203), and Jill E. Preminger (Univ. of Louisville, Louisville, KY 40292)

SPATS is evaluated as a testing and training system for hearing-aid (HA) users and cochlear-implant (CI) users. Criterion measures include the HINT, CNC tests, W22 tests, and Coxs CDT, parts of Gatehouses SSQ and a special SPATS inventory. SPATS measures include the identification of syllable constituents (onsets, nuclei, and codas) and measures of top-down and combined top-down and bottom-up recognizing spoken sentences. Control subjects were measured on criterion and SPATS tests, and then retested after a pause of several weeks. Trained subjects took all of the same tests, but in the time between first and second testing underwent either 12 or 24 hours of systematic training using special SPATS algorithms that focus training on items of intermediate difficulty in quiet and noise. Trained subjects show gains on speech-perception measures in quiet and noise, and in look-and-listen tasks, even though there was no training of visual speech perception. Subjects report that SPATS training and testing gave them a much clearer understanding of the severity of their hearing impairments and led to improved speech perception in everyday life through greater attention to detail, and to differences between talkers. [Supported by Grant No. R44DC006338 from NIH/NIDCD.]

4aPP11. The absolute threshold of hearing changes by perceiving a previous sound in the contralateral, ipsilateral, and both ears. Hiroshi Hasegawa, Junji Yoshida, Masao Kasuga (Grad. School of Eng., Utsunomiya Univ., 7-1-2 Yoto, Utsunomiya-shi, Tochigi-ken 321-8585, Japan), and Yu Watanabe (Utsunomiya Univ., Utsunomiya-shi, Tochigi-ken 321-8585, Japan)

This study investigated the effects of a previous sound on loudness at the absolute threshold of hearing. Change of the absolute threshold of hearing was measured when a pure tone preceded the test tone in the measurement of the threshold. The previous sound at 60 dB SPL was presented first in one ear, and then the test sound was presented in either the contralateral or ipsilateral ear at an interval of 0.5 s. When the previous sound was presented first in both ears, the test sound was presented second in one ear (left or right ear). Both the previous and test sounds had the same frequency of 500 Hz, and the same duration of 3 s. The threshold change was obtained from the difference between the thresholds with and without the previous sound. The threshold was found to be decreased by 3.1 dB when the previous sound was presented in the contralateral ear and increased by 0.9 dB in the ipsilateral ear. On the other hand, the threshold was decreased 2.3 dB when the previous sound was presented in both ears.

4aPP12. Examination of bone-conducted sound pathways. Lynn M. Brault, Charissa R. Lansing, Ron D. Chambers (Dept. of Speech and Hearing Sci., UIUC, 901 S. Sixth St., Champaign, IL 61820), and William D. O'Brien (UIUC, Urbana, IL 61801)

Evidence of bone-conducted sound pathways through the skull have been explored with growth of loudness tests, auditory brainstem response, and otoacoustic emissions; however a comprehensive understanding of these pathways has not been achieved. Less is known about the usefulness of behavioral thresholds, sound pressure levels in the ear canal, and the auditory steady-state response (ASSR) as a function of ear canal occlusion in understanding propagation pathways of bone-conducted signals. Results acquired from multiple, repeated test sessions with a single-subject design, revealed lower behavioral thresholds and higher sound pressure level in the ear canal (Real Ear) in the occluded condition below 2,000 Hz; however, this trend was reversed above 2,000 Hz. The ASSR revealed lower thresholds in the occluded condition at all frequencies. Real Ear measured the bone-conducted signal while the behavioral thresholds show the product of the signal after following the traditional bone conduction pathway through the ear structures and the brain. The ASSR results indicate the responses recorded were suprathreshold, and are the product of bone conduction sound excitation at the level of the auditory nerve. Understanding bone-conducted sound pathways could provide guidance for designing enhanced hearing protection, new diagnostic protocols, and novel approaches for signal processing. [Work supported by AFOSR FA9550-06-0128.]

4aPP13. Infants' and adults' sensitivity to a tone masked by remote frequency tones or noise. Lynne A. Werner (Dept. Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105-6246) and Lori J. Leibold (The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC 27599)

It was previously reported that infants and adults exhibit masking of a tone by two tones of constant, remote frequencies. Neither informational nor energetic masking would be expected for adults under these conditions. In the present experiment, infants' and adults' detection of a tone masked by two constant frequency tones or two narrow bands of noise were compared. The target tone was 1,000-Hz, 300-ms duration with 16-ms rise/fall, at a level expected to produce $rmd' = 1$ in the two-tone masker condition for each age group. The maskers were either tones of 581 and 2,920 Hz or 50-Hz-wide noise bands centered at 556 and 2,895 Hz. The duration of the maskers was 300 ms with 16-ms rise/fall. The maskers repeated at 600-ms intervals. The target tone was presented simultaneously with the maskers on signal trials. Sensitivity was measured using an observer-based procedure. Listeners received feedback whenever they correctly detected the target. Infants achieved a higher rmd' in the two-noise-band condition than in the two-tone condition; there was no difference in rmd' between these conditions for adults. Thus, introducing a timbre difference between the target and masker reduced the amount of masking for infants, but not for adults.

4aPP14. Intelligibility differences between male and female bone conducted speech when presented in high noise environments. Maranda McBride, Meghan Hodges, and Jon French (Embry-Riddle Aeronautical Univ., 600 S. Clyde Morris Blvd., Daytona Beach, FL 32114)

Clear and intelligible verbal communication is essential in many organizations. When designing verbal communication devices, developers must not only consider the electrical and mechanical components of the devices themselves, but also various parameters of both the listening environment and vocal signals being transmitted. Bone conduction communication devices have recently been introduced into several tactical organizations; however, the effectiveness of these devices in transmitting vocal signals has not yet been comprehensively explored. This paper describes a study investigating the impact of voice type (male and female), vibrator location (condyle of the jaw and mastoid process), and background noise level (0, 83, 93, and 103 dB(A) SPL) on the intelligibility of bone conducted verbal communication. The fundamental frequencies of the male and female voices were approximately 100 and 185 Hz, respectively. The background noise used in the study was pink noise. Six male and six female students from Embry-Riddle Aeronautical University participated in the study. The results of the study indicate significantly higher speech intelligibility scores for the male voice than for the female voice, and the condyle was significantly more effective than the mastoid in transmitting intelligible speech in each of the high noise environments tested.

4aPP15. Psychoacoustic evaluation of broadband and conventional reversing alarms localization accuracy. Stephen Lakatos (Washington State Univ., Vancouver, 14204 NE Salmon Creek Ave., Vancouver, WA 98686, lakatos@vancouver.wsu.edu)

Three psychoacoustic tests assessed whether a recently developed class of vehicle reversing alarms with broadband signal characteristics would be more accurately localized than conventional equivalents. Vehicle reversing conditions were simulated by mounting sample alarms on a linear slide whose velocity and displacement could be controlled by computer. Reversals of 4 m displacement were recorded at two velocities (1 m/s and 2 m/s) and at azimuthal orientations varying in 50 increments. The first two tests presented direct hit trajectories recorded with a binaural microphone (Crown SASS-P) and Kemar Mannequin, respectively; a third test presented near miss trajectories with the recording centerline perpendicular to the alarm trajectory and at linear displacements of 0.5 m and 1 m. Participants listened to trajectories over supra-aural headphones and selected the closest match from computer depictions of all possible trajectories. Results for all three tests revealed no significant advantage in localization acuity for the broadband alarm.

4aPP16. Using multichannel communication systems to observe selective and divided attention. Misty Gripper (Biomed., Industrial and Human Factors Eng. Dept., Wright State Univ., 3640 Colonel Glenn Hwy, Dayton, OH 45435, misty.gripper@wright.edu), Tomasz Letowski, and Tim Mermagen (U.S. Army Res. Lab., Aberdeen Proving Ground, MD 21005)

The objectives of this study are to compare the effectiveness of bone- and air-conduction radio communication while listening to natural auditory signals through air conduction and to assess the feasibility of using a mixed bone- and air-conduction system for multichannel radio communication. There are many benefits to both air-conducted and bone-conducted systems, but little research on combining the two into one radio system. The goal of the study is to explore this possibility. The SATASK test developed at the U.S. Army Research Laboratory is used to observe human capabilities to distribute attention across multiple channel communication systems. This test consists of sentences structured as follows: NAME write the number NUMBER on the COLOR OBJECT. Five different multichannel communication systems (three three-channel and two four-channel systems) consisting of various combinations of air-conducted earphones, bone-conducted vibrators, and an array of loudspeakers are used for data collection. Each channel presents different sentences in different voices. Listeners are given a target name and they are to record the corresponding information associated with that name (selective attention) or they are to record information from all channels (divided attention). The results of the study will be discussed at the meeting.

4aPP17. Auditory detection of amplitude modulation in psychophysical notched noise task and electroencephalography. Allison I. Shim, Bruce G. Berg, and Ramesh Srinivasan (Dept. of Cognit. Sci., Univ. of California Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697)

Notched tone-in-noise detection tasks have commonly been utilized to investigate the properties of auditory filters. A similar design is used to study filters for auditory temporal processes; amplitude modulated (AM) noise is used as the signal component rather than tone. The signal is composed of a 200 Hz wide noise-band with a 40 Hz modulation rate and centered at 1,000 Hz. Notched-noise maskers composed of 200 Hz unmodulated noise bands located an equal distance logarithmically on either side of the stimulus create the notch. Psychophysical thresholds were measured using a 2IFC adaptive staircase procedure. A common assumption is that thresholds for signal detection are unaffected until the masker impedes upon the edge of a filter. Data were fit using a leaky integrator model with three forms of the bandpass filter. Results support the existence of wide filters necessary for temporal processing. In an effort to integrate psychophysical and brain imaging methods, a similar study was conducted using electroencephalography. Cortical evoked responses at the gamma frequency (40 Hz) also appear to be affected by narrowing notch widths. [Work supported by the Center for Hearing Research, University of California, Irvine.]

4aPP18. Effects of frequency shifts on the identification of vowels and words in sentences. Peter F. Assmann (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, Box 830688, Richardson, TX 75083) and Terrance M. Nearey (Univ. of Alberta, Edmonton, AB, Canada T6E 2G2)

Studies of the effects of frequency shifts on vowel identification have shown a drop in accuracy when the spectrum envelope is shifted up or down, and when the fundamental frequency (F0) is raised or lowered. We have found an interaction between F0 and spectrum envelope shifts: Performance is better for vowels with matched shifts (both F0 and spectrum

envelope shifted in the same direction) compared to mismatched shifts (F0 shifted up and spectrum envelope shifted down or vice versa). The aim of the present study was to determine the extent to which these effects persist in sentence context. The STRAIGHT vocoder was used to process a set of sentences from the HINT test using the same scale factors as in the vowel identification experiment. Word recognition scores generally followed the same pattern as vowel identification, with poorer performance for lowered F0 and raised spectrum envelope, and the lowest scores in conditions with high F0 and downward shifts in spectrum envelope, compared to the corresponding matched shifts. [Work supported by NSF and SSHRC.]

4aPP19. Audio visual interaction in an auditory streaming task involving vowels. Etienne Gaudrain, Nicolas Grimault (Universit Lyon 1, UMR CNRS 5020, 50 av. T. Garnier, 69366 Lyon Cedex 07, France), Fredric Berthommier (Institut de la Commun. Parle, Grenoble, France), Eric Healy (Univ. of South Carolina, Columbia, SC), and Jean-Christophe Bra (Inserm U556, Lyon, France)

Previous studies suggest that sequential segregation mechanisms might be primary in concurrent speech segregation tasks for hearing-impaired listeners. Other studies involving speech materials indicate that timbre cues and pitch cues primarily modulate sequential segregation. Finally, concurrent speech segregation has been reported to be enhanced by visual cues associated with speech reading. In this study, sequences of vowels having alternating fundamental frequencies (low and high) were presented to normal-hearing listeners who were asked to report the presentation order of the vowels. Lower performance in such a task has been related to increased sequential segregation. By directing listeners to the perception of a single stream, this task leads to an estimation of the pitch-based obligatory streaming evoked by the sequence. Simultaneous with the presentation of each high-frequency vowel, visual cues simulating lip movements were provided. Lip movements were either steady (control condition), open-closed (rhythm condition), or congruent with each particular vowel (congruent condition). As such, lip movements were used in this study as a potential distracter. The lower performance (increased streaming) in the congruent condition relative to control conditions suggest that visual cues may interact with sequential obligatory streaming mechanisms at a sensory level.

4aPP20. Effects of a response format modification on the diagnostic rhyme test. Edward L. Goshorn (Dept. of Speech and Hearing Sci., Univ of Southern Mississippi, 118 College Dr. No. 5018, Hattiesburg, MS 39406, edward.goshorn@usm.edu)

Detection theory measures of listener performance, as well as distinctive feature scoring, have been shown to provide useful additional information to percent correct scores when testing word recognition. The diagnostic rhyme test (DRT) uses a two-item multiple choice (MC) response format to provide percent correct and distinctive feature information but does not measure d. This study examined the effects of modifications to the DRT response format that allow a measure of d. The format was modified to a single-option in which the listener designates yes or no (Y-N) that the single-option matches the stimulus. DRT words were mixed in white noise and presented in sound field at 3 and 6 dB signal-to-noise ratios at 50 dBHL to eight normal-hearing subjects. Results showed a significant difference between response formats for percent correct and a significant interaction between signal-to-noise ratios and feature errors across formats. At 3 dB, there were more temporal-based feature errors (sustension and compactness) for the Y-N response format, but no significant differences between formats for voicing, nasality, sibilation, and graveness. At 6 dB, there were significant differences only for sustension and graveness. The ds for signal-to-noise ratios were significantly different for the Y-N format.

4aPP21. Coupled cross-modal activity in the human auditory and visual cortices. William Winter (Univ. of California, Irvine, 8730 Palo Verde, Irvine, CA, 92617)

From the perspective of systems analysis, the reactions of the brain to a steady state stimulus (steady state evoked fields) represent the brain's system response. In a linear system, these responses are expected to occur only at the stimulus frequency. However, in a nonlinear system, power can disperse to other frequencies, usually at integer combinations of the stimulus frequencies. By stimulating different components of a nonlinear system at various frequencies, it is possible to evaluate the strength of coupling between components. In this study, steady state auditory and visual stimuli are presented at various frequencies to human subjects to evaluate the nonlinear coupling between the auditory and visual systems. Subjects in whom strong responses are found will be further evaluated with simultaneous electroencephalography/functional magnetic resonance imaging and diffusion tensor imaging to attempt to determine the extent to which the cross-modal activity is due to field versus fiber propagation.

4aPP22. Evaluation of the consonant-nucleus-consonant word test as spoken by a female talker. Margaritis Fourakis, John Hawks, Heather Zingler, Lyndsey Persak (Dept. of Communicative Disord., UW-Madison, 1975 Willow Dr., Madison, WI 53706), John Hawks (Kent State Univ., Kent, OH 44242), Heather Zingler, and Lyndsey Persak (UW-Madison, Madison, WI 53706)

The most common recording of CNC words used to test speech perception by persons with cochlear implants utilizes a single male talker. However, research has shown that CI recipients often have difficulty perceiving speech by persons with high F0 and formant frequencies (Loizou, P., Dorman, M., and Powell, V. 1998. J. Acoust. Soc. Am. 103: 1141–1149). Presented here are CNC word lists recorded using an adult female talker. Ten normal hearing adults identified these words with 99% accuracy. Eight CI subjects also listened to the original (male voice) and new (female voice) recordings. The two star subjects identified the female speakers words 15% less accurately than the male speakers words (Male voice: 82% correct; Female voice: 67% correct). For the other CI subjects (mean performance around 40%) there was no effect of voice on their performance. Results of the acoustic analysis of all the stimuli will also be presented. [Research supported by a subcontract from Washington University in St. Louis, NIDCD R01 DC000581.]

4aPP23. The perception of auditory-tactile integration. E. Courtenay Wilson, Charlotte M. Reed, and Louis D. Braida (Harvard-MIT Div. of Health Sci. and Technol., Speech and Hearing Bioscience and Technol. Program; Res. Lab. of Electrons, MIT, 36-761, 50 Vassar St, Cambridge, MA 02139)

This project explores the perceptual interactions between certain auditory and tactile stimuli in an objective manner. Our hypothesis states that if the auditory and tactile systems integrate into a common pathway, the d-prime measure of the two sensory stimuli presented simultaneously will be significantly greater than the d-prime measure of the individual sensory stimuli. If the stimuli are judged independently, the resulting d-prime will be close to the root-squared sum of the individual sensory d-primes. If the stimuli are integrated into a single percept, the d-prime will be close to the sum of the individual d-primes. This experiment presents 250 Hz vibratory stimuli to the fingertip and 250 Hz tones binaurally at the subject's detection threshold. Experimental measurements of d-prime were obtained for auditory-alone, tactile-alone, and combination of auditory plus tactile. All testing used auditory white noise of 50 dB SPL. Effects of relative tactile-auditory phase and temporal asynchrony of the stimuli are explored. Preliminary data suggest that under certain conditions the two sensory stimuli integrate in a common pathway to form a single percept, that the stimuli must be synchronous for integration to occur, and that relative tactile-auditory phase has no effect on integration. [Hertz Foundation, NIH RO1-DC000126-25, ROI-DC000117.]

Session 4aSA

Structural Acoustics and Vibration: Modeling of Vibration and Radiation in Complex Structures

Courtney B. Burroughs, Chair
1241 Smithfield St., State College, PA 16801

Invited Papers

8:00

4aSA1. Numerical difficulties with boundary element solutions of interior acoustic problems. John Fahnline (The Appl. Res. Lab., The Penn State Univ., University Park, PA 16802)

Although boundary element methods have been applied to interior problems for many years, the numerical difficulties that can occur have not been thoroughly explored. Various authors have reported artificial damping due to acoustic radiation and low-frequency breakdowns. In this presentation, it is shown through a simple example problem that the numerical difficulties depend on the solution formulation. When the boundary conditions are imposed directly, the solution suffers from artificial damping due to sound radiation to the farfield. This difficulty can be alleviated by first computing an impedance matrix or admittance matrix, and then using its reactive component to derive the solution for the acoustic field. Numerical computations are used to demonstrate that this technique eliminates artificial damping, but does not correct errors in the reactive components of the impedance or admittance matrices, which then cause nonexistence/nonuniqueness difficulties at the interior resonance frequencies for hard-wall or pressure release boundary conditions, respectively. It is shown that the admittance formulation is better suited to boundary element computations for interior problems because the resonance frequencies for pressure release boundary conditions do not begin until the smallest dimension of the boundary surface is at least one half of the acoustic wavelength.

8:30

4aSA2. Development of a general statistical energy analysis subsystem formulation using finite element periodic structure theory. Vincent Cotoni, Phil Shorter (ESI Group, 12555 High Bluff Dr., Ste. 250, San Diego, CA 92130), and Robin Langley (Univ. of Cambridge, Cambridge CB2 1PZ, UK)

Statistical energy analysis (SEA) represents a field of study in which statistical descriptions of a system are employed in order to simplify the analysis of complicated vibro-acoustic problems. In SEA, a vibro-acoustic system is represented by a collection of subsystems that can receive, store, dissipate, and transmit vibro-acoustic energy. Traditionally, the SEA parameters for a given subsystem are formulated analytically based on consideration of wave propagation through the subsystem. While such analytical algorithms can be readily applied to the majority of systems encountered in practical problems, there are certain types of sections that are difficult to describe using existing analytical formulations. Examples include: Isogrid in launch vehicle fairings, extruded aluminum sections in train floors, and modern corrugated aircraft fuselage constructions. This paper describes the development of a generic SEA subsystem formulation based on the use of finite element (FE) periodic structure theory. A small unit cell of the section is created and computationally efficient algorithms are developed to calculate wave propagation through a large array of such cells. The resulting algorithms are used to calculate the SEA parameters for the section. The approach is described and a number of numerical validation examples are presented.

9:00

4aSA3. The response of a beam-like acoustic sensor to transient sound waves. W. Steve Shepard, Jr. and Bill B.B. Zhang (Dept. of Mech. Eng., The Univ. of Alabama, Box 870276, Tuscaloosa, AL 35487, sshepard@eng.ua.edu)

An analytical approach is presented for studying the response of a beam-like structure being considered to act as an acoustic sensor under the excitation of a transient sound wave. First, the response of a Bernoulli-Euler beam is modeled for a multi-cycle transient wave excitation moving at a constant speed. For verification, the model is compared to other approaches by looking at extremes of the input. Then, to examine sensing requirements for the beams structure, the influence of various parameters on the ability to excite the sensor in the midfrequency region is investigated. The parameters considered include beam length, beam materials, amplitude of the transient pressure excitation, duration of the excitation, and the number of sinusoidal half-cycles present in the transient excitation. Potential effects of the superposition of incident and reflected waves caused by the boundary conditions on the sensors response are discussed. [Work supported by the National Science Foundation.]

9:30

4aSA4. S-matrices for waves in random structures, unitarization of diffuse field waveforms, and mesocopics. Richard Weaver (Dept. of Phys., Univ. of Illinois, Urbana, IL 61801)

Diffuse field methods, such as reverberation room acoustics and statistical energy analysis, predict wave energy flow in complex structures. These methods are applied here to predict the root mean square s-matrix $<s^2(t)>^{1/2}$ for an undamped structure coupled to a single incoming and outgoing channel. Multiplying by a white noise random process produces a candidate s(t). The prediction is statistically exact if the internal scatterings are sufficiently phase incoherent, but it is not unitary and is thus inadmissible in the absence of such scatterings. Here it is shown that it can be made unitary in a minimally invasive manner by filtering with a minimum phase causal filter with modulus $1/|s(\omega)|$. This is illustrated with envelopes $<s^2(t)>$ predicated on a picture of the structure as a single reverberation room, as two coupled rooms, and as a quasi-one dimensional multiply scattering system. The resulting unitarized S matrices are found to exhibit a variety of familiar mesoscopic features and behaviors not present in the original diffuse $<s^2(t)>$. These include enhanced backscatter, quantum echo, power law tails, level repulsion, and Anderson localization. It is remarkable, and of potential practical interest, that quantitative mesoscopic features follow from simple principles. [Work supported by NSF CMS 05-28096.]

9:45

4aSA5. Modeling of statistics of structural elements vibration, induced by near-wall turbulence. Leonid R. Yablonik (Polzunov Central Boiler & Turbine Inst., 3/6 Atamanskaya str., 191167 St.-Petersburg, Russia) and Emmanuil M. Agrest (Johnson & Wales Unv., Charlotte, NC 28202)

The statistical characteristics of elastic elements fluctuations, generated by the dynamic effect of near-wall turbulent pressures field, are analyzed. The probabilistic properties of vibration of elements are determined by the characteristic functional of the random pressure field. A superposition of Gaussian and Poissonian field is used to model turbulent pressure in accordance with [E. B. Kudashev and L. R. Yablonik, Acoust. Phys. **48**(3), 321–324 (2002)]. The Poisson component of turbulent field, which is associated with the processes of the generation of spontaneous splashes in the near-wall zone of flow, has a basic impact on the statistical features of vibrations. The model relationships, which connect flow conditions with the parameters of the characteristic functional of Poisson component, are proposed. Characteristic functions of the amplitudes of vibrations with the different values of the parameters of the acting turbulent field simulations are represented. The statistics of the inertia-free pistons that are usually used in transducers for pressure turbulent field diagnostics is investigated in detail.

10:00–10:15 Break

10:15

4aSA6. A simplified finite-element modeling approach for constrained layer damping. Zhengchao Xie and W. Steve Shepard, Jr. (Dept. of Mech. Eng., The Univ. of Alabama, Box 870276, Tuscaloosa, AL 35487)

In formulating finite-element models for multilayer structures, designers usually use a detailed model that contains many degrees of freedom. In order to reduce the modeling complexity for structures containing constrained layer damping, some researchers have attempted to develop equivalent properties that are applied to regular single-layer finite elements. For example, the analytical formulation developed by Ross, Ungar, and Kerwin is sometimes used to find an equivalent bending stiffness. However, that stiffness is based on an infinite or simply-supported beam, which is often inappropriate. The research presented here uses the governing equations of a regular beam or plate to derive the frequency transfer function between two points on the structure. Curve fitting this transfer function to a reference value found using a more detailed model provides a complex bending stiffness at the first few peak frequencies. The goal is to develop an improved approach for extracting equivalent properties for the simplified modeling of multilayer structures. Numerical examples show that extracted properties can lead to good results when calculating the response of the structure. This approach may provide a means of reducing the modeling requirements for viscoelastically damped structures.

10:30

4aSA7. Dynamics of a hollow rubber-metal laminate structure. Robert W. M. Smith (Penn State Appl. Res. Lab., P.O. Box 30, State College, PA 16804)

Transverse frequencies of vibration of a hollow cylindrical rubber-metal laminate structure, subject to a mean pressure difference across the cylinder wall will be presented. Extending Timoshenko beam theory to model such a structure, we use two independent observations most often credited to J. A. Haringx: First, an analogy exists between an axially loaded Euler beam and a pipe without axial loading but subject to internal pressure, and second, for this type of rubber-metal laminate structure, the transverse component of the axial load acts perpendicular to the bending slope, rather than the deflection slope. When dynamics of the pressurized laminate are considered, theory predicts somewhat unexpected changes in the resonance frequency of transverse bending modes as a function of mean pressure difference across the structure. Experimental results illustrating static buckling behavior will be shown for both positive and negative pressure differentials, corresponding to the analogous axial loading cases of buckling in both compression and tension. In applications where oscillatory pressures are present across the structure, the unusual tuning curve has implications for prediction of parametric instabilities.

10:45

4aSA8. Radiation from a source embedded in a two-layer coating of a spherical shell. Jerry H. Ginsberg (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

Analysis of the acoustic field generated by a small source embedded in a viscoelastic layer encapsulating a spherical shell gives rise to several analytical difficulties not encountered in the classical problem of a point source in the surface of a rigid sphere. The present investigation addresses a double layer configuration in which the layer closest to the shell is acoustically soft, whereas the outer layer, in which the source is embedded, has an impedance that matches that of the surrounding fluid, water. The analysis is carried out by treating the layers as dissipative fluids, while classical Love theory describes the interior shell. The analysis employs spherical harmonic expansions for each medium, which is known as Mie series analysis in related studies of electromagnetic scattering from coated series. Like the electromagnetic problem, computational difficulties associated with the occurrence of spherical Bessel functions of complex argument arise, but the remedies used there require modification.

11:00

4aSA9. Frequency spectra of elliptic cylindrical shells. Jeffrey E. Boisvert (NAVSEA Newport, 1176 Howell St., Newport, RI 02841) and Sabih I. Hayek (Penn State Univ., University Park, PA 16802)

The kinetic and strain energy densities were derived for the vibration of an elliptic cylindrical shell of constant thickness. The elastic strain energy density has seven independent kinematic variables: Three displacements, two thickness-shear, and two thickness-stretch. The shell has a constant thickness h, finite length L, and is simply supported at its ends $(z=0,\ L)$, where z is the axial coordinate. The elliptic cross section is defined by the shape parameter a, and the half-length of the major axis l. Using appropriate comparison functions that satisfy the boundary conditions for a simply supported elliptic cylindrical shell, the system is solved using the Galerkin method. Since the shell has an elliptic cross section, the frequency spectra are categorized into distinct symmetric and antisymmetric displacement fields whose spectra coalesce as the elliptic cross section approaches a circular cross section. Numerical results are presented for two h/l and L/l ratios, and various shape parameters, including the limiting case of a simply supported cylindrical shell ($a\sim100$). [Work supported by the ONR/ASEE Summer Faculty Research Program.]

11:15

4aSA10. Tilted aluminum cylinder acoustic scattering properties and holographic and synthetic aperture sonar (SAS) images. Kyungmin Baik, Christopher Dudley, and Philip L. Marston (Phys. and Astron. Dept., Washington State Univ., Pullman, WA 99164-2814)

A solid bluntly-truncated aluminum cylinder in water is a convenient target for exploring the relationship between free-field scattering properties of short tone-bursts and acoustic images. These experiments concern situations where the product ka of the acoustic wave number and the cylinder radius exceed about 12. There are several prominent features in the tilt-angle dependence of the time-resolved backscattering. Some of these features are also present in prior measurements for tilted stainless steel cylinders using relatively long tone bursts [K. Gipson and P. L. Marston, J. Acoust. Soc. Am. **106**, 1673–1680 (1999); K. Gipson and P. L. Marston, J. Acoust. Soc. Am. **107**, 112–117 (2000)]. Prominent features include responses caused by guided generalized Rayleigh waves associated with meridonal rays, helical rays, and face crossing rays. These features are easily observed for tilted aluminum cylinders in the scattering of short tone bursts for cylinders having length-to-diameter ratios of 3 and 5. Guided Rayleigh waves also influence the appearance of the imaged cylinders in holographic and synthetic aperture sonar (SAS) images constructed using far-field bistatic data. This influence on the images is explained using ray theory. [Supported by ONR.]

11:30

4aSA11. Scattering from penetrable prolate spheroids. Jeffrey E. Boisvert (NAVSEA Newport, 1176 Howell St., Newport, RI 02841) and A. L. Van Buren (Cary, NC)

The problem of acoustic scattering from a fluid-filled elastic layer (shell) is considered. The inner and outer surfaces of the elastic shell are confocal prolate spheroids with the outer surface having major-axis length L and minor-axis length D. A plane wave ensonifies the shell at an arbitrary angle of incidence. The 3-D equations of elasticity in prolate spheroidal coordinates are solved using a Helmholtz decomposition for the displacement vector. The displacement vector is expressed in terms of a scalar potential (dependent on the dilatational wave number) and a vector potential (dependent on the shear wave number). These wave potentials are solutions, respectively, of the scalar and vector wave equations cast in prolate spheroidal coordinates. An eigenfunction expansion of spheroidal wave functions represents the solution of the scalar wave equation. The solution of the vector wave equation is facilitated using expansions of prolate spheroidal vector wave functions. Nearfield and farfield scattering results are presented as a function of incident angle, spheroid shape L/D, and internal fluid (air-filled or water-filled). Comparisons with the rigid case (impenetrable spheroid) are also presented. [Work supported by the NAVSEA Newport ILIR Program.]

11:45

4aSA12. Benchmark solutions for low-frequency structural acoustic target scattering. Mario Zampolli, Alessandra Tesei, and Finn B. Jensen (NATO Undersea Res. Ctr., Vle. S. Bartolomeo 400, 19126 La Spezia, Italy)

Low-frequency scattering problems, typically in the range up to 10 s of kHz for O(1 m) size targets, can be solved by a variety of numerical techniques, each of which has different tradeoffs. In September of 2006, a Target Strength Modeling Workshop was held at NURC with the purpose of establishing a set of benchmark solutions for problems of significance to the underwater structural acoustics modeling community. Problems for low-frequency scattering from spherical and cylindrical objects located in the water column or proud or buried in the seabed are presented. Solution techniques ranging from fully 3-D finite-element/boundary integral techniques, to Fourier decomposition axisymmetric finite-element techniques and semi-analytical methods have been applied to the problem set. The reported solutions show benchmark quality agreement in most cases, and hence could be used as reference solutions for testing low-frequency structural acoustic modeling tools in the future. The results and techniques presented reach beyond the realm of underwater acoustics, and should be of interest to researchers working in the broader field of structural acoustics.

FRIDAY MORNING, 30 NOVEMBER 2007 NAPOLEON B3, 8:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Auditory and Somatosensory Feedback in Speech Production I

Anders Lofqvist, Chair

Haskins Laboratories, 300 George St., Ste. 900, New Haven, CT 06511

Chair's Introduction—8:00

Invited Papers

8:05

4aSC1. Psychophysical detection threshold of formant manipulations during speech production. D. W. Purcell (Natl. Ctr. for Audiol., Univ. of Western Ontario, 1201 Western Rd., London, ON, Canada, N6G 1H1) and K. G. Munhall (Queen's Univ., Kingston, ON, Canada, K7L 3N6)

In altered auditory feedback experiments, participants respond rapidly and alter their speech production to compensate for the induced auditory feedback error. This response does not occur with small feedback perturbations, and thus there must be a threshold for compensation by the auditory-vocal feedback system. What is unknown is the psychophysical threshold during speech production (i.e., how large are manipulations that are consciously detectable?). The purpose of this experiment was to compare the threshold for conscious awareness of perturbation, and the minimal change that induces compensation. A real-time auditory feedback manipulation system was employed in a repeated measures design. The compensation threshold was determined using the change point test during a perturbation ramp of 4 Hz/utterance. In the psychophysical measurement, the size of the feedback manipulation followed a two-alternative forced-choice paradigm. With 17 individuals, the mean psychophysical threshold was 105 Hz (SE: 9 Hz). The mean compensation threshold was 64 Hz (SE: 8 Hz). The significant difference between these thresholds is consistent with the hypothesis that the auditory-vocal feedback system can operate without conscious control and suggests that it may have access to a more sensitive representation of speech formants.

8:30

4aSC2. Suppression of responses to pitch perturbations during Mandarin speech. Charles R Larson, Hanjun Liu (Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, clarson@northwestern.edu), and Yi Xu (Univ. College London, London, UK)

Previous research on the task dependency of voice fundamental frequency (F0) responses to pitch-shifted feedback has shown them to be larger during speech than in a vowel task. The present study investigated how responses to pitch-shift stimuli vary with the intonation of a speech utterance and the timing of the pitch shift stimuli. Mandarin speakers produced a four-word phrase in which the F0 was held relatively steady and then either increased or was held steady on the final word. Pitch shift stimuli (100 cents, 200 ms) were presented 160, 240, or 340 ms after the onset of vocalization. F0 response magnitudes for the 340 ms onset condition were significantly smaller than in the 160 or 240 onset conditions when the F0 level on the final syllable was rising. When the F0 on the final syllable was held constant, there was no difference in response magnitudes between the three onset conditions. The decrease in response magnitude in the 340 ms, F0 rising condition, corresponding to approximately 660 ms before the final syllable, suggests vocal responses can be modulated both by the timing of stimulus presentation and the speech intonation pattern. Possible explanations for this finding will be discussed.

8:55

4aSC3. Role of voice F0 production accuracy and pitch perception ability in modifying compensatory voice F0 responses during sustained vowel productions. Jay Bauer, Marylou Gelfer (Dept. of Commun. Sci. & Disord., Univ. of Wisconsin, Milwaukee, P.O. Box 413, Milwaukee, WI 53201), and James Bashford (Univ. of Wisconsin, Milwaukee, Milwaukee, WI 53201)

The purpose of the present study is to assess whether subject characteristics in terms of voice F0 production accuracy and pitch perception ability are related to the ability to compensate in real-time for altered auditory feedback. To assess the relationship of vocal production, pitch-perception, and voice F0 error-correction, a series of three tasks were completed by typical college-age participants: (1) A pitch-matching task where participants vocally matched the pitch of audible complex tones, (2) two-tone forced choice pitch-discrimination tasks where participants decided if complex tone pairs (200 ms duration for each tone) sounded the same or different, and (3) pitch-shifting tasks where participants produced sustained vowel phonations, while listening to experimentally altered auditory feedback (brief 200 ms pitch perturbations of $+/-20$, 60, or 100 cents were introduced into the feedback). Dependent measures of pitch-matching accuracy, pitch-discrimination sensitivity, and variations in the latency, stability, and magnitude of compensatory pitch-shift reflexes were assessed in all participants. Pilot data ($N=3$) indicate that production accuracy and discrimination sensitivity does not play a role in compensatory control of the voice. Poor pitch-matchers and/or discriminators actually respond to very small magnitude perturbations (20 cents) in a manner similar to accurate pitch-matchers and/or perceivers.

9:20

4aSC4. Modification of acoustic-vocal mappings for fundamental frequency control in singers and nonsingers. Jeffery A. Jones and Dwayne Keough (Ctr. for Cognit. Neurosci. & Dept. of Psych., Wilfrid Laurier Univ., Waterloo, ON N2L 3C5, Canada, jjones@wlu.ca)

Singing requires accurate control of the fundamental frequency (F0) of the voice. Previous work has demonstrated that F0 control involves an interplay between closed- and open-loop control. A frequency-altered auditory feedback (FAF) paradigm was used to examine the acoustic-motor representation of the mapping between F0 feedback and the vocal production system in singers and nonsingers. Participants sang a note while hearing their F0 shifted down one semitone. Both singers and nonsingers compensated for the F0 perturbations by increasing their F0. However, the magnitude of compensations was initially smaller in singers than nonsingers. Moreover, after participants heard their feedback returned to normal, the aftereffects observed for singers were larger than those observed for nonsingers. This pattern of aftereffects generalized to the production of another note different than the one participants produced while hearing FAF. Combined, these observations suggest that singers rely more on an internal model for F0 production during singing than nonsingers.

9:45

4aSC5. Effect of anticipation on voice F0 responses to auditory feedback pitch shifts. Theresa A. Burnett, Elise K. Hindmarsh, and Katie E. McCurdy (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405)

Reflex-like voice F0 responses to anticipated and unanticipated auditory feedback pitch shifts were compared in healthy adults. Participants pressed a button to self-trigger a 100-cent, 100-ms upward shift in their voice auditory feedback over headphones while sustaining the vowel /u/ at a comfortable pitch and loudness. In separate blocks of 30 trials each, pitch shifts were self-triggered either immediately upon button press (Immediate) or after a random delay of 500 to 2000 ms (Random). In both conditions, participants were instructed to ignore the shifts as much as possible. To study any interaction between volitional voice F0 control and anticipation of the pitch shifts, the Immediate and Random conditions were repeated with pitch shifts that were long enough in duration for participants to volitionally oppose them (500 ms). Voice F0 response latency was significantly shorter when pitch shifts were

immediately triggered than when triggered after a random delay, but there was no interaction between stimulus duration and stimulus onset timing for voice F0 response latency, magnitude, or duration. Participants neither suppressed the voice F0 response nor amplified it to completely compensate for anticipated 500-ms pitch shifts. Similarities between the voice F0 response and the long-latency stretch reflex are discussed.

10:10–10:30 Break

10:30

4aSC6. The specificity of sensorimotor learning: Generalization in auditory feedback adaptation. K.G. Munhall, E.J.S. Pile, E.N. MacDonald (Dept. of Psych., Queen's Univ., Kingston, Canada), H.R. Dajani (Univ. of Ottawa, Ottawa, Canada), and D. W. Purcell (Univ. of Western Ontario, London, Canada)

An enduring question about sensorimotor learning is how specific the acquired input-output relationship is. In this presentation, we review a series of studies in which the stimuli and conditions during speech motor learning were manipulated to study when and if generalization occurs. In our studies, the first and second formants of vowels were shifted using a real-time signal processing system; when subjects spoke one vowel, they heard themselves saying another vowel. In response to this auditory feedback perturbation, talkers spontaneously compensated by producing formants in the opposite direction in frequency to the perturbation. These compensations persisted after feedback was returned to normal, indicating that a form of sensorimotor learning had taken place. When different vowels were tested following the perturbation of one vowel, they were found not to show evidence of the feedback perturbation nor did the new vowels have any influence on the perturbed vowels' return to normal baseline levels. Data from studies in which listening versus producing were compared and studies in which the similarity of the feedback voice quality was manipulated will also be presented. In general, the studies suggest that learning is quite local and, thus, that learning does not generalize beyond restricted conditions.

10:55

4aSC7. Speech motor learning without hearing. Sazzad Nasir and David Ostry (McGill Univ., 1205 Dr. Penfiled, Montreal, QC, Haskins Labs., New Haven, CT, nasir@motion.psych.mcgill.ca)

The nervous system receives both auditory and somatosensory information as we talk. Little is known about the role of somatosensory input in speech motor control. It is indeed a longstanding puzzle how post-lingually deaf speakers, in the absence of any substantial auditory input, maintain speech intelligibility. In this study we have assessed the role of somatosensory feedback in subjects with cochlear implants who were tested with their implants turned off. We used a robotic device to alter speech movements, and hence somatosensory feedback, by delivering mechanical loads to the jaw. We found that with training subjects corrected for the loads, such that the motion path approached that normally experienced under no-load conditions. Thus even under no auditory feedback, somatosensory input mediates speech movements in post-lingually deaf adults. In this study all five subjects with cochlear implants showed varying degrees of adaptation compared to only four out of six subjects in an age matched control group. This suggests a prominent role for somatosensory feedback in individuals with hearing loss. Finally, movement kinematic patterns for subjects with hearing loss were no different than those of subjects with normal hearing, which further underscores the reliance on somatosensory feedback in deaf adults.

11:20

4aSC8. Motor and sensory adaptation following auditory perturbation of /s/ production. Douglas M. Shiller, Marc Sato, Vincent L. Gracco, and Shari R. Baum (School of Commun. Sci. & Disord., McGill Univ., 1266 Pine Ave. W., Montreal, QC, H3G 1A8 Canada, doug.shiller@mail.mcgill.ca)

Studies of motor adaptation have provided valuable insights into the role of auditory and orosensory feedback in speech production. While perturbations to the structure of the vocal tract (e.g., pseudopalate), which alter both orosensory and auditory feedback, have been used to investigate the production of a range of consonants and vowels, studies of short-term speech adaptation utilizing purely auditory feedback manipulations (e.g., spectral shifting) have been mostly limited to investigations of vowel production. These studies have revealed that motor plans underlying vowel production are flexible, and that subjects will readily alter their motor output in order to achieve auditory perceptual targets. In the present study, we have extended this work by spectrally shifting the acoustic signal for the sibilant /s/ during the production of short /s/-initial words. Evidence of motor adaptation was observed following a brief, intensive period of practice under altered auditory conditions, indicating a role for auditory feedback in sibilant production. In addition to changes in speech motor output, subjects showed changes to their auditory-sensory representations of /s/ and /ʃ/ following the manipulation, suggesting that adaptation was not limited to the motor domain. These results will be discussed in the context of current models of speech motor control.

11:45–12:00
Discussion

Session 4aSP

Signal Processing in Acoustics, Acoustical Oceanography, and Underwater Acoustics: Session Honoring Leon Sibul I

R. Lee Culver, Chair

Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804-0030

Chair's Introduction—8:00

Invited Papers

8:15

4aSP1. A turbo approach to distributed detection and estimation. Michael Roan (Virginia Polytechnic Inst. and State Univ., Blacksburg, VA 24060)

A framework, based on the turbo principle, for distributed detection and estimation (DDE) using data from multiple sonar sensors is developed. Through the use of a Bayesian framework for DDE, an ideal observer is derived. This distributed observer makes optimal decisions, interpretations, or inferences based on measurements made at multiple sensors that observe the same scene. In this approach, a posteriori estimates at one sensor become prior probabilities for the next sensor. The posterior probability is proportional to the product of the probability of each possible state of the environment before observation (prior density) and the probability of the measurements given the state of the environment. It is the posteriors and the likelihoods representing knowledge about the environment that are shared among sensors. This iterative approach is mathematically similar to turbo coding in communications applications. This talk provides the background on the turbo DDE engine, and approaches for both practical implementation and performance quantification. Examples using Gaussian assumptions on the signal and noise distributions are given using simulations. The optimum approach is shown to be very good under ideal conditions, but not robust to sensor positioning errors. Sub-optimum approaches to improve robustness are introduced.

8:35

4aSP2. The impact of acoustic signal models on time-difference-of-arrival/frequency-difference-of-arrival estimation. Mark L. Fowler (Dept. of Elec. and Computer Eng., Binghamton Univ., Binghamton, NY 13902, mfowler@binghamton.edu) and Xi Hu (Binghamton Univ., Binghamton, NY 13902)

Much research has been done in the area of estimating time-difference-of-arrival (TDOA) and frequency-difference-of-arrival (FDOA) and their use in locating a radiating source. Early work in this area focused on locating acoustic sources using passive sonar processing. Only later was TDOA/FDOA-based location considered for the case of passively locating electromagnetic sources. As a result of this, it is tempting to use results derived for the acoustic case when answering questions about the electromagnetic case. This paper shows that such borrowing can lead to incorrect results. The key factor that drives the significant differences between these two cases is the difference between the signal model assumptions for the two cases: WSS Gaussian process is widely used as a valid model in the acoustic case, but is not usually appropriate in the electromagnetic case. Although the received signal equations may look identical (showing delay and Doppler shift), the resulting Fisher information, Cramer-Rao bound, and maximum likelihood estimator (MLE) are fundamentally different for the two signal scenarios. Thus, we show that there are distinct structures in processing and bounds that arise specifically due to using models appropriate for acoustic signals.

8:55

4aSP3. Model-based array signal processing: A tribute to Dr. Leon H. Sibul. James V. Candy (P.O. Box 808, L-156, Univ. of California, Lawrence Livermore Natl. Lab., Livermore, CA 94551) and Edmund J. Sullivan (EJS Consultants, Portsmouth, RI 02871)

Dr. Leon Sibul's contributions to acoustical signal processing are extensive and highly significant. They are marked by a deep intuition and foresight, leading to new and fruitful concepts, especially in underwater acoustic processing. In this paper we discuss how Dr. Sibul's contributions to adaptive array processing provided the conceptual basis for many of the modern approaches of model-based processing in ocean acoustics. In particular, we show how they were extended to incorporate sophisticated acoustical propagation models. Specifically, we discuss how his incorporation of propagation models in adaptive array processing schemes provided a new paradigm for such problems as source detection and localization. His concepts of an underlying physics-based estimation structure directly influenced efforts to develop environmentally adaptive processors that are capable of providing solutions to detection and localization problems. In this paper, we concentrate specifically on the model-based processing approach to the detection and localization problem, demonstrating how this area has progressed.

9:15

4aSP4. A local particle-like approach to wave propagation. Leon Cohen (City Univ. of New York, 695 Park Ave., New York, NY 10021)

Waves and particles are the main constituents of nature, but waves are our main method of communication, whether biological or technological. Historically, the study of waves has been separated into the stationary (standing wave) and nonstationary (pulse) case. Pulses are of fundamental consideration in radar, sonar, acoustics, fiber optics, among many other areas. A pulse has been called by many terms—transient, wave group, progressive wave, wave packet, nonrecurrent wave, traveling wave, nonperiodic wave—all to describe basically the same thing. The fact that so many names exist is an indication that there is no full understanding of their behavior. We present a simple particle-like view of pulse propagation, which makes pulses easier to understand and deal with intuitively. We have derived exact expressions for many physical quantities such as the spreading of a pulse, the conditions as to when a pulse contracts, the contraction time, and other important physical quantities. What emerges from these considerations is not only a clearer view of how pulses propagate, but a practical means to obtain their propagation in a dispersive medium.

Contributed Papers

9:35

4aSP5. Propagation-invariant classification of sonar signals. Patrick J. Loughlin and Greg Okopal (Dept. of Elec. and Comput. Eng., Univ. of Pittsburgh, 348 Benedum Hall, Pittsburgh, PA 15261)

Underwater sound waves can undergo frequency-dependent effects, such as dispersion and damping, that cause the wave to change as it propagates. In active sonar, these changes can be detrimental to automatic classification because the observed backscatter depends on the propagation environment and how far the wave has traveled. We aim to develop signal processing techniques to enable propagation-invariant classification. Toward that end, we have developed a local phase space view of propagation with dispersion and damping that is physically illuminating and suggests new features for classification. In this talk, we will present moment-like features extracted from the backscattered wave that are invariant to dispersion and damping, per mode. Environmental knowledge is not needed, beyond knowing the general form of the damping. Two forms of damping are considered. Classification results on simulations of the backscatter from different steel shells propagating in a Pekeris waveguide with damping and random variations will be presented. A dispersion-invariant matched-filter receiver approach will also be presented. [Supported by ONR Grant No. N00014-06-1-0009.]

9:50

4aSP6. Improving spatial resolution of a time-reversed focus using frequency-domain signal processing. Brian E. Anderson, Michele Griffa, T. J. Ulrich, Paul A. Johnson (Geophys. Group, MS D443, Los Alamos Natl. Lab., Los Alamos, NM 87545, bea@lanl.gov), and Robert A. Guyer (Univ. of Massachusetts, Amherst, MA 07030)

In the typical application of time reversed acoustics, the spatial resolution of a time-reversed focus is set by the diffraction limit, i.e., a focal spot size cannot be made smaller than a half of a wavelength. A frequency-domain signal processing method is introduced in which the information in the received signal of the forward propagation stage is shifted to higher frequency. The resulting time domain signal is time reversed and rebroadcast to form a focal spot size that is diffraction limited at the higher frequency (shorter) wavelength. This method requires neither a priori information about the original source location nor nearfield signal detection. A theoretical discussion of the problem, along with numerical and experimental results, will be presented. The limitations and potential applications will also be presented. [This work is supported by Institutional Support (LDRD) at Los Alamos National Laboratory.]

10:05–10:20 Break

10:20

4aSP7. Selective source reduction to identify masked sources using time-reversed acoustics. Brian E. Anderson, T. J. Ulrich, Michele Griffa, Paul A. Johnson (Geophys. Group, MS D443, Los Alamos Natl. Lab., Los Alamos, NM 87545, bea@lanl.gov), Marco Scalerandi, and Antonio S. Gliozzi (CNISM, Politecnico di Torino, 10129 Torino, Italy)

This paper describes a time-reversed acoustics (TRA) method of spatially illuminating a source signal, which has been masked by another source signal. The selective source reduction (SSR-TRA) method employs a subtraction technique where one focus is selectively reduced to illuminate the masked focus. In this paper, numerical and experimental results are presented to demonstrate to what degree the SSR-TRA method is successful. The SSR-TRA method is demonstrated for two elastic wave pulses emitted simultaneously from two spatially separated sources of differing amplitudes. Results are presented from experiments conducted with two different solid samples: Aluminum and doped silica glass. Applying the SSR-TRA method, a stronger source, up to 13 times stronger than a weaker one, may be reduced to reveal information about the weaker source. Spatial and/or temporal characteristics of multiple close proximity sources can be resolved with the use of the SSR-TRA method. The results show that the SSR-TRA methods' limitations are chiefly due to imperfect reconstruction of the source function in the time-reversed focal signal. [This work was supported by Institutional Support (LDRD) at Los Alamos National Laboratory.]

10:35

4aSP8. Identifying and integrating likelihood functions and signal parameter probability density functions (PDFs) for sonar signal processing. Richard Lee Culver, Colin W. Jemmott, and Jeffrey A. Ballard (ARL and Grad. Program in Acoust., Penn State Univ., PO Box 30, State College, PA 16804)

A likelihood ratio receiver developed using an estimator-correlator structure requires that the noise probability density function (pdf) belong to the exponential class, but need not be Gaussian. The receiver employs a composite hypothesis that takes into account knowledge (or predictions) of the signal parameter statistics. In order to implement the receiver, we must be able to integrate the a priori pdf for the random parameter times the likelihood function over the domain of the random parameter, i.e., $\int p(r-\theta)p(\theta)d\theta$. The signal fluctuation models used by Swerling (1960) for radar detection theory belong to the chi-squared family of distributions, which are part of the exponential class. Likewise, the maximum entropy method with moments $\ln\theta$ and θ could be used to fit $p(r-\theta)$ with a chi-squared pdf. If both noise and signal parameters are chi-squared distributed, a closed-form solution may exist. This paper considers approaches to fitting pdfs to noise and signal parameter samples and to solving the integral described above. [Work sponsored by ONR Undersea Signal Processing.]

10:50

4aSP9. A coded signal technique for the study of environmental noise propagation. Jamie A. Angus, David C. Waddington, and Phil J. Duncan (Acoust. Res. Ctr., School of Computing, Sci. and Eng., Univ. of Salford, Salford M5 4WT, UK)

This paper describes the design of a coded acoustic signal for the study of environmental sound propagation. An increasing awareness of environmental noise problems, and the need to better specify their needs and extent, means that a better knowledge of the effects of weather on sound propagation is necessary. Low signal-to-noise ratio is a common limitation in outdoor propagation investigations. A further problem is time variance. A coded acoustic signal approach consisting of a frequency carrier biphase-modulated by a specially-designed pseudo-random code sequence is proposed to overcome these limitations. This so-called "inner and outer" code sequence is specially designed for environmental sound propagation investigations, combining simultaneous fine time resolution and large range ambiguity. Experiment measurements are used to illustrate how the amplitude of the transfer functions between the receivers together with accurate times of flight may be calculated from the cross-correlation of the measured coded acoustic signal. It is further shown that the mean effect over the total propagation channel of factors such as wind, temperature gradients, turbulence, barrier, and ground reflections can be determined by measurement of the time-varying propagation transfer function using the proposed complex impulse response correlation technique. [Work supported by EPSRC UK.]

11:05

4aSP10. Performance classification degradation due to shallow-water propagation. Charles F. Gaumond and Ralph N. Baer (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375)

The propagation-induced degradation of classification ability for echoes from a finite cylindrical shell and a from a fish school is investigated by use of the class specific method [P.M. Baggenstoss, "Class-specific feature sets in classification," IEEE Trans. Sig. Proc. **47**, 3428–3432 (1999)]. The study is performed with previously displayed numerically simulated data. In prior work, degradation from additive noise was studied at a variety of signal bandwidths. Here, convolution noise caused by two-way propagation in a shallow water environment is shown to increase the duration of non-propagated echoes. The common propagation path between the two classes causes degradation when the classifier is based on non-propagated training data. Ameliorating effects of using training data comprised of either a fixed range or multiple distinct ranges are shown. The performance is quantitatively displayed as either classifier operating characteristic (CLOC) curves or as the integrals of the CLOC curves. The design trade-off space of bandwidth, signal-to-noise ratio (proportional to source level), and complexity of the training set is shown over a limited range of values. [Work supported by ONR.]

11:20

4aSP11. Accuracy enhancement of underwater target detection with time-frequency analysis techniques. Abdalla Osman, Naser El-Sheimy (Mobile Multi-Sensor Systems Res. Group, Dept. of Geomatics Eng., Univ. of Calgary, Calgary, AB, Canada, amosman@ucalgary.ca), Aboelmagd Nourledin (Royal Military Coll. of Canada, Kingston, ON, K7K 7B4), Jim Theriaul, and Scott Campbell (Dept. of Natl. Defense, Canada)

Underwater target detection is mainly based on acoustic emissions generated by the target of interest (TOI) as it propels itself through the ocean, and operates non-propulsion-related onboard systems. The spectral signature of these acoustic emissions is used for identification and localization of TOI. This paper focuses on underwater target detection using passive sonobuoys. Discrete Fourier transform (DFT) is used in most sonobuoy processing systems to provide spectral analysis of the received signals. Due to the relatively high noise level and the several sources of interference that may exist underwater, the DFT may not determine the spectral signature of TOI with adequate accuracy. The low signal-to-noise ratio (SNR) and the presence of strong interference sources with frequencies close to the TOI frequency jeopardize the detection accuracy, the bearing estimation performance, and target tracking capabilities. The aim of this paper is to: (1) examine the performance of both wavelet packet analysis and high resolution spectral estimation techniques, and (2) provide a comparative study between both methods. Based on underwater acoustic simulation developed in this research, results showed that the proposed methods can achieve robust target detection with low levels of SNR and interferences of nearby signatures that cannot be detected by DFT.

11:35

4aSP12. Cued beamforming of passive arrays in the context of active-passive data fusion. T. W. Yudichak and Minh Le (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

Active-passive data fusion seeks to exploit the complementary information measured simultaneously by active and passive sonar sensors: For instance, active sensors can provide good range estimates for a contact, while passive sensors can provide information on both the bearing of and the spectrum radiated by that contact. One aspect of this subject is the use of feedback, in which prior information is collected about the state of a contact in the data fusion framework, and then exploited to enhance the signal processing of data measured in that framework. Cued beamforming is one such method, in which a dense set of passive beams are steered toward regions of high contact probability, as determined by the result of the fusion of initial active and initial passive state estimates. The denser cued beams may then provide a refinement of the position estimate of the contact, as well as better isolation of its spectral signature. Comparisons of the effect of using cued beams are compared to the effect of using standard beams (those used without prior knowledge) on the state estimate of contacts in various scenarios using simulated data. [Work supported by ONR.]

Session 4aUW

Underwater Acoustics: Underwater Reverberation Measurements and Modeling I

John S. Perkins, Cochair
Naval Research Laboratory, Code 7140, Washington, DC 20375-5350

Eric I. Thorsos, Cochair
Univ. of washington, Applied Physics Lab., 1013 NE 40th St., Seattle, WA 98105-6606

Chair's Introduction—8:00

Contributed Papers

8:05

4aUW1. Overview of the reverberation modeling workshops. John S. Perkins (Naval Res. Lab., Washington, DC 20375) and Eric I. Thorsos (Univ. of Washington, Seattle, WA 98105)

Two workshops on reverberation modeling are being conducted under joint sponsorship from the Program Executive Office C4I, PMW 180 (as funded by the Office of the Oceanographer of the Navy) and the Office of Naval Research. The overall goal of these workshops is to evaluate recent progress made in reverberation-related research efforts and to make recommendations for further development. The first workshop was held in November 2006 and the second is scheduled for March 2008. The focus of the first workshop was on reverberation from the environment, while the second workshop will emphasize system characteristics. At the first workshop, fifteen different reverberation models were represented and extensive comparisons were carried out before, during, and after the workshop. We present the approach used to conduct the first workshop, discuss issues that have been identified, and outline tentative plans for the second workshop. [Work supported by ONR and PMW 180.]

8:20

4aUW2. Scattering models for Reverberation Modeling Workshop I problems. Eric I. Thorsos (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu)

For many of the problems posed for Reverberation Modeling Workshop I, the boundary roughness was presented in terms of roughness spectra. Models and/or tables were also supplied for the bistatic scattering cross section and for the coherent reflection loss. Integral equation simulations were used to verify the accuracy of the bistatic scattering cross sections, and rough boundary PE simulations were used to verify the accuracy of the coherent reflection losses. The results of this work will be briefly summarized. The issue also arises for reverberation simulations whether the coherent reflection loss is appropriate for the boundary loss, or whether some other loss model is more appropriate, or whether the loss due to roughness should be simply ignored. This issue will be discussed in light of rough boundary PE simulations. [Work supported by ONR.]

8:35

4aUW3. Reverberation Modeling Workshop problem definitions. Kevin D. LePage (Naval Res. Lab, 4555 Overlook Ave. SW, Washington, DC 20375)

In November of 2006 the first Reverberation Modeling Workshop was held in Austin, Texas. In this talk, the problems defined for the workshop are presented, along with some selected collective results to give an idea of the level of agreement achieved between the participants. In all, 20 problems were defined, ranging from monostatic 2-dimensional range independent problems up through fully three-dimensional bistatic problems. [Work supported by ONR and PEO C41 and Space (PMW 180).]

8:50

4aUW4. Finite-element modeling of long range, range-dependent acoustic propagation in shallow water. Marcia J. Isakson, R. Abe Yarbrough (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029), and Preston S. Wilson (Univ. of Texas at Austin, Austin, TX 78712-2300)

The current methods of modeling long range, low frequency acoustic propagation in shallow water environments include ray methods, wave number integration, parabolic equations, and normal mode theory. Finite-element (FE) models have also been considered in the past, but computing power and algorithm cost have limited their application. Recent advances in computing power coupled with the advent of low cost, user-friendly, all-purpose finite-element codes have provided new opportunities for the application of FE modeling. We present both time and frequency domain solutions to canonical acoustic problems (scattering from a corrugated surface and propagation in a Pekeris waveguide), and to two-way, range-dependent problems, such as waveguides with surface and bottom roughness. For the latter environment, reverberation time series produced by Fourier synthesis of the FE model output can potentially provide benchmark solutions that no other currently available solution method can provide. [Work supported by ARL IR&D.]

9:05

4aUW5. Parabolic equation techniques for reverberation modeling. Joseph F. Lingevitch, David M. Fromm, and Kevin D. LePage (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375)

Recent developments of parabolic equations for modeling reverberation will be presented. This method is applicable to the 2-D reverberation problems with general sound speed profiles, sediment layering, and interface roughness. The environment is sub-divided into range-independent regions, and the forward and back-scattered fields are coupled at each interface with a single-scattering approximation. This method incorporates a cumulative scattering loss into the forward field, but neglects the non-local coupling terms between the forward and backward scattered fields. The back-scattered field is stored at each interface on a forward sweep through the environment and back-propagated on backward sweep. Reverberation time series are Fourier synthesized from the frequency domain solutions. This technique is fundamentally different than scattering kernel based approaches and provides an independent method for obtaining the forward and back-scattered solutions for reverberation problems. As currently implemented, this model does not include multiple scattering effects, but the algorithm may be generalized to include multiple scattering with iterative sweeps through the propagation domain. The method applied to 2-D reverberation problems from the recent Reverberation Modeling Workshop (November 2006) and results are compared with ray and normal-mode based models. [Work supported by ONR.]

9:20

4aUW6. A mostly time-domain reverberation model with application to Reverb Workshop Problem II. Richard Keiffer (Naval Res. Lab., Stennis Space Ctr., MS 39529, keiffer@nrlssc.navy.mil)

A mostly time-domain reverberation model is described and applied in an isospeed shallow water ocean waveguide that has a flat sandy bottom and a rough air/sea boundary (Reverb Workshop Problem II) to predict the received intensity averaged over many realizations of the sea surface. The model described computes the scattering using a time-domain scattering approach called the wedge assemblage method and accounts for volume attenuation and bottom reflection loss under a ‘‘constant Q’’ approximation. Details of the model are discussed and results are presented for both the point and line source excitation of the waveguide. [This work was supported by ONR through NRL.]

9:35

4aUW7. An investigation into the validity of the Born approximation in reverberation calculations. David P. Knobles, Steven A. Stotts, Marcia J. Isakson, and Preston S. Wilson (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713)

Ocean reverberation models for active sonar applications assume the validity of the Born or single scatter approximation. In many areas of physics, a Born series does not converge due to a combination of the scattering strength and the spatial scale over which the scattering occurs. In these cases, the validity of the Born approximation is at best questionable. It is natural to attempt to examine the validity of the Born approximation and the convergence of the Born series in general for well-defined scattering problems in ocean acoustics. The problems examined in this study involve scattering from specific realizations of bottom roughness for 2-D waveguides. This study examines several idealized problems from the 1st Reverberation Workshop in November 2006 for the purpose of comparing a Born approximation solution to an exact solution provided by a 2-way coupled-mode integral equation approach. The coupled integral equations are solved exactly to within the limitations of finite differences and numerical quadrature. The exact solutions are then compared to those that solve the fundamental integral equation using the Born approximation and to solutions obtained with a finite-element model. [Work supported by ARL:UT IR&D.]

9:50

4aUW8. Rayleigh reflection loss non-linearity approximation and application to reverberation. Chris Harrison (NURC, Viale S. Bartolomeo, 400, 19126 La Spezia, Italy, harrison@nurc.nato.int)

This paper presents an extremely good approximation to the Rayleigh reflection loss for two half-spaces composed of water over sediment. This has some useful properties. First it demonstrates that the usual linear approximation for the reflection loss (inside the critical angle) can be extended to the nonlinear case. Second it shows that the non-linearity can be expressed as a separate function that multiplies the linear loss coefficient. Third this nonlinearity term depends only on sediment density and does not depend on sediment sound speed or volume absorption. Fourth the nonlinearity term tends to unity, i.e., the reflection loss becomes effectively linear, when the density ratio is about 1.27. There are similar findings for the reflection loss phase which lead to the well-known lateral shift or effective depth. A class of closed-form reverberation (and signal-to-reverberation) expressions has been developed at NURC, [Harrison (2003), JASA, 114, 2744–2756; Harrison (2005), JCA, 13, 317–340; Harrison (2005), JOE, 30, 660–675]. The new approximation enables one to convert these reverberation expressions from simple linear loss to much more general reflecting environments. Correction curves are calculated in terms of sediment density and applied to a test case taken from a recent ONR Reverberation Workshop (Nov 2006).

10:05–10:20 Break

10:20

4aUW9. A comparison of ray, normal-mode, and energy flux results for reverberation in a Pekeris waveguide. Dale D. Ellis (DRDC Atlantic, P.O. Box 1012, Dartmouth, NS Canada B2Y 3Z7, dale.ellis@drdc-rddc.gc.ca), M. A. Ainslie (The Hague, The Netherlands), and C. H. Harrison (NATO Underwater Res. Ctr., La Spezia, Italy)

A number of problems were developed for, and presented at, a 2006 Reverberation Modeling Workshop sponsored by the US Office of Naval Research. The simplest of these known to the participants as Problem 11, was a Pekeris waveguide (isospeed water over a flat bottom half-space) with Lambert bottom scattering. The water depth was 100 m and frequencies of 250, 1,000, and 3,500 Hz were specified. A number of source-receiver combinations were specified, but the reverberation predictions are quite insensitive to sensor depth except at 250 Hz. With some benefit from hindsight and the results from other models, we compare our results from ray, normal-mode, and energy-flux approaches. All three approaches agree at intermediate times, say 3 to 50 s. At short times, the steep-angle paths and fathometer returns cause the mode and energy-flux models to underpredict the reverberation. At longer times, the ray models run out of steam: i.e., there are too many contributing ray paths for them to handle so they underpredict the reverberation. By combining the model predictions together with analytical results from an energy-flux model, we propose a composite benchmark solution. [Work supported in part by ONR.]

10:35

4aUW10. Issues associated with solutions to a rough-bottom Pekeris waveguide. Steven A. Stotts, David P. Knobles, and Robert A. Koch (Appl. Res. Labs., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78758)

The solution to reverberation time series in a 2-D shallow-water ocean waveguide with bottom roughness (c.f. Problem 1, Reverberation Workshop 2006) has been obtained using both plane wave Born approximation (PWBA) and two-way coupled-mode (CM) approaches. The source and receiver are range monostatic. The CM solutions obtained can be used to determine the accuracy of the Born approximation. Specified time intervals of the received field from the CM solution are correlated and compared to both the patch size and bottom roughness correlation length in the PWBA solution, thus connecting the two approaches. The minimum number of bottom interactions required for accuracy in the PWBA approach is determined. Propagation effects associated with time bin averaging in the PWBA approach are also discussed. The importance of including continuum components in the CM approach, via leaky modes evaluated in the complex horizontal wave-number plane, is also described. Convergence of the CM solution in an expansion of an orthogonal vector space is determined. [Work supported by ARL:UT IR&D.]

10:50

4aUW11. A Matlab and normal mode based reverberation model and some data/model comparisons. John R. Preston (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804) and Dale D. Ellis (DRDC Atlantic, Dartmouth, NS, B2Y 3Z7 Canada)

A Matlab and normal mode based reverberation model has been developed that uses Ellis' algorithm [J. Acoust. Soc. Am. **97**, 2804–2814] for reverberation predictions and the ORCA normal mode model [Westwood et al., J. Acoust. Soc. Am. **100**, 3631–3645] to compute the eigenvalues and mode functions. The model is currently range independent, but handles bistatic geometries, and towed array beam patterns. Group velocity corrections similar to LePage's [J. Acoust. Soc. Am. **106**, 3240–3254] have recently been added to this model. The matrix formulation in Matlab allows the bistatic calculations to be performed more efficiently. An overview of the model is presented that includes sample model-model comparisons for some problems from the 2006 ONR Reverberation Workshop. The model predictions are also compared with data obtained with the NURC (formerly the NATO Undersea Research Centre, SACLANTCEN) towed arrays during several reverberation experiments on the Malta Plateau. Select data from these experiments are used to compare with model predictions using best estimates of bottom properties in the area. [Work supported by US Office of Naval Research, Code 321OA, Grant No. N00014-05-1-0156.]

11:05

4aUW12. The importance of retaining inter-path coherence in reverberation prediction. Kevin D. LePage (Naval Res. Lab, Code 7144, 4555 Overlook Ave. SW, Washington, DC 20375)

The R-SNAP and BiStaR normal mode-based reverberation models are used to solve a selection of problems from the 1st Reverberation Modeling Workshop held in Austin, TX in November of 2006. R-SNAP and BiStaR are codes that include the effects of inter-mode decorrelation to predict the coherent aspects of reverberation, as well as allowing for the option to compute the ''incoherent'' reverberation. Comparisons between the incoherent R-SNAP and BiStaR predictions and model predictions obtained by other methods show good agreement. However, it is argued here that as defined, the problems require a coherent solution, and differences between the incoherent and coherent solutions can be significant. Comparisons with coherent 2-way PE results for problem I computed by Joe Lingevitch (NRL) show that coherent structure in reverberation exists, is predictable, and should therefore be an integral part of any reverberation calculation. [Work supported by ONR.]

11:20

4aUW13. Propagation modeling via ray bundles. Gary H. Brooke (General Dynam. Canada Ltd., 3785 Richmond Rd., Ottawa, ON, Canada K2H 5B7)

GD Canada has developed an acoustic propagation model, GDRAY, based on a Gaussian ray bundle propagation engine similar to that used in the CASS/GRAB performance prediction software [H. Weinberg and R.E. Keenan, J. Acoust. Soc. Am. **100**, 1421–1431 (1996)]. The GRAB-type model has proved to be useful for a wide-range of underwater acoustic scenarios, including active and passive prediction in shallow and deep water over a wide range of frequencies. Surprisingly, despite its extensive use in USN applications, the theoretical origins of the GRAB model are not well understood. This paper is intended to examine those origins and to outline some of the unique features of the GDRAY model implementation. It will be shown that Gaussian ray bundles, as such, have no particular physical significance in terms of the propagation, but are simply one of a family of possible ray bundle types that can be used to extract eigenray information. Several simple test cases will be employed to demonstrate the underlying concepts and issues. [Work is funded as an internal R&D program at GD Canada.]

11:35

4aUW14. An update on the multi-static model, SPADES, and its application to the reverberation workshop test suite. Gary H. Brooke (General Dynam. Canada Ltd., 3785 Richmond Rd., Ottawa, ON, Canada K2H 5B7)

GD Canada has developed a multi-static sonar performance prediction capability called SPADES. SPADES was applied to the Reverberation Workshop test cases [November 9–13, 2006, Austin, TX] and reported consistent behavior with other models in all cases except those involving refraction. This paper outlines recent progress with the SPADES model to include multi-static target echo, Nx2D propagation environments, and its current status with respect to the reverberation workshop test suite. In particular, it is shown that the model now aligns closer with the norm in all of the test cases. Sensitivities in the SPADES model to workshop problem parameters are detailed with a view to understanding the remaining discrepancies. Included also are new results for range-dependent test cases. [Work is funded as an internal R&D program at GD Canada.]

FRIDAY AFTERNOON, 30 NOVEMBER 2007 GRAND BALLROOM E, 1:30 TO 2:50 P.M.

Session 4pAA

Architectural Acoustics, Noise, and ASA Committee on Standards: Acoustics of Modular Construction

Edward C. Duncan, Chair

Resource Systems Group Inc., 55 Railroad Row, White River Junction, VT 05001

Chair's Introduction—1:30

Invited Papers

1:35

4pAA1. Acoustics of modular construction—Industry overview. Thomas E. Hardiman (944 Glenwood Station Ln., Ste. 204, Charlottesville, VA 22980, tom@modular.org)

This session will provide an overview of the issues and efforts impacting the commercial modular construction industry throughout North America, with particular focus on acoustics in relocatable classrooms. The Modular Building Institute is the international nonprofit trade association representing manufacturers and dealers of commercial modular facilites, both temporary and permanent, serving educational, health care, retail, industrial, military, and multi-family markets.

1:55

4pAA2. Issues in acoustic field testing of quiet modular classrooms. David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683) and Louis C. Sutherland (Acoust., Rancho Palos Verdes, CA 90275)

Modular classrooms are important to American education: About 300,000 modular classrooms are currently in use by public school systems here. Good acoustical conditions for learning are no less vital for students in modular classrooms than stick-built classrooms. In an effort to promote good acoustics in modular classrooms, ANSI S12 Working Group 46 is seeking to standardize acoustic field testing. Their efforts are in response to key acoustical issues of modular classrooms: Excessive noise from HVAC (heating ventilating and air conditioning) systems, and poor airborne sound insulation from exterior noise sources. In a recent and notable advance, an HVAC system provider reported good progress in modular HVAC noise reduction: A ducted wall mounted system was used instead of the usual free blowing system with exposed fans. HVAC noise in the unoccupied room was near the maximum 35 dB level required by ANSI S12.60. Interior noise levels were so low that efforts to confirm their values were confounded by noise contamination from exterior sources. The relatively high interior ambient noise levels were due to poor airborne sound insulation. Lessons learned from recent field testing will be discussed. Results of airborne sound insulation tests, now in planning stages, will be reported if available.

2:15

4pAA3. Performance and design considerations for modular panel systems used for studios and acoustic test chambers. Douglas F. Winker (Acoust. Systems, a Div. of ETS-Lindgren, 1301 Arrow Point Dr., Cedar Park, TX 78613, Douglas.Winker@ets-lindgren.com)

Modular panel construction is used to produce a variety of rooms including studios, medical rooms, hemi- and full-anechoic chambers, and reverberation chambers. Prefabricated panels provide many advantages over conventional construction, including the ability to relocate chambers, minimize on-site construction time, and maintain panel-to-panel performance through tight quality control, while maintaining the ability to customize an enclosure for a particular need. The design and performance of the panel systems will be discussed as they relate to their intended applications. The performance of each panel system was evaluated for both normal incidence and random incidence sound absorption in accordance with ASTM E1050 and C423, respectively. The panel's sound transmission loss was tested in a reverberation chamber suite in accordance with ASTM E90. Fully functioning systems were tested by ASTM E596 noise reduction of an isolating enclosure in one reverberation chamber. The performance of the modular panel systems will also be compared to conventional construction methods.

Contributed Paper

2:35

4pAA4. Noise in modular dwellings. Sergio Beristain (IMA, E.S.I.M.E., I.P.N., P.O. Box 12-1022, 03001, Mexico, D. F., Mexico)

Construction of modular dwellings in Mexico usually has had the problem of the joints; examples are given in some cases where this problem shows up as a low-quality construction, due to bad joints that no one will see. Sometimes, in order to make low cost construction, with the posibility of selling high, modular isolating panels are employed, but the details are seldom observed. Builders do not care to hire an acoustician, and there are infrequent complaints by owners.

FRIDAY AFTERNOON, 30 NOVEMBER 2007 MAUREPAS, 1:30 TO 3:20 P.M.

Session 4pBB

Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Biological Effects and Medical Applications of Stable Cavitation II

Saurabh Datta, Chair

Univ. of Cincinnati, Dept. of Biomedical Engineering, 231 Albert Sabin Way, Cincinnati, OH 45242-0586

Invited Papers

1:30

4pBB1. Stable cavitation in ultrasound image-guided high intensity focused ultrasound therapy. Shahram Vaezy, Wenbo Luo, Michael Bailey, Lawrence Crum (Univ. of Washington, Box 355061, Seattle, WA 981195), Brian Rabkin (Joint Genome Inst.), and Vesna Zderic (George Washington Univ.)

Microbubble activity is significantly involved in both diagnostic and therapeutic aspects of ultrasound image-guided HIFU therapy. Ultrasound interrogation techniques (A-, B-, M-mode, Doppler, harmonic and contrast imaging, and passive and active cavitation detection) were integrated with HIFU. Our results using HIFU devices of 1–5 MHz, and focal, derated intensities of 1,000–10,000 W/cm^2, show the formation of microbubbles (about 100 bubbles/mm^3, 5–100 microns in size) at the HIFU focus. Boiling, stable, and inertial acoustic cavitation activities were detected during therapy. The presence of bubbles allows the observation of the treatment spot as bright hyperechoic regions in ultrasound images, providing an effective method for guidance and monitoring of therapy. The stable cavitation of microbubbles may provide a mechanism for enhanced HIFU energy delivery, as well as induction

4p FRI. PM

of biological responses for stimulation and regulation of specific physiological events such as coagulum and thrombus formation for hemostasis applications, apoptotic activity in treating tumor margins, and stimulation of immune response. Stable cavitation of extrinsic bubbles (contrast agents) is used in detection and localization of internal occult bleeding, using harmonic imaging. There appears to be benefits in utilizing stable cavitation in both diagnostic and therapeutic objectives of ultrasound image-guided HIFU. Funding: DoD, NIH, NSBRI.

1:50

4pBB2. Cavitation-enhanced ultrasound heating in vivo: Mechanisms and implications in MR-guided high-intensity focused ultrasound therapy. Shunmugavelu Sokka (Philips Medical Systems, 3000 Minuteman Rd., MS 027, Andover, MA 01810) and Kullervo Hynynen (Sunnybrook Health Sci. Ctr., Toronto, Canada, ON M4N 3M5)

Focused ultrasound is currently being developed as a noninvasive thermal ablation technique for benign and cancerous tumors in several organ systems. Although these therapies are designed to ablate tissue purely by thermal means, cavitation can occur. These bubbles can be unpredictable in their timing and location, and often interfere with thermal therapies. Therefore, focused ultrasound techniques have tried to avoid bubbles and their effects. However, gas bubbles in vivo have some potential useful features for therapy. In this research, we design and test in vivo ultrasound exposures that induce cavitation at appropriate times and take advantage of their absorption-enhancing properties while yielding reliable lesion sizes and shapes. In addition, MRI and acoustic methods to monitor and potentially control cavitation induction and the associated therapy are investigated. Finally, histology of the resulting cavitation-enhanced heating lesions is performed to assess the type of tissue damage. We demonstrate that cavitation-enhanced heating can be reliable and useful with the appropriate therapy protocol and application. If induced and monitored properly, cavitation in focused ultrasound therapy could potentially be very beneficial. Early MR-guided HIFU clinical systems that can utilize and monitor cavitation approaches will be presented.

2:10

4pBB3. Therapeutic potential of stable cavitation: From enhanced drug delivery to faster hemorrhage control. Vesna Zderic (Dept. of Elec. and Comput. Eng., The George Washington Univ., 801 22nd St. NW, Washington, DC 20052), Shahram Vaezy, and Wenbo Luo (Univ. of Washington, Seattle, WA)

Current studies indicate that mechanical effects of therapeutic ultrasound resulting in a cavitation activity may be advantageous in variety of clinical applications including tumor treatment, treatment of stroke and vascular diseases, targeted gene and drug delivery, and hemorrhage control. Our drug delivery studies using 880 kHz ultrasound have shown that enhancement of drug delivery through the cornea correlated well ($R\hat{2}=0.92$) with stable cavitation activity. The delivery enhancement ranged from two to ten times and the power of subharmonic ranged form 5 dBm to 20 dBm in the intensity range of 0.2 0.6 W/cm^2. Broadband noise, as an indication of inertial cavitation, was also detected at the highest applied intensity. Changes in the front surface layer of the corneal epithelium indicated the presence of both stable and inertial cavitation activity. Our hemostasis studies have shown that introduction of external microbubbles during application of 5.5 MHz high−intensity focused ultrasound (HIFU) led to faster hemorrhage control of solid organ injuries, appearing to result from both stable and inertial cavitation activity at the location of HIFU focus in a bleeding incision. Microbubbles also allowed easier targeting of an incision site under ultrasound guidance, thus facilitating faster localization and sealing of bleeding incisions.

2:30

4pBB4. Image-guided cavitational ultrasound therapy histotripsy. Zhen Xu, Timothy L. Hall, Charles A. Cain (Dept. of Biomed. Eng., Univ. of Michigan, Ann Arbor, MI 48109), J. Brian Fowlkes, and Williman W. Roberts (Univ. of Michigan, Ann Arbor, MI 48109)

Our recent investigations have demonstrated that mechanical tissue fractionation can be achieved using successive short (3–50 μs), high intensity ultrasound pulses delivered at low duty cycles (0.1–5%). This technique can be considered as soft tissue lithotripsy, which we call histotripsy. The acoustic pressures effective for histotripsy are similar to those used in lithotripter shockwave pulses. In bulk tissue, histotripsy produces extensive fractionation of cellular structures, resulting in a liquid cavity within the treated volume. At a tissue-fluid interface, histotripsy physically removes tissue. Histotripsy lesions have sharp boundaries, with several microns between intact and fractionated cells. The primary mechanism of histotripsy is thought to be cavitation. The production of tissue fractionation depends on initiation and maintenance of cavitation detected by acoustic and optical monitoring. Cavitation bubbles generated by histotripsy are easily detected on ultrasound imaging, which is used for targeting and real-time monitoring of the treatment progress. Histotripsy induced lesions are also clearly identifiable using ultrasound and magnetic resonant imaging. Our investigations suggest that histotripsy has potential to be developed into a new noninvasive ablation tool for many clinical applications where tissue ablation, remodeling, or removal is needed. [This research is supported by grants from NIH R01-HL077629 and Wallace Coulter Foundation.]

2:50

4pBB5. Observations of cavitation and boiling in a tissue-mimicking phantom due to high intensity focused ultrasound. Michael S. Canney, Wayne Kreider, Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), Vera A. Khokhlova (Moscow State Univ., Moscow 119992, Russia), and Lawrence A. Crum (Univ. of Washington, Seattle, WA 98105)

Bubbles generated by acoustic cavitation or boiling are often observed during high intensity focused ultrasound (HIFU) medical treatments. In this work, high-speed video imaging, a 20-MHz focused acoustic transducer, and the driving voltage to our 2-MHz HIFU source are used to distinguish between cavitation and boiling in a tissue-mimicking gel phantom at peak focal intensities up to 30,000 W/cm^2. Bubble dynamics are modeled using a reduced order model that accounts for evaporation and condensation, heat and gas transfer across the interface, and temperature changes in the surrounding liquid. The model includes vapor trapping, whereby the noncondensable gas slows diffusion of vapor to the interface, thereby limiting condensation. At the transducer focus, evidence of cavitation is observed in the first millisecond before disappearing. Boiling is observed several milliseconds later, after sufficient heating of the focal volume to 100 °C. The disappearance of cavitation can be explained in part by the observed motion of bubbles away from the focal region due to radiation-pressure forces and in part by the softening of bubble collapses by vapor trapping. Thus, at clinical HIFU amplitudes, bubble dynamics and their impact on image-feedback and/or therapy change dramatically in only milliseconds. [Work supported by NIH DK43881 and NSBRI SMS00402.]

3:05

4pBB6. Magnetic resonance imaging of boiling induced by high intensity focused ultrasound. Tatiana D. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105 and Dept. of Optics, Phys. Faculty, Moscow State Univ., Moscow, Russia), Michael R. Bailey, Michael S. Canney, Vera A. Khokhlova, Donghoon Lee, and Kenneth I. Marro (Univ. of Washington, Seattle, WA 98105)

Bubble activity in high intensity focused ultrasound (HIFU) medical therapy is commonly but not rigorously divided between mechanically induced cavitation (μ size gas bubbles) and thermally induced boiling (mm size vapor bubbles). Our goal was to confirm that boiling occurred at 100 °C. A 2 MHz focused transducer (42 mm aperture, 44 mm focal length) was used to heat tissue phantoms in a 4.7 Tesla magnet. Temperature was measured by magnetic resonance imaging (MRI) proton resonance frequency shift and calculated from acoustic absorption. The MRI voxel was $0.3\times0.5\times2$ mm, and acquisition time was 1.3 s. Boiling was observed as a dark spot in MRI images and fluctuation in the transducer drive voltage. At 30 MPa peak pressure, boiling occurred in 7 s. Calculations yielded 100 ^{circ}C in 7 s and a temperature half maximum width of 1 mm. Averaging the calculated temperature field over the MRI voxel yielded a maximum of 73 ^{circ}C, which was the peak temperature measured in the last MRI slice before boiling. In conclusion, boiling appeared when the peak temperature reached 100 °C, and the results warn that MRI monitoring alone may underestimate the peak temperatures. [Work supported by NIH DK43881, NIBIB R21EB005250, and NSBRI SMS00402.]

FRIDAY AFTERNOON, 30 NOVEMBER 2007 GRAND COUTEAU, 1:30 TO 5:30 P.M.

Session 4pMU

Musical Acoustics and Speech Communication: Session in Honor of Max Mathews with Electronic and Computer Music Concert

Thomas D. Rossing, Cochair
Stanford Univ., CCRMA, Dept. of Music, Stanford, CA 94305

Julius O. Smith, III, Cochair
Stanford Univ., CCRMA, Dept. of Music, Stanford, CA 94305

Chair's Introduction—1:30

Invited Papers

1:35

4pMU1. A study of ambiguities in the acoustic-articulatory mapping. Bishnu Atal (Dept. of Elec. Eng., Univ. of Washington, Seattle, WA 98185)

At Bell Laboratories, in 1976, Max Mathews, John Tukey, J. J. Chang, and I investigated the relationship between the shape of the vocal tract and its acoustic output for studying articulatory compensation. Although the mathematics of the forward transformation, from the vocal-tract shape to its acoustics, is reasonably clear, the same cannot be said about the inverse transformation from acoustics to the corresponding vocal-tract shape. Max, recognizing the increasing power of digital computers, suggested an interesting approach to the inverse problem, based on computer-sorting. Inversion by computer-sorting consisted of computing the output for many values of the input and sorting the resulting output-input pairs into a convenient order, according to the output data. The paper will highlight this approach and present some interesting results of this study on articulatory compensation. We were successful in determining articulatory regions (fibers) that map into a single point in the acoustic space and produce the same sound [Atal et al., J. Acoust. Soc. Am. **63**, 1535–1555 (1978)].

2:00

4pMU2. Max V. Mathews, the computer music pioneer. F. Richard Moore (Dept. of Music, Univ. of California, San Diego, La Jolla, CA 92093-0326, frm@ucsd.edu)

Max Vernon Mathews (born November 13, 1926) is the pioneer of computer music sound synthesis and control. After receiving a Sc.D in electrical engineering from M.I.T. in 1954, Mathews directed the Acoustical and Behavioral Research Center at Bell Laboratories at Murray Hill, New Jersey from 1962 to 1985, where he designed and implemented the first important computer programs to synthesize musicals sounds, starting with MUSIC I, in 1957. Mathews' ingenious strategies traversed many computer generations of improvement, and now form the backbone of much of the music synthesis industry. In 1968, Mathews turned to real time control of musical synthesis, resulting in the GROOVE system, which, like the MUSIC programs, provided a conceptual basis for further developments in computer-based musical performance. Mathews has influenced many musicians and researchers, including Pierre Boulez, who based major research aspects of his *Institute de Recherche et Coordination Acoustique/Musique* (IRCAM) in Paris on the work of Mathews and his colleagues. Mathews has received awards from the NAS, NAE, ASA, AES, and IEEE, as well as the Republique Francaise, and has been a professor of musical research at Stanford University since 1987.

2:25

4pMU3. Unit-generator architectures in computer music. Julius O. Smith III (CCRMA, Dept. of Music, Stanford Univ., Stanford, CA 94305, jos@ccrma.stanford.edu)

Max Mathews is well known as the "Father of Computer Music," having written the first program for generating digital sound samples from high-level "note" and "instrument" specifications. Instruments were specified as interconnections of elemental building blocks called *unit generators*. This presentation reviews some of the history and impact of the unit-generator concept in computer music over the past half-century. Widespread dissemination of the unit-generator architecture for sound synthesis began with the Music V Program (the first Fortran version), written by Max Mathews in 1968. Derivatives soon appeared at research universities, such as csound (Vercoe, MIT), Mus10 (Chowning's group, Stanford), and cmusic (Moore, UCSD). Hardware acceleration for unit generators appeared in the 1980s. Later, Music-V descendants included Lisp-based music compilers such as Common Lisp Music (CLM), the Synthesis Tool Kit in C++ (Cook *et al.*, Princeton), and graphical programming languages for real-time music processing/synthesis, such as Max/MSP and Pure Data (PD) started by Miller Puckette. More recent descendants include SuperCollider (McCartney, UCSB) and ChucK (Wang *et al.*, Princeton). Thus, the unit-generator architecture, introduced by Max Mathews half a century ago, has firmly taken root as a preferred modular approach to the problem of constructing virtual musical instruments and digital audio effects.

2:50

4pMU4. The Mathews legacy on the $100 Laptop. Barry Vercoe (Media Lab., M.I.T., 20 Ames St., Cambridge, MA 02139, bv@media.mit.edu)

The Mathews Music 4 program was the inspiration for a series of audio-processing languages by this author: Music 360 (1968), Music 11 (1973), Csound (1985), XTCsound (1997), and CsoundXO (2007). Instruments modeled in any one of these systems will run almost unmodified in any other, a testament to the long reach of Max's original concepts. But Max's longest reach is yet to come, for CsoundXO has been developed for the $100 Laptop—a machine designed by One Laptop Per Child (OLPC) which aims to put a laptop in the hands of every child on the planet (www.laptop.org). These machines are mesh-networked so that a child in one village can connect to a child in a neighboring village, and so on, until an international access point is reached. But CsoundXO on each machine is also designed for collaborative performance, so that a child in one country can "share" his/her original composition with a child in a similar or different culture. And if the work is interactive, each child can perform collaboratively with others who may be near or far. This paper will describe how all this works, along with a live demonstration.

3:15

4pMU5. Tapping into the internet as an acoustical/musical medium. Chris Chafe (CCRMA/Music, Stanford Univ., Stanford, CA 94305, cc@ccrma.stanford.edu)

Recent work in network audio transport transforms advanced networks into a new kind of acoustical medium in which sound waves propagate, as if traveling through air, water, or solids. Waves sent through the medium are reflected or altered as they bounce between hosts. Propagation delays are used to create echo chambers and build the resonances for "distributed musical instruments." As a side-effect, tones created by network resonances can be used to monitor the quality of the underlying network. The presentation presents three areas of research: (1) Auditory methods for monitoring QoS, especially for networks supporting real-time, interactive, bidirectional flows; (2) remote musical collaboration using professional-quality, low-latency audio; (3) empirical study of human factors affected by some unique acoustical properties of the medium network latency; jitter and delay asymmetry affect the speed of sound and are never uniform. By creating distributed virtual sound objects like instruments and rooms, and by studying distributed ensembles, we can begin to understand this new sound world. Some effects have been measured empirically and the results contain some surprises.

3:40

4pMU6. Max Mathews in real-time. David Wessel (Ctr. for New Music and Audio Technologies (CNMAT), Dept. of Music, Univ. of California Berkeley, Berkeley, CA 94720, wessel@cnmat.berkeley.edu)

Some of the most important contributions of Max Mathews stem from his emphasis on live-performance computer music. His performer-oriented perspective has won over musicians ranging from Pierre Boulez to Stevie Wonder. The most prominent real-time computer language in use today, Max/MSP, bears his name. In the 1960s, after making an overwhelmingly convincing argument for

the digital computer as a musical instrument, he and F. R. Moore at Bell Labs built a hybrid system, GROOVE, consisting of analog synthesis modules under real-time digital control. Though not a fully digital system—a practical move—it anticipated much of what we in the field are currently doing with laptops and gestural controllers in performance. Indeed his GROOVE conducting program impressed both Pierre Boulez and Stevie Wonder when it was demonstrated to them in the mid-1970s. I will strive to honor Max's contributions to musical performance with examples from his work and those influenced by him. Along the way there will be a few stories.

4:05

4pMU7. Max Mathews' magical music machines. John Chowning and Maureen Chowning (CCRMA, Dept. of Music, Stanford Univ., Stanford, CA 94305)

For nearly 40 years, Max Mathews has worked on real-time control of computers that he first programmed to make music in 1957. Beginning with a simple telegraph key to control tempo, followed by more sophisticated controllers that he developed while at Bell Telephone Laboratories and at Stanford's Center for Computer Research for Music and Acoustics (CCRMA), his latest and most sophisticated controller is his radio baton. We will demonstrate the extraordinary impact of Mathews' work on the world of music performance as the newest of musical instruments—computer and radio baton—joins with the oldest of musical instruments—the singing voice—in the performance of compositions where the soprano controls her computer generated accompaniment and computer processes. The presentation will include *Sea Songs* by Dexter Morrill, and Richard Boulanger's *Solemn Songs for Evening*, which uses the Bohlen-Pierce scale.

4:30–4:35 Break

4:35–5:00
Mini-Concert of Electronic and Computer Music

FRIDAY AFTERNOON, 30 NOVEMBER 2007 NAPOLEON A1/A2, 1:00 TO 5:15 P.M.

Session 4pNS

Noise and Animal Bioacoustics: Advances in Measurement of Noise and Noise Effects on Animals and Humans in the Environment II

Brigitte Schulte-Fortkamp, Cochair
Technical Univ. Berlin Inst. of Fluid Mechanics and Engineering, Secr TA 7, 10587 Berlin, Germany

Ann. E. Bowles, Cochair
Hubbs Sea World Research Inst., 2595 Ingraham St., San Diego, CA 92109

Chair's Introduction—1:00

Invited Papers

1:05

4pNS1. Quantifying lost opportunities to hear natural sounds. Kurt Fristrup (Natural Sounds Program, Natl. Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525)

Hearing provides an omnidirectional alerting sense for wildlife that seems to be universal: No deaf vertebrate species are known and invertebrates display a remarkable diversity of hearing mechanisms. Anthropogenic noise elevates ambient sound levels, which masks natural sounds that would otherwise be heard. The costs of this masking can be assessed by calculating the loss of listening area or alerting distance that results. Listening area metrics are appropriate when a search function might be affected (e.g., foraging), while alerting distance metrics are appropriate when the distance to the sound source mediates the function (e.g., avoiding predation). Analytical approaches for calculating loss of listening area and alerting distance should incorporate available hearing data to account for the effects of hearing thresholds and critical bandwidths. A range of models for masking can be used. Very simple models may sacrifice accuracy to suggest metrics that are readily calculated using existing noise models. More complex models can capture the idiosyncrasies of each species hearing capabilities to render more detailed results. Examples of applying these metrics to National Park Service contexts are discussed, to illustrate the use of these concepts to render environmental acoustic data for resource managers and NPS leadership.

1:30

4pNS2. Noise impacts on birds: Assessing take of endangered species. Edward W. West (Jones & Stokes, Inc., 2600 V St., Sacramento, CA 95818), Robert J. Dooling, Arthur N. Popper (Univ. of Maryland, College Park, MD 20742), and David M. Buehler (Jones & Stokes, Inc., Sacramento, CA 95818)

Section 9 of the Endangered Species Act makes it illegal to take (harass, harm, pursue, hunt, shoot, wound, kill, trap, capture, or collect) an endangered species. Harass includes any action that would significantly disrupt normal behavioral patterns including breeding, feeding, or sheltering. "Harm" includes any action that actually kills or injures a listed species. To better understand how and to what extent anthropogenic noise can harm or harass birds, we examined the potential for noise effects at three levels: Physiological stress, hearing impairment, and interference with communication through signal masking. Pile driving and blasting ($\approx$100 dBA at 15 m) can potentially cause temporary or permanent hearing impairment in birds. Chronic intense noise (e.g., oil field compressor station: 92 dBA at 20 m) may induce physiological stress in some species if they cannot avoid exposure. Finally, highway noise as low as 45 dBA can potentially mask acoustic communication and modify breeding and other behaviors in many species. Thresholds of effect and effect distances are largely species-specific, reflecting differences in tolerance capacity, spacing requirements, and bioacoustic profiles. Indices of take can be physiological, behavioral, or ecological, but must be verifiably correlated with significant changes in species viability.

1:55

4pNS3. Environmental noise: Is there any significant influence on animals? Klaus Genuit and André Fiebig (HEAD Acoust. GmbH, Ebertstrasse 30a, 52134 Herzogenrath, Germany, klaus.genuit@head-acoustics.de)

It is stated that there is no convincing evidence that animals are deterred by intense sounds (Heffner 2007). Nevertheless, it must be thoroughly examined to what extent animals are impaired by specific environmental noise with regard to their orientation, communication, behavior, etc. Such interferences would significantly influence the animals' life. Similar to the auditory sensitivity of animals, the human hearing system shows a high performance in amplitude, frequency, and time resolution. It pays attention to more factors and noise features than only to absolute SPLs. Psychoacoustic parameters as well as further hearing-related parameters can capture such specific factors and features of noise, which allow us to draw conclusions concerning noise annoyance or, respectively, the pleasantness of noise. The adequate measurement and use of such parameters resulting in an advanced analysis and description of environmental noise—beyond simple sound pressure level considerations—provide valuable information with respect to noise annoyance, stress-related reactions, and interference of daily life activities of humans. The presentation will highlight the relevance, the determination, and interpretation of psychoacoustic and other hearing-related parameters in the context of environmental noise, with respect to the hearing sensation of humans.

2:20

4pNS4. Role in science in assessing noise impacts on wildlife under National Environmental Policy Act (NEPA). Sheyna Wisdom (URS Corp., 2700 Gambell St., Ste. 200, Anchorage, AK 99502)

Data uncertainty is an unfortunate reality of wildlife management. Data gaps can lead to decision uncertainty when assessing noise impacts on terrestrial and marine wildlife through the National Environmental Policy Act (NEPA) process and other regulations. How can managers take a precautionary approach to effects of noise on wildlife in light of imperfect information without imposing unacceptable restrictions on activities? This talk will describe the existing policies and/or guidelines used to assess noise impacts to marine and terrestrial wildlife under NEPA. It will provide specific examples of how noise impacts on wildlife have been assessed using GIS-based technology and industry-accepted acoustic propagation models, such as the Federal Highway Administrations Traffic Noise Model. There will also be a discussion of important gaps in our understanding of wildlife susceptibility to noise. Using principles of adaptive management, wildlife managers can implement policies that address data gaps, protect wildlife, and yet allow human activity.

2:45

4pNS5. Monitoring, prediction, and management of sonic booms in a valued ecosystem. Kenneth J. Plotkin (Wyle Labs., 241 18th St. S., Ste. 701, Arlington, VA 22202, kenneth.plotkin@wylelabs.com), Louis LaPierre (Inst. for Environ. Monitoring and Res., Moncton, NB E1C 9X5, Canada), and J. Wayne Boulton (RWDI Air Inc., Guelph, ON N1K 1B8, Canada)

Goose Bay, Labrador, is a sensitive ecosystem under airspace that has been host to military flying operations since World War II. Since 1995, the Institute for Environmental Monitoring and Research has documented and helped mitigate the effects of low altitude flight operations, serving to protect the welfare of aboriginal people and the survival of wildlife species in the area. There are current plans to conduct supersonic operations in part of the airspace. Based on experience with this type of operation in other places, there is an expectation that resultant sonic booms can be safely accommodated, but it is necessary to monitor effects. The Institute has sponsored the development of a sonic boom forecast model that combines real-time three-dimensional weather forecasts with sonic boom ray trace modeling. A set of new digital noise monitors has been developed to record noise and sonic booms in the airspace. Field biologists will observe behavioral response of key species. These elements will be key components of an adaptive management system that will ensure preservation of this highly valued ecosystem.

3:10

4pNS6. Hierarchical method for single-observer, continuous sound source observer logging. Richard Horonjeff (Acoust. and Noise Control, 81 Liberty Square Rd. #20-B, Boxborough, MA 01719) and Grant Anderson (Acoust. and Statistics, Concord, MA 01742)

Sound environments, whether urban or remote, often contain a variety of audible sound sources, each with different onset and offset times. Obtaining a continuous log of these sources can be a daunting task, especially for a single observer. In situations where it is impractical to deploy several observers to a single location, a hierarchical, forced-choice method has been developed. The method stratifies the sources into layers, each layer containing sources with common attributes, such as human-related or natural-indigenous. The method collects sufficiently detailed information for study objectives, without taxing the observers' cognitive and recording abilities, and thereby introducing large observation or recording errors. So long as the onsets and offsets of audible sources in one layer are independent of those in another, statistical descriptors, such as the percent of time audible, can be determined for each layer. The highest layer in the hierarchy becomes an exhaustive sample, whereas lower layers become statistical samples with random blocks of time removed. The method has been applied in a number of national parks over the past 15 years, by several organizations and sponsors. Advantages and disadvantages of the method, along with inter-observer logging consistency results, are presented.

3:35–3:50 Break

Contributed Papers

3:50

4pNS7. Environmental noise studies in Puget Sound. David Dall'Osto and Peter H. Dahl (Appl. Phys. Lab. and Dept. of Mech. Eng., Univ. of Washington, Seattle, WA 98110)

The ambient noise environment at a site in North Puget Sound, Washington, depth ~100 m, located within the nearby Smith Island marine sanctuary, is studied. The measurement system consists of a buoy for which the surface expression houses a microphone at nominal height 2 m above the sea surface, with multiple underwater acoustic sensors suspended in the water column. The recording bandwidth for the air system corresponds to audio band, whereas the underwater system records ambient noise at frequencies up to 50 kHz. The two systems are recorded coherently. One goal of this pilot study is to examine different components of the noise budget, including injection of noise from airplane flyovers, and correlation between pressures above and below water. Another goal relates to properties of the noise field that could possibly impact echolocation by southern resident killer whales; these properties likely being restricted to noise levels at frequencies above 25 kHz. Besides representing an important marine mammal habitat, a key advantage of this site is the availability of meteorological and sea surface wave data, obtained from Smith Island and a nearby NOAA buoy, and vessel and air traffic data. Results from the summer 2007 field work will be presented.

4:05

4pNS8. A procedure for the assessment of residential low-frequency noise complaints. David C. Waddington, Andrew T. Moorhouse, and Mags D. Adams (Acoust. Res. Ctr., School of Computing, Sci. and Eng., Univ. of Salford, Salford M5 4WT, UK)

Although the number of people that complain about low-frequency noise (LFN) is comparatively small, those that do tend to suffer severe distress. LFN is a recognized problem in many developed countries, and this paper describes a procedure for the assessment of LFN complaints that has recently been developed in the UK. Human reaction to sound is dependent not just on the sound itself, but on a complex array of other factors including personal associations with the sound. Consequently, the procedure does not provide a prescriptive indicator of nuisance, but rather gives guidance notes and a pro-forma report with step-by-step instructions to help environmental health practitioners to form their own opinion. In particular, an interview-based questionnaire is used to complement physical recordings of low-frequency sounds in the complainant's home, together with an event log completed by the sufferer. Examples of field measurements and application of the procedure are presented. [Work funded by the Department for Environment, Food and Rural Affairs (Defra) UK.]

4:20

4pNS9. Speech intelligibility in a passenger train compartment. Tze Leung Yip, William W. S. Fung, Chun Wah Leung (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong), Glenn H. Frommer (MTR Corp., Kowloon Bay, Hong Kong), and Kai Ming Li (Purdue Univ., West Lafayette, IN 47907-2031, mmkmli@purdue.edu)

An effective delivery of an informative announcement through a public address system in a mass transit train is very important from the viewpoints of safety and passengers' comfort. A high quality of speech intelligibility is needed in the compartment to facilitate the effective communication between train operators with passengers. The reverberation time is often used as one of the key elements for assessing and determining the quality and perception of the speech sound in an enclosure. Not only does it indicate the level of absorption and reflection in the train compartment, it also affects the overall noise levels and the clarity of speech. In the current study, a simple ray model is presented to estimate the sound field in a train compartment. It is then used to predict the decay curve of noise levels in which the reverberation time is calculated. Full-scale field measurements were conducted in a train compartment operating in a tunnel, for validation of the numerical model. This study aims to understand the speech intelligibility of a train compartment in a set of realistic operating conditions for setting achievable contractual specifications for the design of underground trains operating in tunnels.

4:35–5:15
Panel Discussion

Session 4pPA

Physical Acoustics: Atmospheric Acoustics

D. Keith Wilson, Chair
U.S. Army Cold Regions Research Lab., Engineering Research and Development Center, 72 Lyme Rd., Hanover, NH 03755-1290

Contributed Papers

1:30

4pPA1. Experimentally measured diffraction of sonic boom waves around a house. Victor W. Sparrow (Grad. Prog. in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu), Jacob Klos, and Ralph D. Buehrle (NASA Langley Res. Ctr., Hampton, VA 23681)

During June 2006, a series of sonic boom flight tests were conducted as a part of the NASA low-boom/no boom program. One objective was to evaluate structural acoustic building response on a house. This presentation gives some results of that test regarding the outdoor pressure response on and near the house. The passage of the sonic booms over a nearby wall and the house itself have been examined to determine the maximum pressure loading. Both conventional N-waves and low-amplitude N-waves have been considered. The results indicate that the walls of the house facing the direction of the incoming sonic boom receive the maximum loading in agreement with image theory. The findings also confirm the temporal and spatial dependence of the structural loading. [The first author is supported by NASA and the FAA/NASA/Transport-Canada PARTNER Center of Excellence.]

1:45

4pPA2. Analysis of reverberation characteristics for sniper localization in an urban environment. Brad Libbey and James Perea (Army RDECOM CERDEC NVESD, 10221 Burbeck Rd., Fort Belvoir, VA 22060)

Acoustic sniper localization algorithms have proven useful to soldiers, but are difficult to implement in urban environments, due to reverberation and diffraction. Research is being performed to assist traditional localization methods by interpreting reverberant responses present in urban environments. Measured sniper responses are processed to establish if the first acoustic path is direct or indirect, based on characteristics of the received signal. If the first arrivals on several receivers are direct, an accurate relative time delay can be determined for each, which leads to accurate source localization. Indirect microphone arrivals can adversely affect the localization algorithm. Various metrics were used to characterize received signals: Arrival energy, reverberation decay time, initial arrival crest factor, and relationships between first and subsequent arrivals. Combining these factors aids in the identification of obstructed receivers, providing information about the effectiveness of a channel in a localization algorithm.

2:00

4pPA3. Acoustic surface waves and long-range detection of mortars and artillery. Xiao Di, Carrick L. Talmadge, Roger Waxler, and Kenneth E. Gilbert (Univ. of Mississippi, Natl. Ctr. for Physical Acoust., Coliseum Dr., University, MS 38677)

A natural terrain surface, because of its porosity, can support an acoustic surface wave that is a mechanical analog of the familiar vertically polarized surface wave in AM radio transmission. At frequencies of several hundred Hertz, the acoustic surface wave is attenuated over distances of less than about 100 m. At lower frequencies (e.g., below 100 Hz) the attenuation is much less, allowing surface waves to propagate thousands of meters. Thus, understanding the low-frequency surface wave is important for detecting and locating mortars and artillery. Data from recent field experiments will be shown demonstrating the persistence of the surface wave under different meteorological conditions. Propagation calculations will be presented using typical source signatures for mortars and artillery, and the implications for long-range detection will be discussed. [Research supported by the U.S. Army TACOM-ARDEC at Picatinny Arsenal, New Jersey.]

2:15

4pPA4. Refraction corrections to source localization using an acoustic array suspended below an aerostat. Vladimir E. Ostashev (NOAA/ Earth System Res. Lab., 325 Broadway, Boulder, CO 80303 and Dept. of Phys., New Mexico State Univ., Las Cruces, NM 88003), Michael V. Scanlon (U.S. Army Res. Lab., Adelphi, MD 20783), and D. Keith Wilson (U.S. Army Engineer Res. and Dev. Ctr., Hanover, NH 03755)

Acoustic sensor arrays suspended below tethered aerostats have several advantages in detection and localization of sources on the ground in comparison with ground-based sensors. Aerostats can be elevated up to 1-2 Km above the ground where the temperature and wind velocity can significantly differ from their values close to the ground. Therefore, due to refraction, a ray trajectory that connects a source on the ground and an aerostat can bend significantly resulting in different directions of sound propagation along this trajectory. However, so far, refraction has been ignored in source localization using aerostats. In this paper, a theory is developed that allows one to find the source coordinates more accurately by taking into account refraction of sound signals emitted by the source. The source coordinates are expressed in terms of direction of sound signal propagation near the aerostat, the coordinates, and orientation of the aerostat, and the vertical profiles of temperature and wind velocity. Other meteorological parameters may also be incorporated, if available, such as time-of-day, geographical features, and historical weather trends.

2:30

4pPA5. Acoustic tomographic study of the mesoscale coherent structures in the lower atmosphere. Igor Chunchuzov Petrovich, Sergey Kulichkov, Vitali Perepelkin (Obukhov Inst. of Atmospheric Phys., 119017, 3 Pyzhevskii Per., Moscow, Russia), Astrid Ziemann, Klaus Arnold, and Anke Kniffka (Univ. of Leipzig, Germany)

The results of acoustic tomographic monitoring of the coherent structures in the lower atmosphere and the effects of these structures on the acoustic signal parameters (travel time, duration and direction of propagation) are studied. From the measurements of acoustic travel time fluctuations (periods 1 min-1h) by a net of distanced receivers the temporal fluctuations of the effective sound speed are retrieved along different ray paths connecting a pulse source and the receivers. By using a coherence analysis of the fluctuations near spatially distanced ray turning points, the internal wave-associated fluctuations are filtered and their spatial characteristics (coherences, horizontal translation velocities, and spatial scales) are obtained. A possible explanation of the presence of the dominant periods in the observed mesoscale variations of the acoustic parameters, wind shears and turbulent fluxes of momentum near ground is done.

2:45

4pPA6. Effects of internal gravity waves on acoustic pulse propagation in the atmospheric wave guides. Igor Chunchuzov, Sergey Kulichkov, and Vitali Perepelkin (Obukhov Inst. of Atmospheric Phys., 119017 Moscow, 3 Pyzhevskii Per., Russia)

Recently developed theoretical model of a random internal wave field in the atmosphere is used for calculating the statistical characteristics of the parameters of acoustic pulses propagating in the atmospheric wave guides. The variances, frequency spectra and spatial structure functions of the internal wave-associated fluctuations of the acoustic pulse travel times and grazing angles are calculated and compared to those obtained in the experiments on pulse sounding of the atmosphere. The effects analyzed here include: a scattering of acoustic waves from highly anisotropic wind speed and temperature inhomogeneities into the so-called acoustic shadow zones; a discrete character of the observed frequency spectra of the fluctuations of acoustic pulse parameters in the range 1min-1h; an error caused by internal waves in determining of the angle of arrival of the acoustic signal in the waveguide.

FRIDAY AFTERNOON, 30 NOVEMBER 2007 GRAND BALLROOM E, 3:00 TO 4:35 P.M.

Session 4pSA

Structural Acoustics and Vibration and Noise: Ground Vibration Impact on Buildings and General Structural Acoustics

James E. Phillips, Chair
Wilson Ihrig and Associates, Inc., 5776 Broadway, Oakland, CA 94618-1531

Invited Papers

3:00

4pSA1. Assessment of ground vibration impact from automotive and transit sources on future biotechnology research center using finite element analysis (FEA). James E. Phillips (Wilson, Ihrig & Assoc, Inc., 5776 Broadway, Oakland, CA 94618)

A new science and biotechnology research center was to be built in a metropolitan industrial area. There was concern that ground vibration from a nearby freeway, street traffic, and trains on an adjacent railroad would impact vibration sensitive research equipment inside the building. Ground vibration was measured at the project site prior to construction. Finite Element Analysis (FEA) was used to develop a computer simulation of the building structure using the measured ground vibration as input to the FEA model. The study determined the building floor vibration due to exterior sources would achieve the project's criteria for the building structure as designed.

3:25

4pSA2. Groundborne noise and vibration control at the Toronto Four Seasons Centre for the Performing Arts. Steven L. Wolfe (Wilson, Ihrig & Assoc, Inc., 5776 Broadway, Oakland, CA 94618)

The Four Seasons Centre for the Performing Arts, Canada's first purpose-built opera house had its grand opening on June 14, 2006. Occupying an entire block of the financial and theatre district in downtown Toronto (across from the Sheraton Hotel), the facility with its 2,000 seat auditorium, is the new home of the Canadian Opera Company and the performance venue for the National Ballet of Canada. A major challenge for the project's downtown location was isolating the auditorium from the vibration and noise from the adjacent subway system on University Avenue and surface streetcars on Queen Street. A "building within a building" vibration isolation design was developed that set the entire auditorium, stage, and rehearsal hall on rubber isolation bearings. This paper provides design details and the procedures used to ensure that the groundborne noise from outside activities would be inaudible. Results of follow-up measurements are presented which confirmed that the vibration design effectively mitigated the outside vibration and noise intrusion inside the auditorium to achieve background noise levels from exterior sources equivalent to the threshold of human hearing.

Contributed Papers

3:50

4pSA3. An adaptive tunable dynamic vibration absorber. Barbara Tiseo, Antonio Concilio, Antonio Gianvito (CIRA, The Italian Aerosp. Res. Ctr., Via Maiorise, Capua (CE), Italy), and Gary Koopmann (The Pennsylavania State Univ., State College, PA 16801)

This paper examines a new concept of adaptive tuned dynamic vibration absorbers (ATDVAs) using shape memory alloy (SMA) elements instead of spring elements. Shape memory is a class of alloys that shows a reversible change in crystalline structure. In the martensite phase, the material exhibits a low elastic modulus and yield strength. Subsequent heating of the material induces the change to austenite, with a corresponding higher elastic modulus and yield strength. The result of the phase transformation is a sizable change in the geometry, in the internal tension, a considerable deformation, and a concomitant frequency shift, as in the case where the SMA is used as the stiffness component of a tuned dynamic absorber. In this research, experimental investigations have been focused to verify the capability of an ATDVA to control vibration on a free-free aluminum panel.

4:05

4pSA4. A simplified method for approximating the natural frequencies of acoustic-structure systems. R. Benjamin Davis (Pratt School of Eng., Duke Univ., Durham, NC 27708)

It is well known that the natural frequency of a flexible structure in contact with a fluid-filled cavity can be approximated with a closed-form expression that considers the coupling of only two component modes (i.e., a rigid-wall acoustic mode and an in vacuo structural mode) [F. Fahy, *Sound and Structural Vibration* (Academic, London, 1985), Chap. 6, pp. 249–256]. However, this expression requires volume and area integrations over the component mode shapes. For many practical configurations, the effort to determine the component mode shapes and compute the necessary integrals counteracts much of the time and computational savings afforded by the closed-form expression. Here, the closed-form expression is recast as a function of the nondimensional frequency separation between the component modes and a new nondimensional coupling parameter, Ψ. Design curves representing the value of Ψ for common geometries and boundary conditions are presented. With the use of these design curves and a knowledge of the component modes frequencies, one can quickly compute reasonable approximations of the coupled natural frequencies by hand.

4:20

4pSA5. Vibration of circular plate and its frequency parameters. Li-Feng GE (School of Electron. Sci. and Technol., Anhui Univ., Hefei 230039, China)

The vibration of circular plates with inplane forces has been classically discussed [T. Wah, J. Acoust. Soc. Am. 34(3), 275–281, 1962, and A. Leissa, Vibration of Plates, ASA, 1993, originally, NASA, 1969]. In these literatures each of two characteristic values is expressed as a function of two stiffness factors, i.e., the inplane tension and the bending stiffness of the plate. The author has extended it to a general case in which the stretched plate is also supported by a massless elastic medium (or foundation). The corresponding two characteristic values are expressed as a function of three stiffness factors, which includes an additional stiffness of the foundation [L.-F. Ge, 16th ICA/135th ASA Meeting, Proc. 1081-1082, 1998, and J. Acoust. Soc. Am. 103, 2870, 1998]. In this paper the two characterized values and the frequency parameter have been calculated and checked further. It is found that some values given in the previous literatures have some errors, the biggest one of which is 4.0%. Furthermore, this paper developed a 3-D plot of the characterized values, and gave the corrected tables of the characterized values and frequency parameter for both simply supported and clamped plates. [Work supported by NSFC (60374044).]

FRIDAY AFTERNOON, 30 NOVEMBER 2007 NAPOLEON B3, 1:30 TO 3:45 P.M.

Session 4pSC

Speech Communication: Auditory and Somatosensory Feedback in Speech Production II

Anders Lofqvist, Chair
Haskins Laboratories, 300 George St., Ste. 900, New Haven, CT 06511

Chair's Introduction—1:30

Invited Papers

1:35

4pSC1. Compensatory responses to unexpected jaw loading during speech observed with magnetometry. Mark Tiede (Haskins Labs & MIT R.L.E., 300 George St., New Haven, CT 06511), Takayuki Ito (Haskins Labs, New Haven, CT 06511), and David Ostry (McGill U.)

Observations were made of articulator and formant trajectories during speech perturbed by unexpected mechanical loads applied to the jaw. Subjects produced multiple repetitions of the phrase "see X avis," where X was one of "raw," "rob" or "rod." Perturbation forces were applied using a jaw-coupled robot, and triggered using a thresholding criterion applied to real-time tracking of the initial low to high transition in the first formant. Perturbations were delivered one out of every five repetitions, selected at random, with half applied upwards and half downwards, and forces sustained throughout the remainder of the utterance. Three EMMA sensors located on the anterior midsagittal surface of the tongue in addition to lower mandibular and upper lip sensors were used to track articulator positions. Audio and surface EMG (masseter, ABD, and OOS) were collected concurrently. Formants show initial deviation from control trajectories and then recovery that begins approximately 75 ms after the onset of perturbation. Compensation in most instances is nearly complete, even though jaw position does not recover its unperturbed trajectory. Perturbed tongue sensor trajectories are strongly distinguished by coda type, but show a pattern converging on the unperturbed tongue shape consistent with the tongue's presumed role in effecting formant recovery.

2:00

4pSC2. Sensory movement goals and control mechanisms in speech production. Joseph Perkell (Res. Lab. of Electrons, Massachusetts Inst. of Technol., Rm. 36-591, 50 Vassar St., Cambridge, MA 02139-4307)

An overview of speech production is described in which the goals of phonemic speech movements are implemented in auditory and somatosensory domains and the movements are controlled by a combination of feedback and feedforward mechanisms. Findings of motor-equivalent trading relations in producing /u/ and /r/, cross-speaker relations between vowel and consonant production and perception, and speakers use of a saturation effect in producing /s/ support the idea that the goals are in sensory domains. Results of production experiments in which auditory feedback was modified and interrupted, provide insight into the nature of feedback and feedforward control mechanisms. The findings are all compatible with the DIVA model of speech motor planning [Guenther et al., Brain & Language **96**, 280–301(2006)], which makes it possible to quantify relations among phonemic specifications of utterances, brain activity, articulatory movements, and the speech sound output. [Research supported by NIDCD/NIH.]

2:25

4pSC3. Somatosensory influence on the perception of speech sounds. Takayuki Ito, Mark Tiede (Haskins Labs., 300 George St., New Haven, CT 06511, taka@haskins.yale.edu), and David J. Ostry (McGill Univ., Montreal, QC, Canada)

Along with auditory information, sensory information associated with articulatory movements may also be involved in the neural processing that subserves speech perception. Although somatosensory organs in the orofacial system provide this kind of input, there have been few studies assessing the influence of somatosensory afferents on speech perception. To explore this idea, we examined the possibility that by altering somatosensory information, we might modify the perception of speech sounds. We tested native speakers of American English in a speech identification task using a 10-step synthesized continuum between gheadh and ghad.h Subjects listened to the stimuli one at a time through headphones and classified them as gheadh or ghad.h For half of the trials, during the presentation of the auditory stimulus, we used a robotic device to stretch the facial skin stretch lateral to the oral angle in order to modulate somatosensory information. The mechanical perturbation was timed so that its start and end coincided with that of the auditory stimulus. The results show a systematic modification of speech perception due to altered somatosensory input, which varied systematically according to the direction of the skin stretch perturbation. This suggests that somatosensory information influences the neural processing of speech signals.

2:50

4pSC4. Neural mechanisms underlying sensory feedback control of speech. Jason Tourville and Frank Guenther (Dept. of Cognit. and Neural Systems, Boston Univ., 677 Beacon St., Boston, MA 02215, jtour@cns.bu.edu)

The DIVA model predicts that perturbation of speech will result in activation of sensory error cells that encode the mismatch between the expected and realized sensory consequences of speech production. Cells coding auditory and somatosensory errors are hypothesized to lie in posterior superior temporal gyrus and supramarginal gyrus, respectively. Activation of these error cells drives compensatory articulator movements, marked by increased activation of ventral motor cortex. To test these hypotheses, blood oxygen level dependent responses during normal and perturbed feedback conditions were assessed in two fMRI experiments. In one experiment, transient, unexpected auditory perturbations were induced by shifting F1 upward or downward; in another, a small balloon was inflated during speech production to induce a somatosensory perturbation. Increased bilateral activation in higher-order sensory cortices in response to the feedback perturbation was consistent with model predictions. Increased activation of frontal regions was found primarily in the right hemisphere. The results suggest that detection of feedback error by bilateral sensory error cells results in a shift from left-lateralized feedforward control of speech to feedback-based control mechanisms that rely more heavily on right hemisphere lateral frontal regions. [This work was supported by NIDCD Grant No. R01 DC02852 (F. Guenther, PI).]

3:15

4pSC5. How is auditory feedback processed during speaking? John F. Houde (Dept. of Otolaryngol.—Head and Neck Surgery, Univ. of California San Francisco, San Francisco, CA 94143) and Srikantan S. Nagarajan (Univ. of California San Francisco, San Francisco, CA 94143)

Several studies have shown that speaking suppresses the normal response to speech sounds in auditory cortex and associated regions. Our own studies using magnetoencephalography (MEG) suggest a model of how auditory feedback is processed during speaking where speaking-induced suppression (SIS) arises from a comparison between actual auditory input and a prediction of that auditory input. Over the past several years, we have been testing this model in various experiments. We have studied the specificity of SIS by examining how SIS is modulated by different types of feedback alterations. We have also looked at how choice of speech target produced modulates SIS. Our model also makes predictions about how auditory cortex would respond to transient perturbations of speech feedback, which we have tested in experiments examining the auditory response to pitch feedback perturbations. Finally, we postulate that our model of speech feedback processing is really just a special case of how auditory feedback is processed in all motor tasks with associated auditory feedback. Our recent experiments looking at responses to auditory stimuli induced by a self-initiated motor acts are consistent with this more general interpretation of our model. [Work supported by NSF Grant No. BCS-0349582 and NIH/NIDCD Grant No. R01-DC006435.]

Contributed Paper

3:40

4pSC6. A neuroimaging investigation of auditory-motor learning. Kevin J. Reilly, Frank H. Guenther, Jason A. Tourville (Dept. of Cognit. & Neural Systems, Boston Univ., 677 Beacon St., Boston, MA 02215), and Jason W. Bohland (Cold Spring Harbor Lab., Cold Spring Harbor, NY 11724)

Brain activity was measured in 11 subjects while learning a novel speech-like auditory-motor task. Subjects learned to move a dot on a screen to one of three target locations by changing their formant frequencies during production of vowel-like sounds. The targets corresponded to regions of the formant space not associated with American English vowels. Functional images were acquired using a sparse sampling technique that allowed performance of the task in the absence of scanner noise and avoided artifactual signal changes resulting from articulator movements. The accuracy of subjects' production of the novel vowel sounds increased significantly over the course of the scanning session. Significant correlations between performance error and brain activity were observed in a number of areas, including superior temporal gyrus, inferior frontal gyrus, primary motor cortex, supplementary motor area, and medial and lateral cerebellum. Brain regions exhibiting a positive correlation with performance error were generally consistent with regions that have been associated with feedback error detection/correction for the production of learned speech sounds. Greater right hemisphere activation of frontal and temporal areas raises the possibility of hemispheric specialization for this aspect of speech motor control.

3:55–5:00
Panel Discussion

Session 4pSPa

Signal Processing in Acoustics, Acoustical Oceanography, and Underwater Acoustics: Session Honoring Leon Sibul II

R. Lee Culver, Chair
Pennsylvania State Univ., Applied Research Lab., P.O.Box 30, State College, PA 16804-0030

Contributed Papers

1:30

4pSPa1. Classification of impulsive-source active-sonar echoes based on a model of auditory perception. Jason E. Summers, Charles F. Gaumond, Derek Brock, and Ralph N. Baer (Naval Res. Lab., Washington, DC 20375-5350)

Models of auditory perception are investigated as the basis for a biomimetic classifier of impulsive-source active-sonar echoes. Multidimensional scaling estimates the perceptual space in which listeners perform classification [J. E. Summers et al., J. Acoust. Soc. Am. **120**, 3125 (A) (2006)]. In the resulting space of perceptual dimensions, stimuli form distinct clusters and target is discriminated from clutter along a single perceptual dimension. Dimensions in this space do not correspond to features having simple algorithmic representations. Consequently, conventional methods to develop a mapping from signal space to feature space fail. Instead, dimensions reflect untrained categorical perception manifested through the mixtures of top-down and bottom-up processes used by listeners: A high-level cognitive process for interclass dissimilarity ratings and a low-level signal-based process for intraclass comparisons. Behaving as expert systems, listeners rapidly assign stimuli to categories based on prior experiences, a process analogous to the statistical description of classes in the class-specific method. In contrast, within-class comparisons reflect signal-derived features found to be most efficacious for differentiating between the signals, a process similar to generation of features by singular-value decomposition. Implications of these findings for design of hybrid generative/discriminative human-mimetic classifier architectures are discussed. [Work supported by ONR.]

1:45

4pSPa2. Acoustic dopplergram. T.C. Yang (Naval Res. Lab., Washington, DC 20375)

Lofargram (or spectral gram) is widely used in sonar for detection of narrowband tonal signals and/or wideband transient signals. Such signals at a level comparable to the noise level are difficult to detect using snapshots of the signal spectrum, as they look like the noise. However, as noise is random in time while the signals are persistent over time, they can be detected by line-association (often referred to as eyeball integration) using the Lofargram. This concept for passive signal detection is carried forward to active signal detection using the Dopplergram. In an active system, the challenge is to detect a target echo from the noise and/or the reverberation signals. In a noise-limited situation, there is only so much power one can put out due to hardware and environmental (marine mammal) limitations. In a reverberation-limited environment, the problem is the high probability of false alarms due to reverberations. Active detections using a Dopplergram are rid of these problems in a natural way. [Work supported by ONR.]

2:00

4pSPa3. The Cramer-Rao lower bound (CRLB) on the bearing estimate for a moving array of vector sensors. Edmund J. Sullivan (EJS Consultants, 46 Lawton Brook Ln., Portsmouth, RI 02871)

The Cramer-Rao lower bound (CRLB) on the variance of the bearing estimate for a moving array of vector sensors is derived. It is shown that the motion itself provides a reduction in the variance of the estimate by utilizing the bearing information inherent in the Doppler. This is referred to as the passive synthetic aperture effect. In order to achieve this, it is necessary to jointly estimate the source frequency along with the bearing. Further, depending on the estimator used, it is necessary to provide a priori information on the source frequency. This can be done by using the observed frequency as a first estimate of the source frequency. It is shown that the CRLB decreases with the processing time and array speed, thereby significantly outperforming the conventional bearing estimator. It is then shown that by including vector sensors colocated with the pressure sensors in the bearing estimate, the CRLB decreases even further. This is due both to the increase in the effective number of sensors, and the inherent directivity provided by the vector sensor.

2:15

4pSPa4. Removal of non-Gaussian, broadband interferers using blind source separation and spatial processing. Elizabeth Hoppe and Michael Roan (131 Durham Hall, Virginia Tech., Blacksburg, VA 24061)

A method is introduced where blind source separation of acoustical sources is combined with spatial processing to remove non-Gaussian, broadband interferers from space-time displays such as bearing track recorder (BTR displays). This is quite different from the standard technique of placing nulls in the direction of interferers. The algorithm is implemented and tested against standard adaptive beamforming techniques, such as minimum variance distortionless response beamforming. In order to illustrate the utility of the approach, experiments were carried out in an anechoic chamber using two acoustic sources. The interferer source remained stationary at boresight, while the signal source moved, in steps of two deg, from negative to positive 20 deg. At each step, one second of data was recorded using a 64-element microphone array that had been calibrated at each subband frequency. All 64 channels of the array were sampled simultaneously at a rate of 22,050 Hz. Spatial processing is implemented using a broadband beamformer. The broadband beamforming algorithm first divides the signals into subbands, basebands, resamples, and finally low-passes the subbands, so that they can be processed using standard frequency domain narrowband beamformers.

2:30

4pSPa5. Performance analysis and modeling of underwater acoustic communications at low input signal-to-noise ratios using direct sequence spread spectrum. T.C. Yang and Wen-Bin Yang (Naval Res. Lab., Washington, DC 20375)

Direct sequence spread spectrum (DSSS) uses a code sequence to spread the symbols at the transmitter and a de-spreader at the receiver to recover the transmitted symbols. The de-spreader (a correlator or a

matched filter) provides a processing gain (matched filter gain), which enhances the symbol energy over noise, thus allowing communications at low input signal-to-noise ratio (SNR). For underwater communications, additional processing is needed to mitigate the effect of the rapid temporal fluctuation of the propagation channel. This paper analyzes DSSS data collected during the TREX04 experiment, which uses an m-sequence as the spreading code. More than 1,000 packets have been analyzed at input SNR, varying from -15 dB to $+23$ dB. Zero-bit errors were achieved for input SNR as low as -8 to -10 dB for two processors used. Performance loss due to inaccurate synchronization, inaccurate channel estimation, and signal fading are quantitatively modeled as a function of decreasing SNR. [Work supported by ONR.]

FRIDAY AFTERNOON, 30 NOVEMBER 2007 GRAND BALLROOM B, 3:00 TO 5:20 P.M.

Session 4pSPb

Signal Processing in Acoustics: Three-Dimensional Arrays, Machine Noise and Vibration

Natalia A. Sidorovskaia, Chair
Univ. of Louisiana at Lafayette, Dept. of Physics, P.O. Box 44210, Lafayette, LA 70504-4210

Chair's Introduction—3:00

Contributed Papers

3:05

4pSPb1. The analogy between acoustic holography and sound field reproduction. Ji-Ho Chang and Yang-Hann Kim (Ctr. for Noise and Vibration Control, Korea Adv. Inst. of Sci. and Technol., Sci. Town, Daejon 305-701, South Korea)

Acoustic holography is to predict sound field by sound pressure values on measurement plane while sound field reproduction is to generate sound field by loudspeakers. They have absolutely different objectives but handle sound field so that they have a common mathematical approach according to the shape of the sound field of interest. Hence several methods of sound field reproduction, WFS, ambisonics, BPC, mode matching method, etc., correspond with each category of acoustic holography: plane holography, cylindrical holography and spherical holography, etc. In this paper, mathematical and technical similarities are introduced and discussed. [This work was supported by the Korea Science and Engineering Foundation (KOSEF) through the National Research Lab. Program funded by the Ministry of Science and Technology (M10500000112-05J0000-11210).]

3:20

4pSPb2. Alternative closeness functions for eye microphone array. Hedayat Alghassi (Dept. of Elec. and Comput. Eng., Univ. of British Columbia, Vancouver, BC, V6T 1Z4, Canada), Shahram Tafazoli and Peter Lawrence (Univ. of British Columbia, Vancouver, BC, V6T 1Z4, Canada)

A new signal processing algorithm, accompanied by a novel hemispherical microphone array structure for sound source localization in three-dimensional spaces was presented [H. Alghassi et al., ''Acoustic source localization with eye array,'' JASA **120**(5) (2006)]. This localization methodology, which has some analogy to the eye in localization of light rays, uses concepts of two-microphone (pinhole) or three-microphone (lens) cell structures alongside a special closeness function (CF) to approximate the proximity of the sound source direction to each of the hemisphere microphone directions based on particular similarity measures among signals. The CF plays a major role in the accuracy of the final source direction estimation. The earlier multiplicative CF (MCF) operates based on vector multiplication of spatial derivative and time derivative of microphone signals. This work presents two additional categories of CFs and compares them with MCF. The difference CF (DCF) is based on subtraction of delayed reference signal and shell microphone signals, while the correlative CF (CCF) is based on multiplication of delayed reference signal and shell microphone signals. Similar to MCF, both DCF and CCF perform demonstrated linear output versus deviation angle. Although DCF and MCF have not shown improved experimental accuracy compared to MCF, they attained lower computational complexity.

3:35

4pSPb3. A space domain complex envelope. Choon-Su Park and Yang-Hann Kim (Novic Ctr., Dept. of ME, KAIST, 373-1 Guseong-dong, Yuseong-gu, Daejeon, Republic of Korea, yanghannkim@kaist.ac.kr)

Sound visualization tools, for example, beam forming and acoustic holography, exhibit the spatial look of sound in time or frequency domain. However, they normally require a significant amount of computation time to draw well the sound picture in space. The picture contains a great deal of information: Sound pressure distribution, intensity pattern, or energy information with respect to space. The information is often far more than what we need in practice. For example, when we want to know only the location of the sound sources and somewhat averaged sound pressure distribution, we need a means that can provide only what we need. The complex envelope in time domain can be a good starting idea to deal with the problems we have. A method to generate a spatial domain envelope has been theoretically developed. We found that the method not only has an advantage to show rather simple spatial sound distribution, but also reduce significantly the computation time. The latter makes it possible to see the sound picture faster than before: About ten times faster. This method has been applied to many practical examples: For example, sound from a musical instrument and sound from machinery.

3:50

4pSPb4. Acoustic source identification using a generalized regressive discrete Fourier series for tomographic reconstruction. Joris Vanherzeele, Roberto Longo, Steve Vanlanduit, and Patrick Guillaume (Dept. of Mech. Eng., Vrije Universiteit Brussel, Pleinlaan 2, 1050 Brussels, Belgium)

When measuring three-dimensional phenomena such as acoustic fields using an interferometric technique, one is prone to measure different angles of view to obtain a full three-dimensional representation of the phenomenon under investigation. This is due to the fact that an interferometric technique measures a line integral across the optical path. To obtain the full three-dimensional view, the different angles of view are passed through a tomographic algorithm. The most widely used tomographic method is filtered back projection. However, this process suffers from a series of drawbacks, the most important one being the fact that substantial truncation errors occur in the back projection step. In this article, a method is devised to eliminate these errors, based on a parametric frequency domain approach called generalized regressive discrete Fourier series (GRDFS). The method will be applied to laser doppler vibrometer measurements on a loudspeaker. This source will be rotated to obtain continuous angle views of the acoustic field, hence eliminating the tedious process of rotating the measurement set-up. By demodulation of a measured signal, it is possible to determine the position of the loudspeaker in space. The results obtained with the GRDFS will be compared to the classical filtered back projection method.

4:05

4pSPb5. Ultrasonic Doppler vibrometry using direct undersampling. Asif Mehmood, Paul M. Goggans (Dept. of Elec. Eng., Univ. of Mississippi, Anderson Hall Rm. 302B, University, MS 38677, amehmood@olemiss.edu), and James M. Sabatier (Natl. Ctr. for Physical Acoust., University, MS 38677)

In ultrasonic Doppler vibrometry (UDV) systems, sound returned from a moving point scatterer is frequency modulated by the component of the scatterer's velocity in the direction of the system's colocated ultrasonic transducers. Because of this, time-frequency analysis of the receiver transducer output can be used to reveal the velocities of multiple scatterers moving within the sensor's field of view. This principle can be used, for example, as in [Zhang et al., EL110 J. Acoust. Soc. Am. 121 (2007)] to study human gait. Because it is band-pass limited, the received signal can be expressed equivalently in terms of low-pass limited in-phase and quadrature (I&Q) components. To reduce the amount of data to be stored and analyzed, it is advantageous to sample the I&Q signals rather than the received signal. This paper reports an implementation of undersampling [J. L. Brown, Jr., IEEE Trans. Information Theory IT−26, 613−−615 (1980)] to capture samples of the I&Q using a data acquisition card without the use of external oscillators or mixers. Undersampling yields I&Q samples that are interlaced rather than simultaneous in time. Here, a spectrogram using interlaced I&Q samples is derived using Bayesian spectrum analysis.

4:20

4pSPb6. On-line failure detection of a vibrating structure: A model-based approach. B. L. Guidry, J. V. Candy, K. A. Fisher, D. H. Chambers, and S. K. Lehman (LLNL, P.O. Box 808, L-154, Livermore, CA 94551)

Model-based failure detection is based on the principle that the MBP for a normal or pristine structure is developed first and tuned during the calibration stage assuring a statistically validated processor. Once developed, the MBP is used as the integral part in a sequential detection scheme to decide whether or not the structure under investigation is operating normally. When an abnormality is detected, a failure alarm is activated and the type of failure is classified based on a library of potential failure modes. Here we use high-order parametric models to capture the essence of the structures over a limited frequency band known to be sensitive to structural changes. These estimated or identified models for normal operations are then used to develop the MBP which in this instance is a recursive Kalman filter. The filter is known to produce zero-mean/white residuals when optimally tuned to the data. Failure is declared when these properties are no longer valid. Once the detection is accomplished, the next step is to classify the type of failure mechanism and eventually the locations. Here we show results of the designs on both simulated and measured data.

4:35

4pSPb7. Fault localization of moving machinery in a noisy environment. Jong-Hoon Jeon, Choon-Su Park, and Yang-Hann Kim (Ctr. for NOVIC, Dept. of Mech. Eng., KAIST, 373-1 Guseong-dong, Yuseong-gu, Daejeon 305-701, Republic of Korea)

Faults of rotating parts of a machine normally generate unexpected frequency band or impulsive sound, which has a period when it moves with a constant speed. The former can be detected by the moving frame acoustic holography method [S.-H. Park and Y.-H. Kim, ''An improved moving frame acoustic holography for coherent bandlimited noise,'' J. Acoust. Soc. Am. **104**, 3179–3189 (1998)]. We have attempted to apply the method to the latter case in the test site. The keywords are, therefore, the periodic impulsive sound, which is obviously different from those which can be visualized by the method, and the signal-to-noise ratio, which determines the success of early-fault detection. This research shows how the problems related with these keywords can be resolved. The main idea is that periodic impulsive signal can be expressed by an infinite set of discrete pure tones. This enables us to obtain alot of holograms that visualize periodic impulsive sound field and noise; therefore, holograms can be averaged to improve the signal-to-noise ratio until having reliable information that exhibits where the impulsive sources are. Theory, experiment, and application results to the train on a test rig are described. [Work supported by BK21 and KRRI.]

4:50

4pSPb8. Design of multi-input interleaved multisine excitation signals for ultrasonic testing. Roberto Longo, Steve Vanlanduit, Joris Vanherzeele, and Patrick Guillaume (Dept. of Mech. Eng., Vrije Universiteit Brussel (VUB), Pleinlaan 2,1050 Bruxelles, Belgium)

Multisines are periodic signals consisting of harmonically-related sine waves. By optimizing the phases of the different sine waves, a periodic signal is obtained with a small crest factor resulting in high signal-to-noise ratios. One disadvantage of multisines when several ultrasonic actuators are used simultaneously is that, in general, it is very difficult to distinguish from a measured ultrasonic signal the contribution of the different actuators. This is easier when pulses are used and when they do not overlap. In this contribution, a new approach for periodic continuous wave signals will be presented to separate the contribution coming from the different actuators (senders) at every receiver. The proposed approach is based on the use of multi-input interleaved multisines. Interleaved multisines contain energy at different spectral lines allowing an easy separation by means of a discrete Fourier transform. The separation becomes more complex when non-linear effects are present, but even then it is possible to apply this approach by properly selecting the excitation lines. The use of multi-input interleaved multisine offers new applications for ultrasonic NDT (e.g., combined transmission and reflection measurements) as well as in the field of material identification and biomedical applications.

5:05

4pSPb9. Bayesian spectrum estimation of termite signals using laser Doppler vibrometry. Asif Mehmood (Dept. of Elec. Eng., The Univ. of Mississippi, P.O. Box 4206, University, MS 38677), Orwa M. Tahaineh, Tom Fink, Vijay. P. Ramalingam, John. M. Seiner (Natl. Ctr. for Physical Acoust., University, MS), and Alan R. Lax (USDA-ARS Southern Regional Res. Ctr., New Orleans, LA)

Laser vibrometry provides a sensitive non-invasive means of measuring substrate vibration. These measurements include landmine detection, automotive testing, production testing, aerospace, and structural testing. We employed laser vibrometry for termite detection. The vibratory signal generated by termite head banging is picked up by the laser Doppler vibrometer. In laser vibrometry, the surface motion is monitored by heterodyne laser Doppler vibrometry, and the received heterodyne signal is sampled to produce time-series data. The time-series data thus obtained is the velocity signal of the vibrating object. We consider here a statistical signal-processing approach to termite head banging data, which is based on Bayesian inference. In this approach, the quantity of interest is the frequency of vibration caused by termite head banging, while phase and quadrature amplitudes are considered nuisance parameters. We employ a single-frequency model to determine this frequency. Because of the optimal prior knowledge about the signal of interest, the frequency can be measured with much greater precision and greater noise immunity than using Fourier transform. Our results show that the Bayesian method of processing an acoustic signal exhibits excellent performance in determining the vibrational frequency.

Session 4pUW

Underwater Acoustics: Underwater Reverberation Measurements and Modeling II

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Contributed Papers

1:00

4pUW1. Reverberation modeling with BiRASP—The bistatic range-dependent active system prediction model. David Fromm (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5350, david.fromm@nrl.navy.mil)

The calculation of bistatic reverberation is a computationally intensive exercise. Efficient computation of timely results typically involves numerous physical and algorithmic assumptions or approximations. Consequently, it is not unusual for predicted time series from different models to differ in level or structure. At a recent Reverberation Modeling Workshop [9–13 November 2006, Austin, TX], a set of test cases were proposed with the intent of establishing benchmark reverberation problems and solutions. This talk will focus on the test case performance of BiRASP, the Naval Research Laboratory's bistatic reverberation model. BiRASP is a ray-based code, originally developed for deep-water, long-range, low-frequency reverberation predictions and data analysis. Results for the various test case problems will be shown. The consequences of some of the typical algorithm and physical assumptions/approximations will be demonstrated and discussed. [Work supported by the Office of Naval Research.]

1:15

4pUW2. Constrained comparison of ocean acoustic reverberation: Theory and observations. Charles Holland (The Penn State Univ., Appl. Res. Lab, State College, PA)

Measurements of long-range (order 10^4 m) shallow-water reverberation in the Straits of Sicily at 900 and 1800 Hz are compared with theoretical predictions. All of the required environmental inputs for the theory are obtained independently, that is to say there are no free parameters. The reflection coefficient and the scattering strength are measured by direct path methods; both quantities show strong frequency dependence. The theoretical reverberation predictions using these measurements are in good agreement with directional reverberation data, i.e., within the expected uncertainty bounds. The good agreement suggests that the supporting environmental measurement techniques are robust and that the physics associated with reverberation in a waveguide is reasonably well understood, at least in simple environments. The ability to independently measure the seabed scattering strength and reflection coefficient is a crucial step for the advancement of inverse methods using reverberation (e.g., rapid environmental assessment) inasmuch as it provides the means for quantitatively measuring the robustness of those methods. [Research supported by the Office of Naval Research OA321 and the NATO Undersea Research Centre.]

1:30

4pUW3. Reverberation modeling issues highlighted by the first Reverberation Modeling Workshop. Eric I. Thorsos (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu) and John S. Perkins (Naval Res. Lab., Washington, DC 20375)

Participants at Reverberation Modeling Workshop I presented solutions to posed reverberation problems using a wide variety of methods. The reverberation problems were posed in both two- and three-space dimensions, and included cases with different levels of boundary roughness, different sound speed profiles, and in some cases, range dependence. A number of important reverberation modeling issues became evident when the results of this workshop were considered in detail following the workshop. These include the important role of coherent effects in determining reverberation structure at short ranges, and the important role of boundary reflection loss in affecting the reverberation level at long ranges. A summary of modeling issues highlighted by this workshop will be presented. [Work supported by ONR and PMW-180.]

1:45

4pUW4. A comparison of two modeling approaches for reverberation in a shallow-water waveguide where the scattering arises from a sub-bottom interface. Dale D. Ellis (DRDC Atlantic, Dartmouth, NS Canada, dale.ellis@drdc-rddc.gc.ca) and Charles W. Holland (The Penn State Univ., State College, PA)

In shallow water environments with a low-speed sediment layer overlying a higher-speed sub-bottom, the observed reverberation may be dominated by scattering from the sub-bottom. Here, reverberation predictions from normal mode and energy flux models are compared for the case where the scattering arises from a sub-bottom half-space under the low-speed sediment layer. It is shown that in such an environment, the position of the angle of intromission, in addition to the angular dependence of the scattering kernel, is a factor controlling the vertical angle distribution. It is also shown that the reverberation from a sub-bottom horizon is typically governed by higher grazing angles than the case where the scattering occurs at the water-sediment interface. There was generally very close agreement between the models as a function of frequency (200–1,600 Hz), layer thickness (0–8 m), and range (1–15 km). The model comparisons, showing some differences, illuminate the effect of different approximations in the two approaches. [Work supported in part by the Office of Naval Research.]

2:00

4pUW5. A model of narrow-band normal-mode reverberation in shallow water. Anatoliy Ivakin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

A unified model of reverberation in a shallow-water waveguide caused by the volume heterogeneity and rough interfaces is proposed. Normal modes are used to describe propagation of a narrow-band signal from a point source to the scattering volume (a vertical column of the waveguide including both water and sediment) and from the scattering volume to the receiver. A local scattering matrix describes a process of reradiation from one normal mode to another within the scattering volume. The case of statistically axial symmetry, where the source and receiver are separated only in the vertical direction and the medium is statistically homogeneous and isotropic in the horizontal plane, is considered in more detail. A simple relationship of the temporal dependence of the reverberation intensity with the scattering matrix, attenuation, and group velocities of the normal modes is obtained. Contributions of different components of reverberation due to rough air-water and water-sediment interfaces, volume heterogeneity of the water column, and the sediment are discussed.

2:15

4pUW6. The effect of different environments on shallow-water reverberation: Measurements and modeling. Roger Gauss (Naval Res. Lab., Code 7144, Washington, DC 20375-5350) and John Preston (Appl. Res. Lab., The Penn State Univ., State College, PA 16804)

Low-frequency reverberation in shallow water is a complex and difficult quantity to predict as its mean levels can be due to a combination of seafloor, sea surface, and fish scattering, coupled with propagation and system-dependent effects. In this paper the importance of having a physics-based modeling capability and the correct spatially-dependent environmental description will be demonstrated via data-model comparisons corresponding to monostatic data the authors have collected over the last 15 years in a range of distinct acoustic environments, from silt to sand to chalk to basalt. Additionally, the use of in situ measurements to both refine predictions and mitigate uncertainty is discussed. [Work supported by ONR.]

2:30

4pUW7. Frequency and angular dependence of bottom scattering strength in shallow water with a sandy seabed. Ji-Xun Zhou and Xue-Zhen Zhang (Georgia Inst. of Technol., Atlanta, GA 30332-0405 and Inst. of Acoust., Chinese Acad. of Sci., Beijing 100080, China)

The bottom scattering strength at low frequencies and small grazing angles is difficult to directly measure in shallow water. It is generally derived from mid- and long-range reverberation measurements. Reliable estimation of the bottom scattering strength from shallow-water reverberation requires a correct reverberation model, quality reverberation data, and an appropriate seabed geo-acoustic model that controls two-way sound propagation. In this paper, an effective Biot model for sand-silt bottoms, derived from low-frequency field measurements at 18 locations in different coastal zones around the world is used [J. X. Zhou and X. Z. Zhang, J. Acoust. Soc. Am. **117**, 2494 (2005) and **119**, 3447 (2006)]. A simple closed form expression for the reverberation in shallow isovelocity water [J. X. Zhou, Acta Acustica **5**(2), 86–99(1980) in Chinese] as well as the normal-mode expression for the reverberation in shallow water with an arbitrary sound speed profile are used to derive the bottom scattering strength. The bottom scattering strength as a function of frequency and angle, derived from broadband reverberation data, will be reported and discussed. [Work supported by ONR.]

2:45

4pUW8. High frequency acoustic bottom backscatter at shallow grazing angles. Nicholas P. Chotiros (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78713-8029, chotiros@arlut.utexas.edu)

Variability is an important feature of high frequency acoustic bottom backscatter. A relationship had been developed between grain-size and backscattering strength in the laboratory, but it rarely applies in the field. The main problem is that real ocean sediments have surface roughness and volume scatterers or inclusions that are continually evolving. The result is a rather large range of variations as a function of space and time. In some cases, estimates of the variability may be obtained from the extant database of published measurements. [Work supported by the Office of Naval Research, Ocean Acoustics.]

3:00

4pUW9. Demonstration of coherent gain from synthetic aperture for a midfrequency sonar in shallow water. Kevin D. LePage (Naval Res. Lab, Code 7144, 4555 Overlook Ave SW, Washington, DC 20375)

The Clutter 07 experiment was carried out this May in the Straits of Sicily on board the R/Vs Alliance and Oceanus with the participation of NATO Undersea Research Centre, NRL, Applied Research Lab-Pennsylvania State University, and Defense Research and Development Canada. High repetition rate monostatic active sonar data collected off the alliance on the Malta Plateau during this experiment is processed to show significant inter-ping correlation indicative of the utility of performing coherent synthetic aperture sonar (SAS). Strip SAS images of scattering from a rocky outcrop on the bottom known as the Ragusa ridge show that the lateral extent of returns from compact objects can be reduced through SAS techniques for real shallow water environments using autofocusing techniques. [Work supported by ONR.]