Session 2pAA

Architectural Acoustics: Update on Acoustic Products, Treatments, and Solutions

Matthew V. Golden, Cochair
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Chair's Introduction—1:15

Invited Papers

1:20

2pAA1. Products focused on improving acoustic performance. Ron Freiheit (Wenger Corp., 555 Park Dr. Owatonna, MN 55060, ron.freiheit@wengercorp.com)

New product information with a focus on performance, practice, and rehearsal room acoustics will be presented. Performance-area acoustics are enhanced by full-stage acoustical shells, ceiling panels, and clouds for the audience area. These range from standard products to unique customizations. Portable shells that roll through standard doorways will also be shown. New modular sound-isolating practice rooms feature enhanced acoustic performance, improved aesthetic appearance, and upgraded optional floating floor. Active acoustics technology can be retrofit in an existing room, such as a teaching studio. Interesting applications of acoustical doors will also be highlighted. Other products include instrument storage cabinets with tunable absorption integrated into the design. An acoustic shield positioned behind an individual performer helps reduce potentially damaging sound energy from behind, as indicated by binaural measurements and recordings. Finally, an acoustician-specific, restricted-access website features third-party test data and other commonly requested technical information.

1:40

2pAA2. A re-introduction of pre-stressed molded fiberglass isolation pads for floating floors. Matthew Golden (Kinetics Noise Control, 6300 Irelan Pl., Dublin, OH 43017)

The pre-stressed molded fiberglass isolation pad has been available for over 50 years. In that time, no other material has been developed that exhibits the unique performance features of fiberglass. Fiberglass pads are a very non-linear spring. The dynamic stiffness of fiberglass pads also changes under load. Together these features allow them to provide consistent performance over a very wide load range. Recently, the next generation of pre-stressed molded fiberglass isolation pad has been introduced to the market. This talk will cover what has been improved with this new generation of pad along key performance dynamics of floating floors for acoustical and vibration isolation applications.

2:00

2pAA3. Sound rating of a new air handling product. Stephen J. Lind (Ingersoll Rand, Trane Bldg 12-1, 3600 Pammel Creek Rd., La Crosse, WI 54601)

It is important in building design to have an accurate acoustical description of equipment that will operate in the building. Noise from air conditioning systems is often a significant contributor to the building acoustic environment. In order to design the building correctly, the sound levels produced by the equipment must be known. Current theoretical or computer modeling methods are not able to accurately predict the resulting unit sound levels. The sound levels produced depend on many factors including equipment design, operating conditions, options chosen, and sound component. Empirical models are needed. The test method to develop the models follows Air Conditioning, Heating, and Refrigeration Institute Standard 260. Sound power levels for a range of fan sizes, types, and operating conditions are measured for the discharge, inlet, and casing sound components. Equipment options that affect the sound as it propagates through the equipment are also measured. The test results are used to create mathematical models to describe the unit sound power. The resulting mathematical models are then incorporated into a computer program to allow the user to accurately determine the sound in the product as configured for each job.

2:20

2pAA4. Solutions to retain acoustical functionality against the tide of fad, fashion, and finance. Kenneth Good (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

Too often in recent years, building acoustics have been the casualty of other design considerations. Changes in design styles, flexibility, and building densities have left many building occupants stressed and distracted. The Center for the Built Environment and other data suggest that the recent Green focus has made the situation even worse as the (acoustical) function of buildings is being compromised to fad, fashion, finance, and other considerations. This paper will look at the acoustical properties of products to meet these changes, solutions to maintain the acoustical functionality of interior spaces.

2:40

2pAA5. Active acoustic treatment. Roger W. Schwenke (Meyer Sound Labs., 2832 San Pablo Ave., Berkeley, CA 94702)

We call building materials "acoustic treatment" when they change the response of a room to sounds that occur in it. Absorption reduces the reverberation time of a room as well as strength. Reverberant chambers increase the cubic volume of the room and increase the reverberation time without significantly changing the strength. Diffusion changes the distribution of reflections in time and level. Electro-acoustic systems using microphones, signal processors, and speakers can also be used to change the response of a room to the sounds that occur in it. This paper will illustrate many ways that Electro-acoustics can be thought of as an active acoustic treatment—one of the many acoustic treatments acousticians have in their palette to achieve their design goals.

3:00

2pAA6. Acoustical performance of lightweight construction materials. Erin L. Dugan (USG Corp., Corporate Innovation Ctr., 700 N. US Hwy. 45, Libertyville, IL 60048, edugan@usg.com)

Traditionally, acoustical privacy between spaces has been achieved by adding denser, heavier materials to the partition. Recently, several lightweight construction materials have been released into the construction market. Advances in material science have provided a new generation of lighter weight gypsum panels with multiple benefits to builders. Although lighter and significantly easier for contractors to install, the reduced mass of the gypsum panels raises concerns about its effects on the sound attenuation performance of these products. However, analyses have shown that these lightweight gypsum panels are able to provide nearly comparable levels of sound transmission performance as their heavier counterparts. An investigation of the intrinsic mechanical properties contributing to the overall acoustical performance and full-scale sound transmission results is presented comparing wall systems constructed with lightweight panels and standard-weight panels. Alternative assemblies used to increase the sound-attenuating performance are also discussed.

3:20–3:35 Break

3:35

2pAA7. Use of damped drywall in architectural acoustics. Benjamin M. Shafer and Brandon Tinianov (Serious Energy, Inc., 1250 Elko Dr., Sunnyvale, CA 94089)

Damped drywall, specifically QuietRock, has been tested both by various NVLAP-accredited laboratories and *in situ* in a variety of assemblies as an acoustic treatment for the transmission of airborne sound through building partitions. The application of damped drywall and its use for noise control through building partitions continue to expand as the body of test data grows and analytic models are developed to support this class of materials. However, the manufacturer of QuietRock has conducted several research studies that illustrate how damped drywall can be used to improve the transmission loss over a broad range of frequencies and in various assemblies. This presentation is a summary of these research studies and serves as a guide for the application of damped drywall in building construction.

3:55

2pAA8. RPG Diffusor Systems: Overview of nearly 30 years of research and development. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

Each company is defined by its mission. RPG Diffusor System mission is to provide continuing innovation and expands the acoustical palette through a commitment to pioneering fundamental acoustics research. Rather than focus on any particular products, this presentation will provide an overview of nearly 30 years of research and development of a unique range of absorptive, reflective, and diffusive acoustic tools. With respect to absorption, the presentation will describe the progression from traditional fabric covered porous absorbers, to binary amplitude diffsorbers (attenuating the unneeded high frequency absorption), to absorptive wood Helmholtz absorbers, to transparent microperforated and microslit absorbers, to absorptive/diffusive CMU, membrane and plate resonator absorbers. With respect to reflection, we will describe the advancement from flat reflectors, to combined reflectors, to one and two-dimensional curved shapes in wood and glass reinforced gypsum. Finally, the evolution of diffusive surfaces from the venerable QRD in the early 1980s to modulated optimized diffusors, which provide unlimited bandwidth and freedom from grating lobes and flat plate limitations, to shape-optimized wood and glass reinforced gypsum spline, bicubic and other curvilinear architectural shapes. As part of the historical development, theoretical explanations, proof of performance metrics, and installation photos of completed projects will be provided.

Contributed Papers

4:15

2pAA9. Acoustic characteristics of floor treatments for elementary school classrooms. Ari M. Lesser, Adam P. Wells, Michelle C. Vigeant, and Robert D. Celmer (Acoust. Prog. and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117)

Phase 1 of this study [J. Acoust. Soc. Am. **129**(4), 2523 (2011)], determined the effects of hard versus soft flooring on overall speech and activity noise levels in elementary classrooms. A significant decrease in overall levels was found in carpeted rooms. This phase sought to investigate a range of floor materials and their pertinent properties. Nine different floor materials were mounted to 3 in. concrete slabs and evaluated using a battery of acoustic, impact, and chair scrape tests. Tested materials included vinyl composition tile, resilient rubber athletic flooring (virgin, blended/synthetic, and recycled), polyurethane, vinyl cushion tufted textile carpet, and rubber-backed commercial nylon carpet. Impedance tube measurements of sound absorption were made using International Organization for Standardization (ISO) 10534-2, while sound power measurements according to ISO 3741 were made while either (a) tapping on each mounted sample with a standard tapping machine or (b) while reciprocating an elementary classroom chair back and forth to produce repeatable scraping sounds. In general, the two carpet samples resulted in the lowest sound levels and the highest absorption. The relative performance of each material will be presented along with a discussion of additional usability factors, such as maintenance, cost, and durability. [Work supported by Paul S. Veneklasen Research Foundation.]

4:30

2pAA10. The dB focus tube uses to improve the transmission loss efficacy of walls, ceilings, and pipes. Bonnie Schnitta (SoundSense, LLC, 46 Newtown Ln., Ste. One, E. Hampton, NY 11937, bonnie@soundsense.com)

Studies indicate that locating and treating even very small acoustic leakage points in wall, floor, and/or ceiling partitions will significantly improve the FSTC performance of these partitions. The dB focus tube, patented on March 22 of this year, is an inexpensive small mechanism that can easily be transported to and from job sites. It is audible even during high levels of construction activity. The resulting patented small tube can be utilized by an acoustic engineer to locate even the smallest of acoustic leakage points in a partition which, when sealed, could increase the FSTC performance of the partition by typically 10 points. Additionally, when an engineer is trying to determine the main paths of flanking, this tube proves to be an ideal mechanism for facilitating this problem solving method. An acoustic installer can use the focus tube to confirm the optimum efficacy of the acoustic treatment of PVC piping or duct work. In addition, another interesting application is using the dB focus tube to locate air leaks in window assemblies which, when sealed, improve the window performance in regards to not only the STC and air infiltration, but also its thermal capacity.

4:45

2pAA11. Modeling treatments to reduce sound transmission through an open window into a room: Effect of window thickness. Caleb F. Sieck and Siu-Kit Lau (Architectural Eng., Univ. of Nebraska-Lincoln, 203C PKI, 1110 S 67th St., Omaha, NE 68182-0681, cfsieck@gmail.com)

Sound transmission loss through an exterior wall is limited by its weakest structure, generally a window, especially if it is open. Considering the acoustic modes within a window and a room are important because much of the acoustic energy from noise sources such as traffic and large wind turbines is in the low to middle frequency range. Previous models of open windows have either neglected the thickness of the window or the influence of room modes on the sound transmission. The present investigation considered a baffled rectangular aperture of finite thickness backed by a rigid walled cavity. An impedance/mobility approach was used to study the effect of the thickness of an open window on the insertion loss and sound pressure levels inside the cavity. The insertion loss study was confirmed using FEM modeling, and the difference in sound pressure levels was compared to experimental results. Increasing window thickness decreases the amount of sound transmitted at frequencies below the second order modes of the cavity for both window sizes under investigation. Using the impedance/mobility approach was effective in this study and allows the model to be easily extended.

5:00

2pAA12. Increasing diffuser efficiency through asymmetrical third dimension modification. Richard L. Lenz, Jr. (RealAcoustix LLC, 887 N. McCormick Way, Layton, UT 84041, RL@RealAcoustix.com)

In 1975, Dr. Manfred Schroeder presented his seminal work on quadratic-residue diffusion, giving us a frequency-based method for creating acoustic diffusers. The QRD equation defines the method for creating the depths (*Z* plane) of the wells in a diffuser. Subsequent work has given us understanding of the effects of the width of the diffuser wells (*X* plane) on high frequency performance. One artifact not addressed in the Schroeder equation is the inherent absorption realized in the execution of the design. This absorption becomes more pronounced as the design gets deeper, thereby limiting the practical low-frequency cutoff of QRDs. Experiments by the author, utilizing an asymmetrical cellular acoustic diffuser design, have yielded surprising results in the efficiency of acoustic diffusion. By reducing both the width of the cells and separating the diffuser into asymmetrical zones in the *X* and *Y* (vertical) planes, it has been shown that the absorption below the design cutoff frequency of the diffuser design can be dramatically reduced. The presentation will show the evidences of these experiments and the increased efficiency of QRD design through asymmetrical third dimension modification. [Work support was provided by Ron Sauro of NWAA Labs.]

5:15

2pAA13. Conference rooms: Criteria, guidelines, acoustical problems solutions (case studies). Jack B. Evans and Joshua D. Leasure (JEAcoust., 1705 W Koening Ln, Austin, TX 78756, Evans@JEAcoust.com)

Conference rooms are subject to privacy and sound containment issues, intrusive and distracting noise, excessive continuous background sound, and reflection/reverberation problems, particularly those with microphones and loudspeakers. Good acoustical environments are necessary for intelligible speech communications and remote signal transmissions, while sound isolation is needed for privacy and prevention of distraction. Acoustical criteria are presented with practical guideline parameters. Case studies cover acoustical problem issues that needed correction in existing conference rooms. On-site observations, acoustical measurements and analyses of facility plans were used to diagnose problems and determine correction approaches. Photographs are shown to illustrate difficult conditions. While little is original, the case studies point out classic problems of flutter echoes, ceiling-mounted or suspended microphones, back radiated loudspeaker noise in ceiling plenums, glass walls, multiple sound flanking paths, and similar problems found. Solutions developed or implemented are presented. In some cases, before and after data are provided to show results.

Session 2pAO

Acoustical Oceanography: Tomographic, Geoacoustic, and Ambient Noise Inversions

Shane C. Walker, Chair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

1:00

2pAO1. The build up rate of broadband spatial coherence between multi-sensor passive arrays in the ocean waveguide. Shane Walker (Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC 0238, La Jolla, CA 92093)

Broadband noise correlation methods for the passive extraction of information about the point-to-point propagation of waves between distant sensor locations have received considerable attention in the literature. This talk addresses the rate at which the wave coherence accumulates to overcome stochastic random fluctuations in a spatially correlated random wave field. It is shown that the expected magnitude of the random uncertainty associated with a single realization of the sample cross-correlation function depends on the total power incident on the sensors. This notion is applied to quantify the emergence rate of the coherence between correlated beams in the ocean waveguide. The point-to-point build-up rate is compared to the gain achieved through the application of directional filters over multi-sensor line arrays. Various array geometry scenarios of experimental interest are considered. These results are straight forwardly extendable to other environments such as seismics, helioseismics, and nondestrutive testing.

1:15

2pAO2. Low-frequency broadband noise correlation processing in deep-water. Stephanie Fried (Marine Physical Lab. of the Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238), Karim G. Sabra (Georgia Inst. of Technol., 771 Ferst Dr., NW Atlanta, GA 30332-0405), W. A. Kuperman (Univ. of California, San Diego, La Jolla, CA 92093-0238), and Mark K Prior (Preparatory Commission for the Comprehensive Nuclear-Test-Ban Treaty Organisation, 1400 Vienna, Austria)

The Comprehensive Nuclear-Test-Ban Treaty Organization operates an International Monitoring System (IMS). The IMS includes hydroacoustic stations composed of hydrophones deployed in the ocean deep-sound-channel in a two-kilometers-side triangular configuration(referred to as triad) in the horizontal plane. Data are continuously recorded on hydrophone triads (with a sampling frequency of 250 Hz) and have been archived during the last decade. Previous experimental studies have demonstrated that coherent waveform can be extracted from broadband coherent processing of ocean ambient noise, typically above $f > 100$ Hz [e.g., see Roux *et al.*, J. Acoust. Soc. Am. **116**(4), 1995–2003 (2004)] . We investigated here the emergence of coherent arrivals from the correlation processing of the low-frequency broadband ambient noise recorded during the years 2006–2007 on IMS hydrophones located in the Southern Hemisphere. This low-frequency acoustic ambient noise includes various components from anthropogenic and biological sources as well as from seismic origin (e.g., earthquakes and microseims) and also significant ice-breaking noise originating from Antarctica especially during the Austral summer period. The feasibility of passive basin scale tomography using long-term monitoring of ocean noise will be discussed.

1:30

2pAO3. Bayesian inversion of seabed interface-wave dispersion from ambient noise. Cuilin Li, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, B.C. V8W 3P6 Canada), Hefeng Dong (Norwegian Univ. of Sci. and Technol., NO-7491 Trondheim, Norway), Lanbo Liu (Univ. of Connecticut, Storrs, CT 06269), and Dingyong Yu (Ocean Univ. of China, Qingdao, China 266100)

This paper applies Bayesian inversion to estimate seabed shear-wave speed profiles and their uncertainties using interface-wave dispersion curves extracted from ocean ambient noise, and compares the resolution of seabed structure for fundamental-mode and multi-mode data. Ambient-noise recordings were collected for 2.15 h at hydrophones of an entrenched ocean-bottom cable in the North Sea. Scholte-wave dispersion curves for the fundamental mode and several higher-order modes within the frequency range 0.26–3.8 Hz are extracted from cross-correlations of noise recordings at sensor pairs via slowness-frequency transform. The Bayesian information criterion is used to determine the preferred model parameterizations in terms of the number of sediment layers supported by the data for inversions based on the fundamental mode alone and on the first three modes. Adaptive-hybrid optimization and Metropolis-Hastings sampling are applied to estimate the most-probable shear-wave speed models and to compute marginal posterior probability profiles. The results show quantitatively that multi-mode inversion provides higher-resolution of shear-wave speed structure at shallow depths and smaller uncertainties at all depths than inversion of the fundamental mode alone. [We thank StatOil for providing data.]

1:45

2pAO4. A computationally light method for simulating the physics and statistics of weak signals in spatially correlated random wave [and vibration] fields. Shane Walker (Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC 0238, La Jolla, CA 92093)

Ocean noise (natural plus man made) incident on an array has coherent and incoherent components. Simulations must accurately include both components so that the synthetic data are faithful to realistic scenarios wherein spatial correlations associated with random wave fields emerge over time. Indeed, noise simulations involving a large number of sensors over long time intervals can be quite computationally expensive. This talk introduces a computationally light method for simulating stochastic (time-domain) realizations of arbitrary duration of spatially correlated noise characteristic of the ocean soundscape. The method is well suited for studying the physics of weak signals in noise and provides an opportunity for studying the emergence of spatial correlations associated with random wave fields. While this talk focuses on passive SONAR simulations, the method can be generally applied to the study of random wave fields and vibrations in other environments such as seismics, helioseismics, and nondestrutive testing.

2:00

2pAO5. Coherent averaging of the passive fathometer response using short correlation time. James Traer and Peter Gerstoft (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr. La Jolla, CA 92093)

The passive fathometer algorithm was applied to data from two drifting array experiments in the Mediterranean, Boundary 2003 and 2004. The passive fathometer response was computed with correlation times from 0.34 to 90 s and, for correlation times less than a few seconds, the observed signal-to-noise ratio (SNR) agrees with a 1 D model of SNR of the passive fathometer response in an ideal waveguide. In the 2004 experiment, the fathometer response showed the array depth varied periodically with an amplitude of 1 m and a period of 7 s consistent with wave driven motion of the array. This introduced a destructive interference, which prevents the SNR growing with increasing correlation time. A peak-tracking algorithm applied to the fathometer response of experimental data was used to remove this motion allowing the coherent passive fathometer response to be averaged over several minutes without destructive interference. Multirate adaptive beamforming, using 90 s correlation time to form adaptive steer vectors which were applied to 0.34 s data snapshots, increases the SNR of the passive fathometer response.

2:15

2pAO6. Performance bounds on geoacoustic inversions. Arthur B. Baggeroer (Depts of Mech. and Elec. Eng., MIT, Cambridge, MA 02139)

Lower bounds on the mean square error for parameter estimates using geoinversion methods are developed in terms of the tomographic version of the Cramer Rao bound. Two approaches are considered: (i) an active, coherent source with known waveform and time synchronization as in ocean acoustic tomography (OAT) and (ii) a passive, broadband source with known spectrum but with no time synchronization as in matched field tomography (MFT). Both approaches assume the receiver to be an array, e.g., vertical line array or towed horizontal line array (HLA). The formulation requires specifying the Green's functions connecting source and the sensors of the receiving array. The bounds demonstrate (i) OAT performance depends upon the mean square group speed spread of modes and (ii) MFT performance depends upon the phase speed spread where both are integrated across the band of the respective sources. The bound also indicates the coupling among parameter estimates. Uncorrelated estimates are usually desirable for efficient parameterization formulations. An example using a Pekeris waveguide with three unknown parameters: the ocean and bottom sound speeds and the depth. The extension to Bayesian problems where a joint a priori probability is given. This permits the evaluation enabled by the observed data.

2:30

2pAO7. Passive geoacoustic inversion in a dispersive waveguide. Julien Bonnel, Cedric Gervaise (ENSTA-Bretagne, Pole STIC, 2 rue F. Verny, 29200 Brest, France, julien.bonnel@ensta-bretagne.fr), Barbara Nicolas, and Jerome Mars (GIPSA-Lab, Image-Signal Dept, Grenoble INP, France)

This study introduces a single-receiver geoacoustic inversion method adapted to shallow water and low frequency sources. Because of the single receiver context, most existing methods are based on the time-frequency (TF) analysis of the received signal, and their practical applications are restricted to impulsive sources. The proposed method is different and allows for geoacoustic inversion using unknown frequency-modulated sources. To perform the inversion, the modes are first filtered from the received signal using advanced TF analysis. Then, the filtered modes are processed in the frequency domain using a new transformation called modal reversal. This transformation, parametrized using environmental information, undoes dispersion for a given mode. When environment is well known, dispersion is perfectly compensated and all the reversed modes are in phase. This is not the case when modal reversal is ill-parametrized. Consequently, modal reversal can be used as the core of an original inversion scheme. Inversion results are obtain through a specific cost function adding up the reversed modes. Source/receiver range and source frequency-law are obtained as a by-product of the geoacoustic inversion. The method can thus be adapted to the study of low-frequency calls of marine mammals in shallow water.

2:45

2pAO8. Seismo-acoustic propagation effects at the seafloor with applications to geoacoustic inversion using hybrid parabolic equation solutions. Adam M. Metzler (Rensselar Polytechnic Inst., 110 8th St., Troy, NY 12180) and Jon M. Collis (Colorado School of Mines, Golden, CO 80401)

Accurate and efficient prediction of propagation over realistic models of elastic ocean sediments has been achieved recently using parabolic equations. Historic treatments that ignore effects due to elasticity may not be accurate for certain scenarios, specifically for example shallow-water environments. Neglecting elastic phenomena does not account for second order effects such as energy loss due to frequency dependent attenuation. In this presentation, a three-layer ocean environment is examined consisting of a fluid overlying a transitional solid layer overlying an elastic basement. The transitional solid layer is investigated for various types of media, including fluid, poro-elastic, and elastic. Solutions obtained from hybrid parabolic equations are used to quantify the nature of the transitional solid layer and establish regimes where each approximation would be appropriate. An application in geoacoustic inversion to identify bottom type is shown through comparisons of interface wave structure for different transitional solid layers. [Work supported by the ONR.]

3:00–3:15 Break

3:15

2pAO9. Stabilizing reverberation inversion by regression relations involving a grain size parameter. L. Abrahamsson, B. L. Andersson, S. Ivansson, J. Pihl (Swedish Defence Res. Agency, Underwater Research, 16490 Stockholm, Sweden, leif.abrahamsson@foi.se), M. Ainslie, F. Benders, M. Colin, R. van Vossen (TNO Defence, Security and Safety, The Hague, Netherlands), B. Chalindar, X. Cristol, B. Juhel (Thales Underwater Systems, 06560 Sophia-Antipolis, France), J. Dybedal, G. K. Olsen (Kongsberg Defence Systems, 7501 Stjordal, Norway), and E. Eidem (FFI, Forsvarets forskningsinstitutt, N-3191 Horten, Norway)

Within the European Defence Agency (EDA) project Rumble-2, an operational low-frequency active sonar system has been used to collect reverberation data at several sea trials in the North Sea. A global optimization method is used to determine the bottom parameters that provide the best match between measured and modeled time traces. A fast ray model is used for the forward computations. The bottom parameters are the Lambert back-scattering parameter and the sound speed c, density ρ, absorption α, and thickness of the sediment. The reverberation data do not constrain all these parameters to unique values, however, and different approaches have been tried in the project to reduce the ambiguity problems. The approach reported here is to use the mean grain size M_z as a common descriptive parameter. From regression relations by Hamilton and Bachman, c, ρ, and α can be set as functions of M_z. More ambitiously, the regression relations could be applied as *a priori* constraints, with uncertainties, in a Bayesian framework. The obtained inversion results are consistent with ground truth for the grain size, as measured from bottom samples. Moreover, similar results are obtained for trials in the same area with quite different environmental conditions.

3:30

2pAO10. Geoacoustic inversion in shallow water using broadband synthetic aperture and single hydrophone acoustic data. Bien Aik Tan, Peter Gerstoft, Caglar Yardim, and William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238)

A majority of geoacoustic inversion experiments sample the acoustic field on long arrays. This paper uses a moving single hydrophone to create a large synthetic aperture array for geoacoustic inversion. Practically, this is operationally attractive compared to using long arrays. For example, one possible application is the use of autonomous underwater vehicles (AUVs) to perform acoustic field sampling and pre-processing for geoacoustic inversion. The approach comprises synthetic aperture formation and a directed Monte Carlo Bayesian broadband frequency coherent geoacoustic inversion. This is demonstrated with simulated and real data from the MAPEX2000 experiment conducted by the NATO Undersea Research Center, using only one hydrophone of a towed array and a moored source in the Mediterranean Sea. The method yielded similar results compared to an equivalent physical array.

3:45

2pAO11. Estimation of shear speed using interface wave dispersion. Jennifer Giard, Gopu R. Potty, James H. Miller, Jeannette M. Greene (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882), Andrew R. McNeese, and Preston S. Wilson (The Univ. of Texas at Austin, 1 Univ. Station C2200, Austin, TX 78712)

Our recent work has highlighted the effect of shear on the dispersion of acoustic normal modes. Specifically, sediment shear speed can significantly impact compressional modal arrival times near the airy phase. In addition to underwater acoustic propagation effects, shear properties of sediments are directly related to the strength of the sediments in geotechnical applications. All of these factors emphasize the importance of estimating shear speeds in semi-consolidated shallow water sediments. One of the most promising approaches to estimate shear speed is to invert the shear speed profile using the dispersion of interface waves (Scholte waves). The propagation speed and attenuation of the Scholte wave are closely related to shear-wave speed and attenuation over a depth of 1–2 wavelengths into the seabed. Results from the tests conducted at Narragansett Bay in water depths ranging from 10 to 25 m using the shear measurement system, developed at the University of Rhode Island based on this concept, will be presented. Combustive sound source (CSS) will be used to generate interface waves. Data collected during these tests will be shown and preliminary estimates of the shear speed will be presented and compared with ground truth data. [Work supported by Office of Naval Research.]

4:00

2pAO12. Perspective of tomography inversion using direction-of-arrival and direction-of-departure. Florian Aulanier (Gipsa Lab., DIS, Grenoble INP, 961 rue de la Houille blanche, BP 46, F-38402 Grenoble Cedex, France, Florian.Aulanier@gipsa-lab.grenoble-inp.fr), Barbara Nicolas (Gipsa Lab, DIS, Grenoble INP, 38402 Saint Martin d'Hères, France), Philippe Roux (ISTerre, Maison des Gosciences, 38400 Saint Martin d'Hères, France), and Jèrôme Mars (Gipsa Lab, DIS, Grenoble INP, 38402 Saint Martin d'Hères, France)

In the ocean, local sound speed variations induce acoustic path changes. Travel-time (TT) variations of acoustic paths are classically used to perform ocean tomography inversion. Initially introduced to cope with multi-arrival interferences and to separate eigenray paths, source–receiver arrays combined with array processing techniques now give access to new observables that could be used for tomography such as direction-of-arrivals (DOAs) and direction of departure (DOD). The cumulative use of TT, DOA, and DOD in the inversion process first requires to study the forward problem which links sound speed variations to these observables measured through array processing from two source–receiver arrays. The so-called sensitivity kernels are established using (1) the first order Born approximation that relates the sound speed variation to the amplitude and phase change of the perturbed received signal and (2) a first order Taylor development which links the received signal perturbations to the relevant observables. In the present work, theoretical TT, DOA, and DOD sensitivity kernels are compared with parabolic equation simulations and tank experiment estimations.

4:15

2pAO13. Interferometry for three-dimensional acoustics in shallow water. Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE 19716), Boris G. Katsnelson (Voronezh State Univ., Voronezh, Russia 394006), and James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Interferometry in optics has been known for decades. It requires identification of specific refracted rays via some reflecting front and existence of a mechanism for interference between rays to occur (i.e., constructive and destructive amplitude and phase information). In underwater acoustics, the principal phenomenon is the same. However, identification of the refracted (or reflected) rays requires a well defined geometry between the acoustic source and receiver path and the reflecting (or refracting) front. The interference between the direct and reflected wave fronts from sea surface has been shown for some time. Recent experimental observations have reported the identification of the interference in the horizontal plane [J. Acoust. Soc. Am. **129**(4), EL141 (2011)]. Theoretical description of the horizontal interference phenomenon as well as the follow up work for potential use of interferometry techniques for three dimensional acoustic wave propagation in shallow water is presented. Theoretical and experimental results are shown. [Work supported by ONR.]

4:30

2pAO14. Asymptotic behavior of vertical travel-time sensitivity kernels with increasing propagation range. E. K. Skarsoulis (Inst. of Appl. and Computational Mathematics, Foundation for Res. and Technol. Hellas, 100 N. Plastira St., 70013 Heraklion, Crete, Greece), B. D. Cornuelle, and M. A. Dzieciuch (Univ. of California, San Diego, CA 92093-0230)

Vertical travel-time sensitivity kernels (VTSKs) describe the effect of horizontally uniform sound-speed changes on travel-times in range-independent environments. Wave-theoretic VTSKs can be obtained either analytically, through perturbation of the normal-mode representation, or numerically, as horizontal marginals of the corresponding 2D and 3D travel-time sensitivity kernels. In previous works it has been observed that, as the propagation range increases, wave-theoretic VTSKs approach the corresponding ray-theoretic sensitivity kernels even for low frequencies. In the present work an asymptotic expression of the wave-theoretic finite-frequency VTSKs is obtained, using a stationary-phase approach. Numerical results show that wave-theoretic VTSKs converge with increasing range toward the asymptotic form, which in turn lies very close to the ray-theoretic VTSK. [Work supported by ONR.]

4:45

2pAO15. Automatic detection of the number of raypaths in colored noise using short-length samples. Longyu Jiang and Jérôme Mars (GIPSA-Lab/DIS, Grenoble INP, 961 rue de la Houille Blanche BP 46F, 38402, Grenoble Cedex, Long-Yu.Jiang@gipsa-lab.grenoble-inp.fr)

In ocean acoustic tomography (OAT) (especially in shallow water where raypaths are mixed), knowledge of the number of raypath is crucial for inversion algorithm. In this paper, a noise-whitening exponential fitting test (NWEFT) is presented in this context for detecting the number of raypaths. Classically, two suggested approaches are the Akaike information criterion (AIC) and the minimum description length (MDL). Based on ideal assumption of ergodic Gaussian random processes and white Gaussian noise, MDL is shown to be asymptotically consistent, whereas the AIC tends to overestimate the order of model. However, these assumptions could not be fulfilled in practical case of OAT. In order to be adapted for real case of OAT, noise-whitening processing is applied as first step. Then, NWEFT bases on the fact that the profile of the ordered eigenvalues fits an exponential law for short-length samples of white Gaussian noise. The number of raypaths could be detected when a mismatch occurs between observed profile and exponential model. The fact that NWFET works on short-length samples is very important as a long duration of the received signal in OAT is unavailable. Its performance is studied with synthetic and real data set and compared with classical algorithms.

5:00

2pAO16. Krylov methods in inverse scattering and imaging. Paul E. Barbone, Gonzalo R. Feijóo (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu), and Assad A. Oberai (Dept. of Mech., Aerosp. and Nuclear Eng., Rensselaer Polytechnic Inst., 5048 JEC, 110 8th St., Troy, NY 12180)

Inverse scattering requires the measurement and inversion of a compact operator. The compactness of the operator implies that its range is low dimensional (i.e., sparse.) This implies the theoretical possibility of measuring the full operator with relatively few measurements and inverting it on a sparse basis. One issue, however, is that the basis on which the operator is sparse is unknown *a priori*. We show that Krylov methods can be used to simultaneously identify an efficient basis for the measurements and facilitate the inversion for imaging purposes. In particular, we show how imaging via Multiple SIgnal Classification (MUSIC) and by Krisch's factorization

method can be efficiently implemented in a Krylov space context. This method allows us to make most efficient use of *all* available acoustic sensors with few measurements and with minimal mutual interference.

5:15

2pAO17. Passive matched-field inversion using a horizontal planar array. Donald R. DelBalzo and James H. Leclere (QinetiQ North America, 40201 Hwy 190 East, Slidell, LA 70461)

Shallow-water acoustic predictions are severely limited by uncertainty in sediment property characteristics. Inverse methods with controlled active sources and vertical arrays have been used to estimate seabed properties; however, some applications require a covert approach and horizontal bottomed arrays. This study addresses the accuracy of low-frequency (100–200 Hz) matched-field correlations using broadband signals from surface ships with unknown source levels at unknown ranges. Matched-field techniques are applied in a realistic shallow-water environment with a horizontal planar array and high signal-to-noise ratios. The simulations indicate significant potential for accurate estimates of thick-sediment characterizations of grain size out to ranges of tens of water depths in shallow water, despite moderate mismatch conditions in the environmental model. The results show that: (1) the horizontal aperture should contain at least three hydrophones per wavelength to ensure high quality inversions; (2) the horizontal aperture should be several times longer than a vertical aperture; (3) coherent (phase-only) matched-field processing outperforms standard intensity processing by about 2 dB in good input SNR conditions; (4) incorrect assumptions about the assumed sound-speed profile (e.g., incorrect mixed-layer-depth) do not significantly affect the inversion results. [Work sponsored by QinetiQ North America.]

TUESDAY AFTERNOON, 1 NOVEMBER 2011 PACIFIC SALON 6/7, 1:00 TO 5:00 P.M.

Session 2pEA

Engineering Acoustics: General Topics in Engineering Acoustics

David A. Brown, Cochair
Electroacoustic Research Lab., Univ. of Massachusetts, Dartmouth, 151 Martine St., Falls River, MA 02723

Sairajan Sarangapani, Cochair
Ocean Engineering, Univ. of Rhode Island, 217 Sheets Bldg., Narragansett, RI 02882

Contributed Papers

1:00

2pEA1. Scattering reduction of an acoustically hard cylinder covered with layered pentamode metamaterials. Jeffrey E. Boisvert (NAVSEA Div. Newport, 1176 Howell St., Newport, RI 02841), Clyde L. Scandrett (Naval Postgrad. School, Monterey, CA 93943), and Thomas R. Howarth (NAVSEA Div. Newport, Newport, RI 02841)

Unlike traditional target strength reduction coatings that rely on energy dissipation or other mechanisms to mitigate reflection, a coating comprised of metamaterials would behave as an acoustic waveguide that diverts sound energy around the object, thus reducing scattered energy. The majority of the literature has featured theoretical ideal metamaterial designs that have unrealizable properties, i.e., infinite mass, vanishing bulk modulus. However, our analysis has suggested that it may be possible to obtain effective scattering reductions with realizable material properties in a layered configuration using metafluids. In this context, realizable implies material properties that are constrained to lie within reasonable ranges relative to the density and bulk modulus of water. The multistatic scattering reduction of an acoustically hard cylinder covered with layered metafluids for plane wave incidence is analyzed. A range of coatings are considered, from those comprised of fluid layers that are isotropic in bulk moduli with anisotropic density (inertia) to those having anisotropic bulk moduli and isotropic density (pentamode). [Work supported by NAVSEA Division Newport ILIR.]

1:15

2pEA2. An acoustic directional antenna with isotropic materials. Christopher N. Layman, Jr., Theodore P. Martin, and Gregory J. Orris (Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375-5320)

New acoustic metamaterial devices offer promising applications, ranging from tunable sound blocking with superior efficiency to acoustical diodes. Transformation acoustics (TA), relying on the invariance of field equations under coordinate transformations, in conjunction with metamaterial features, has further expanded the range of functionalized acoustic materials. However, for a large variety of devices based on TA, such as ones relying on Pendry's concept, physical realization remains limited owing to the requirement of anisotropic effective properties. Here we examine the behavior of a directional four-wave acoustic antenna, designed from finite embedded coordinate transformations (FECT), which eliminate the constraint of anisotropy by way of a suitable conformal mapping. The two dimensional antenna consist of a square rod with an inhomogeneous and isotropic distributed mass density and bulk modulus designed from the FECT perspective. The rod is imbedded in an acoustically matched matrix and subsequently subjected to an internal coaxially line source to evaluate its transmitting performance across a large bandwidth. Passive characteristics are also studied by probing the antenna with a point source-receiver setup. Experimental data are compared to both established analytical FECT models and full-wave simulations. [Work is supported by the Office of Naval Research.]

1:30

2pEA3. Compact directional acoustic sensing using multi-fiber optical probes. Joseph Bucaro (Excet, Inc., 4555 Overlook Ave., Code 7130, Washington, DC 20375-5350, joseph.bucaro.ctr@nrl.navy.mil), Nicholas Lagakos (Sotera Defense Solutions, Crofton, MD 21114), Brian Houston, Saikat Dey, and Maxim Zalalutdinov (Naval Res. Lab., Washington, DC 20375-5350)

A compact directional acoustic sensor concept is described, which uses an multi-optical fiber probe, a light emitting diode source, a photo-diode detector, and a short, slender cylindrical cantilever to the end of which is attached an optical reflector. A portion of the light exiting one fiber is collected by a second fiber after reflection from the mirror. Acoustically induced transverse displacement of the cantilever tip modulates the light collected by the second fiber, which then conveys the light to a photo-detector. Directional sensitivity is achieved by virtue of the dependence of the collected light on the cosine of the angle between the line connecting the probe fiber centers and the direction of displacement of the cantilever tip (the acoustic wave direction). An analytic model of the acoustic response of the cantilever tip is constructed, which is partially verified using a finite element-based model and experimentally validated using measurements of the acoustic response in air. The model is used to predict its acoustic response versus frequency, how that response depends upon damping near the cantilever resonance frequency, and to what extent and over what frequency band that response depends upon the acoustically generated flow force. [Work supported by ONR.]

1:45

2pEA4. Synchronized vibrations measurements at multiple locations using a single continuously scanning laser doppler vibrometer. Applications to non-contact sensing of human body vibrations. Muhammad Salman and Karim G. Sabra (School of Mech. Eng. Georgia Inst. of Technol., 771 Ferst Dr., NW Atlanta, GA 30332)

Laser Doppler vibrometry (LDV) is a non-contact method to measure surface velocity. Typical LDV configurations uses one fixed laser (He-Ne) beam at a specific point and orientation on a surface under test. A continuously scanning laser Doppler vibrometry (CSLDV) technique using a single laser beam performing quickly a long scan was developed to measure surface velocity at each laser position during its scan. This fast CSLDV can then replace several fixed LDVs. This technique is especially advantageous for sensing human body natural vibrations (typically below 100 Hz) at multiple locations (e.g., along small muscles) as it does not require traditional skin-mounted sensors (e.g., accelerometers array), which eliminates mass artifacts and the set-up time to attach those sensors. Experimental measurements were conducted using a CSLDV with a 200 Hz linear scan rate, over scan lengths up to 5 cm, to measure low-frequency vibrations (f<100 Hz) on gel samples which mimic human soft tissues. Validations of the CSLDV measurements were done using an array of several fixed LDVs distributed along the same scan line. The effects of speckle noise on CSLDV measurements will be quantified. Applications of this CSLDV technique for active and passive elastography measurements will be presented.

2:00

2pEA5. Response surface optimization of a directional endfire microphone array for hearing aids. Thomas Burns (Starkey Labs, Inc., 6600 Washington Ave. S., Eden Prairie, MN 55344, tburns@starkey.com)

The optimal operating parameters for a directional microphone array worn *in situ* are not necessarily equivalent to the optimal parameters while operating in the absence of head and torso related scattering. These parameters include the relative magnitude and phase of the microphones and their positional placement on the head, characterized as factors, operating over a range of levels, characterized by their production spread and susceptibility to drift. The goal is to understand how these factors operating over their levels contribute to the *in-situ* directional responses on a measurement manikin, characterized by the directivity index and the unidirectional index. Using 614 impulse responses acquired in ten deg resolution on the manikin, a simple central composite design of experiments was conducted to fit a quadratic polynomial and generate a response surface to the aforementioned directional indices, thereby leading to the critical first-order and two factor interactions of the system. The interactions, statistical validity of the predictive polynomials, and the sweet spot of operation will be described for some common hearing aid microphone arrays.

2:15

2pEA6. Optimization of tuning and matching of broadband transducers with power switching amplifiers. Corey Bachand, Boris Aronov, and David A. Brown (BTech Acoust. LLC, ATMC, UMass Dartmouth, 151 Martine St., Fall River, MA 02723, corey.bachand@cox.net)

Underwater transducers for broadband communication rely on effective tuning and matching to a power amplifier for maximum signal bandwidth and efficiency. This analysis follows a systematic approach to design an efficient and effective broadband acoustic transmit system. Power switching (class D) amplifiers use a variety of modulation schemes to reduce the losses incurred at the high power amplification stage. Lowpass filtering at the output stage of the switching amplifier is often employed to attenuate the high frequency carrier signal from the modulation stage. A matching transformer steps up the voltage delivered to the transducer. The tuning network can be designed to provide optimum cancellation of reactance over a wide band, thus improving the power factor bandwidth.

2:30

2pEA7. Development of an optical transducer for an electro-acoustic guitar. S. K. Cho and Y. W. Park (Dept. of Mechatronics Eng., Chungnam Natl. Univ., 99 Daehangno, Yuseong-gu, Daejeon 305-764, Korea, chosk79@nate.com)

This paper presents the development of an optical transducer (OT) for an electro-acoustic guitar. Two conceptual designs are proposed: one with one infrared light emitting diode (IR LED) and one photodetector, and the other with two IR LEDs and one photodetector. Both concepts are based on the top-to-bottom structure: IR LED is on the top, and the photodetector is at the bottom. After the preliminary tests, the latter design is selected as the proposed OT. The OT is fabricated on the PCB with proper electronic circuit, and mounted on the guitar. The developed OT is subjected to the performance evaluation with a dedicated measuring device. The performance of the OT is compared with commonly used piezoelectric transducer. Findings are summarized: (1) The output signals from the OT are much higher that those from the piezoelectric transducer in both average and peak-to-peak voltages. (2) The noise level from the OT is similar or less than that from the piezoelectric transducer. (3) SNR with the OT is increased by 45% in average, compared with the piezoelectric transducer.

2:45

2pEA8. Calculating piezoelectric parameters of stripe-electroded cylinders and bars with continuous no-uniform electric fields. Sairajan Sarangapani, David A. Brown (Acoust. Res. Lab., Adv. Technol. and Manufacturing Ctr. and Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth,151 Martine St., Fall River, MA 02723, ssarangapani@umassd.edu), and Boris Aronov (BTech Acoust., LLC, Fall River, MA 02723)

Tangentially poled thin-walled stripe-electroded piezoelectric bars and hollow cylinders are used in several electromechanical and electroacoustic applications. The 33 mode properties of the stripe-electroded bars and hollow cylinders are not fully realized due to the non-uniform polarization and non-uniform operational electric field due to fringing from the surface electrode geometry. A numerical finite difference method (FDM) was used to analyze the nonuniform electric field in the bar but under the assumption that the piezoelectric element is fully polarized. The effective electromechanical coupling coefficient $k_{3'3}$, the effective piezoelectric modulus $d_{3'3}$, the effective compliance $s^E_{3'3}$, and the effective relative dielectric constant $\epsilon_{3'3}$ (where the prime indicates field nonuniformity) are calculated using the energy method by accounting for the effects of nonuniform operational electric field and non uniform strain distributions. Analytical and experimental results are in good agreement and design optimizations are presented.

3:00–3:15 Break

3:15

2pEA9. Modeling piezoelectric parameters of bending mode transducers. Sairajan Sarangapani, David A. Brown (Acoust. Res. Lab., Adv. Technol. and Manufacturing Ctr. and Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723 ssarangapani@umassd.edu), and Boris Aronov (BTech Acoust., LLC, Fall River, MA 02723)

The piezoelectric bender bar vibrator is commonly used for generating low frequency flexural plate mode vibrations. This study considers the excitation of "benders" using stripe-electroded piezoelectric elements of various electrode patterns and poling configurations using the $k_{3'3}$ and $k_{3'1}$, where the prime indicates field nonuniformity. A numerical analysis is used to calculate the nonuniform electric field lines and the corresponding coupling coefficients as well as other related electromechanical parameters by developing a Lagrangian description of the electromechanical body and using the energy method. The effective electromechanical coupling coefficients are calculated by taking into account the internal energies due to transverse, longitudinal, and shear vibrations in the bar. The contribution of the passive and the active capacitance is also explained and taken into account while calculating the electromechanical parameters.

3:30

2pEA10. Limits of dissipative coefficients in piezoelectric transverse isotropic materials. Gordon E. Martin (3675 Syracuse Ave., San Diego, CA 92122, gemartin@ieee.org)

This paper relates to limiting forms of complex coefficients in passive piezoelectric systems due to hysteretic dissipation The specific application is polarized ferroelectric ceramic materials. The research required three features. (1) Mathematical models require the theory of physically realizable networks with distributed components like electrical transmission lines. Such theory is well established for discrete components and special forms of distributed components. Theory for the general case is reported here. (2) Passive systems cannot create energy so complex coefficients of mathematical models must have limiting values for both real and imaginary coefficients of the constitutive matrices. (3) The mathematical model can be expressed in complex symmetric form. It is proved all imaginary parts of matrix coefficients have changes of first order due to small dissipative perturbations. For piezoelectric spectroscopy purposes, measured immittances have two kinds of limits due to the addition of small dissipative effects. They cause small changes in complex values and corresponding frequencies of zeros and poles. It is proved (a) all critical immittance values have small changes to first order and (b) all corresponding frequencies have small shifts to second order.

3:45

2pEA11. Evaluating piezocrystal and piezoceramic transducer bandwidth and effectiveness. Corey Bachand, David A. Brown, and Boris Aronov (BTech Acoust. LLC, ATMC, UMass Dartmouth, 151 Martine St., Fall River, MA 02723, corey.bachand@cox.net)

Conventional piezoceramic transducers offer moderate bandwidth and performance to serve the majority of underwater acoustic applications. The manufacturability of piezoceramic elements in a variety of shapes (bars, cylinders, and hemispheres) makes them a cost-effective solution in may transducer designs. However, the emergence of piezocrystals in transducer designs has significantly increased the usable bandwidth while reducing the device footprint. This is enabling in terms of size and weight for use on mobile platforms (UUVs), especially when considering that one piezocrystal transducer may replace several piezoceramic transducers and reduce the number of hardware (power amplifier) channels. There are still fabrication and operational challenges with piezocrystal transducers that need to be overcome before they are widely adopted in the underwater community.

4:00

2pEA12. Design for a modular and scalable sonar source using displacement amplifying lamina. Richard H. Lyon (60 Prentiss Ln., Belmont, MA 02478-2021, rhlyon@lyoncorp.com)

A sonar sound source is described that is capable of radiating increased sound power at low frequencies in a non-resonant mode of operation. Non-resonant operation is used so that the amplitude and phase of the generated signals are smooth over a range of frequencies. The improvement in output is achieved in part by the use of Galfenol, a fairly new magnetostrictive (MS) material with a high MS strain coefficient. The enhanced output is also due to the use of non-resonant amplifying volume displacing lamina. The system is modeled using a simplified equivalent circuit linear model that allows prediction of several quantities of interest, such as the radiated output, the strain in the MS material, the excitation power, and the sensitivity to ambient pressure fluctuations. [Work supported in part by the US Air Force and the US Navy.]

4:15

2pEA13. Comparison of three experimental methods for assessing the blocked electrical impedance of a moving-coil loudspeaker driver. Daniel R. Marquez, Timothy W. Leishman, and Rex P. Price (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N-283 Eyring Sci. Ctr., Provo, UT 84602, danmarquez7@yahoo.com)

In this paper, three methods are discussed for experimentally isolating the blocked electrical impedance of a loudspeaker driver. The first involves measurement of the voice-coil impedance in a vacuum, before and after an added mass is applied to its cone. The second involves the use of a scanning laser Doppler vibrometer in conjunction with frequency-dependent electrical measurements at the driver terminals. The third involves the traditional destructive method of potting the driver in a hard-drying compound to allow direct measurement of the blocked impedance. The advantages and disadvantages of each method are discussed. The impedances determined by the three methods are used to predict the frequency-dependent cone velocities of several drivers while under operation. Actual measured velocities are compared with the predictions to substantiate the accuracy of each method.

4:30

2pEA14. Comparison and verification of analogous circuit models for dynamic moving-coil transducers. Rex P. Price, Daniel R. Marquez, and Timothy W. Leishman (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., ESC N-283, Provo, UT 84602, rexprice@yahoo.com)

For decades, analogous circuits have been used to model the electro-mechano-acoustical properties of moving-coil transducers. In recent years, many enhanced models have been proposed to improve the accuracy of their estimated voice-coil impedances. In this work, an iterative complex curve-fitting routine has been used to best fit and compare the various models to the measured complex input impedance data of several loudspeaker drivers in free air. The estimated blocked electrical impedance of each driver was extracted and used to predict its frequency-dependent cone velocity while under operation. Actual driver cone velocities were measured experimentally using a scanning laser Doppler vibrometer, and the predictions were compared to further substantiate the accuracy of each model. The results highlight those models with the greatest predictive capabilities and reliability.

4:45

2pEA15. Modeling and validation of magnetostrictive sound transducer including flat panel. H. J. Park and Y.W. Park (Dept. of Mechatronics Eng., Chungnam Natl. Univ., 99 Daehangno, Yuseong-gu, Daejeon 305-764, Korea, imal_hjpark@cnu.ac.kr)

This paper contains models of the magnetostrictive actuator and flat panel for the investigation of interaction between actuator and flat panel. (1) A transfer function of the magnetostrictive actuator between a displacement U_a and input current I: $G_a(s) = U_a(s)/I(s) = nd/(1 + c/c_o + sr/c_o + s^2m/c_o)$, where n is number of coil turns, d is magnetostrictive constant, c is stiffness of prestress spring, c_o is open circuit stiffness, r is damping coefficient, and m is effective mass. (2) Transfer functions of a flat panel: $G_f(s) = U_f(s)/U_a(s) = -A/(-B + C + D)$, $A = m_1k_1s^2/(m_1s^2 + c_1s + k)$, $B = m_2s^2 + (c_1 + c_2)s + (k_1 + k_2)$, $C = (c_1s + k_1)^2/(m_1s^2 + c_1s + k_1)$, $D = (c_2s + k_2)^2/(m_3s^2 + (c_2 + c_3)s + k_2 + k_3)$, where m, c, and k are parameters of the actuator and flat panel models, and parameters are determined experimentally. The final transfer function of actuator and flat panel is expressed by multiplying $G_a(s)$ and $G_f(s)$. Simulations are performed through commercial program under the conditions applying a white noise to the final transfer function. The simulated and experimental frequency responses are compared.

Session 2pEDa

Education in Acoustics and Physical Acoustics: Tools for Teaching Advanced Acoustics II

Kent L. Gee, Cochair
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Scott D. Sommerfeldt, Cochair
Dept. of Physics and Astronomy, Brigham Young Univ., N181 Eyring Science Center, Provo, UT 84602

Invited Papers

1:00

2pEDa1. Systematic circuit model construction for complex interconnected acoustic systems. Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78712-0292)

Complex, highly interconnected acoustic systems can be difficult to model for students and inexperienced practitioners. The systematic lumped-element circuit model construction method presented here is easy to learn and teach and allows for quick and easy, error-free circuit model construction. The present author discovered the method in a book by Mario Rossi [*Acoustics and Electroacoustics*, Artech House Publishers (1988)] and has included it in the electroacoustic transducers course taught at The University of Texas at Austin since 2003.

1:20

2pEDa2. Fourier: Making Waves—An interactive simulation for visualizing fourier analysis. Wendy K. Adams (Dept. of Phys., Univ. of Northern Colorado, CB 127, Greeley, CO 80639)

In this presentation, the PhET simulation "Fourier: Making Waves" http://PhET.colorado.edu/en/simulation/fourier, will be presented including the research behind the simulation, how students react to it and ideas for use in class. Students typically learn the math needed to do Fourier transforms and learn how to express a function in time or space and in terms of wavelength, wave number, or mode. However, many of these relationships are only memorized for the short term (exam) and are not retained. This simulation is designed to help students visualize how a combination of simple sines and cosines can create a more complicated function and listen to the sounds produced by each harmonic. This simulation features 11 adjustable harmonics which can be used to demonstrate various auditory perceptions. There is also a game tab with ten different levels that challenges students to choose the correct harmonics to match more and more complicated functions. For more mathematical explorations, students can investigate each of the symbols lambda, T, k, omega, and n to learn what each represents on the graph and their relationships with one another. Finally there is a tab to help students visualize moving from a discrete to a continuous series.

1:40

2pEDa3. Understanding sound wave propagation using computer animations. Jorge P. Arenas (Inst. of Acoust., Univ. Austral of Chile, P.O. Box 567, Valdivia, Chile)

It is well known that sound waves are often difficult, if not impossible, to visualize which makes their nature and effects much more difficult to explain than those of other kinds of waves. In this article, a visualization tool for enhancing the students' learning process for a fundamental of acoustics course is reported. The visualization is done through particle displacement computer animations of different sound propagation cases using a simple MATLAB script. Several examples that can enhance the material discussed during class time are presented, in particular those topics involving diffraction of sound waves. It is observed that visual displays used during lectures improve the students' retention of new material. Students seem to make better association between wave motion and particles in a medium. A visual advantage of the particle displacement animations is that they use tangible items to represent the invisible process, by visualizing the invisible particles of air as dots on the computer screen. In addition, this instructional tool used to visualize sound fields enhances the understanding of many conceptual aspects underlying sound wave motion and can be used to motivate a discussion of the wave equation later.

2:00

2pEDa4. A student-friendly algorithm for planar mode propagation in arbitrary transmission lines. Jerry H. Ginsberg (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 5661 Woodsong Dr., Dunwoody, GA 30338, jerry.ginsberg@me.gatech.edu)

Many textbooks treat the propagation of a harmonic plane wave in an acoustic transmission line. Their scope generally is limited to a small number of interconnected uniform cross section branches, and they emphasize fitting the propagation properties to continuity conditions at junctions. In contrast, the network formalisms advocated for intricate engineered systems, such as that offered fairly

recently by Panigrahi and Munjal [J. Acoust. Soc. Am. **118**, pp. 2860–2868], make them unsuitable for a first graduate-level course in acoustics. The present work offers an algorithmic approach that is simple to formulate, yet more efficient than available alternatives, and capable of treating arbitrary networks. The steps required to implement the algorithm are sequential numbering of the branch nodes and of the junctions, definition of a connectivity matrix that indicates which nodes are common to a junction, and statement of the continuity and termination conditions in terms of the port pressures and particle velocities. Beyond that the algorithm operates automatically. The result is a set of simultaneous equations for the complex pressure amplitudes at the junctions. In contrast, the approach offered by Panigrahi and Munjal derives simultaneous equations for the nodal pressures and port velocities, whose number is far greater than the number of junctions.

Contributed Papers

2:20

2pEDa5. Acousto-mechanical modeling of an Edison tinfoil phonograph. Jason D. Sagers, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758), and Preston S. Wilson (The Univ. of Texas at Austin, Austin, TX 78712-0292)

A homemade reproduction of an Edison tinfoil phonograph was demonstrated at the 161st ASA meeting in Seattle, WA [JASA **129**, 2581 (2011)]. While past work focused on the history and development of the device, the present work is focused on analyzing the acousto mechanical behavior of the homemade device. A dynamical model is presented and is used to predict the frequency dependent vibration of the phonograph diaphragm due to an acoustic input. The model predictions are compared with experimental measurements of the diaphragm vibration and potential solutions for optimizing the device are discussed.

2:35

2pEDa6. Demonstration of coupled membrane modes on a musical drum. Randy Worland (Dept. of Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Musical drums such as tom toms and snare drums typically consist of two circular membranes attached to a cylindrical shell. Due to the enclosed air and the shell itself, a drum with two heads exhibits coupling of the lower frequency membrane modes, while the higher frequency modes of the two heads remain essentially independent. A simple demonstration has been developed that illustrates several aspects of drumhead vibrations, including the distinction between strongly and weakly coupled modes. Methods for determining the relative phases and amplitudes of the coupled oscillations have also been developed. The demonstration is useful at a variety of pedagogical levels and can be supplemented with more advanced experiments, including the use of electronic speckle pattern interferometry to identify the mode shapes on both membranes.

TUESDAY AFTERNOON, 1 NOVEMBER 2011 TOWNE/ESQUIRE, 2:55 TO 4:00 P.M.

Session 2pEDb

Education in Acoustics: Take 5's

Andrew Morrrison, Chair
Physics Dept., DePaul Univ., 2219 Kenmore Dr., Byrne Hall, Chicago, IL 60614

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

Session 2pMUa

Musical Acoustics: Music Perception and Cognition

Diana Deutsch, Chair
Dept. of Psychology, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

2:00

2pMUa1. Why does musical training benefit the neural encoding of speech? A new hypothesis. Aniruddh D. Patel (The Neurosci. Inst., 10640 John Jay Hopkins Dr., San Diego, CA 92121)

Mounting evidence suggests that musical training benefits the neural encoding of speech. This paper offers a hypothesis specifying why such benefits occur. The OPERA hypothesis proposes that such benefits are driven by adaptive plasticity in speech-processing networks, and that this plasticity occurs when five conditions are met. These are (1) **O**verlap: there is anatomical overlap in brain networks that process an acoustic feature used in both music and speech (e.g., waveform periodicity, amplitude envelope), (2) **P**recision: music places higher demands on these shared networks than does speech, in terms of the precision of processing, (3) **E**motion, (4) **R**epetition, and (5) **A**ttention: the musical activities that engage this network elicit strong positive emotion, are frequently repeated, and are associated with focused attention. According to the "opera" hypothesis, when these conditions are met neural plasticity drives the networks in question to function with higher precision than needed for ordinary speech communication. Yet since speech shares these networks with music, speech processing benefits. The OPERA hypothesis is used to account for the observed superior subcortical encoding of speech in musically trained individuals and to suggest mechanisms by which musical training might improve linguistic reading abilities.

2:15

2pMUa2. The octave illusion revisited: Performance measurements for handedness categorization. Michael Oehler (Musicological Inst., Univ. of Cologne, Cologne, 50674 Germany), Christoph Reuter, Harald Schandara, and Michael Kecht (Univ. of Vienna, Vienna, Austria)

An extended replication study of the octave illusion (Deutsch 1974, 1983) is presented. Since the first description of the octave illusion in 1974, several studies showed that the perception of the two-tone patterns depends on subjects' handedness. Partially almost 90% of the right-handed subjects reported to hear the high tone of the octave at the right ear. In all related studies the handedness categorization was done by means of a questionnaire, e.g., the handedness inventory of Varney and Benton (1975). Several current studies (e.g., Kopiez, Galley, Lehmann, 2010), however, showed that objective non-right-handed persons cannot be identified by handedness inventories. In concordance with Annett's "right shift theory" (2002), performance measurements as speed tapping seem to be a much more reliable handedness predictor. Therefore in the replication study (N=158) Varney and Benton's inventory as well as a speed tapping task were used to categorize left- and right-handed subjects. The results of Deutsch's study could be replicated when using the same handedness inventory. The performance measurement task, however, led to a significantly clearer distinction between the left- and right-handed subjects (w=0.39 in contrast to w=0.26 in the replication) and more structured perception patterns could be observed within the left-handed group.

2:30

2pMUa3. Large-scale direct-test study reveals unexpected characteristics of absolute pitch. Diana Deutsch (Dept. of Psychol., Univ. of California, San Diego, La Jolla, CA 92093), Jinghong Le (East China Normal Univ., Shanghai 200062, China), Jing Shen (Univ. of California, San Diego, La Jolla, CA 92093), and Xiaonuo Li (Shanghai Conservatory of Music, 20 Feng Yang Rd., Shanghai 200031, China)

Absolute pitch, the ability to name a musical note in the absence of a reference note, is very rare in North America and Europe, so that attempts to characterize its features in the western world have involved small numbers of subjects, informal self-report, questionnaires, or web-based exploration. The study reported here capitalized on the high prevalence of absolute pitch in China to explore its features in detail using direct, on-site testing of 160 subjects in a Chinese music conservatory. As expected, performance levels were extremely high, and there was a large effect of age of onset of musical training, with those who began training by age 5 scoring on average 83% correct not allowing for semitone errors and 90% correct allowing for semitone errors. It was found that errors tended to be on the sharp side. An advantage to white keys over black keys was also found; however this was not due to early experience with the piano, as had been hypothesized by others, since performers on different instruments showed an effect that was as large or larger. Furthermore, the special status for note A that had been hypothesized by others was not found, even for orchestral performers.

2:45

2pMUa4. Songs, cell phones, absolute pitch: Long-term pitch memory for familiar stimuli. Kevin Dooley (Dept. of Psychology, Univ. of California, San Diego, La Jolla, CA 92093)

Absolute pitch (AP) is a rare phenomenon as formally defined, but long-term pitch memory appears much more common when tests involve familiar musical material and do not require the use of formally learned pitch labels. It is unclear whether AP possession confers additional advantages to long-term pitch memory in such tasks or merely combines a rare ability to form pitch-label associations with a more general capacity for pitch memory. To test this, 36 trained musicians—18 AP possessors and 18 non-possessors with equivalent age of onset and duration of musical training—were asked to recall and vocalize a familiar song, and their responses were compared with the pitches of the actual recordings; this was repeated with their cell phone ringtones. Both groups were significantly more accurate than chance on the song task, but only the AP possessors performed above chance on the ringtone task. The findings confirm the existence of widespread long-term pitch memory but also point to an AP advantage under some circumstances.

Session 2pMUb

Musical Acoustics: Music and Auditory Space

Diana Deutsch, Chair
Dept. of Psychology, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

3:30

2pMUb1. The Instrument & the Room: A study of the grand piano focused on the needs of audio education. Brett Leonard, Grzegorz Sikora, and Martha de Francisco (Graduate Program in Sound Recording, Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., 555 Sherbrooke St., West Montreal, QC H3A 1E3, brett.leonard@mcgill.ca)

Through their training and education, aspiring recording engineers often encounter literature on the radiation patterns of typical musical instruments. The study of this information can greatly inform the placement of microphones and facilitate one's learning about the way acoustic instruments work. These musical acoustics studies, however, typically employ an anechoic or near-anechoic environment to minimize reflections from interfering with the instrument under test. Since the recording engineer works almost exclusively in environments with reflective surfaces, this causes a disconnect and can inhibit a full understanding of the relationship between the instrument and its environment. A case study of the acoustic grand piano is presented in which the instrument and the non-anechoic room are presented as a single, coupled acoustic system. Over 1300 measurement points are used to characterize the instrument/room combination. The study is conducted in both a small recording space and a large scoring stage, yielding non-room specific results that show areas of high frequency energy that are not present in typical anechoic measurements. Exploration of these differences and potential causes are presented.

3:45

2pMUb2. Methods for automating multichannel directivity measurements of musical instruments in an anechoic chamber. Nicholas J. Eyring II, Timothy W. Leishman, Kristina M. Sorensen, and Nathan G. W. Eyring (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, eyringj@gmail.com)

Comprehensive directivity measurements of musical instruments in anechoic environments involve several experimental challenges. However, by adapting methods currently used to characterize loudspeaker directivity (e.g., through high-resolution balloon plots), one can obtain highly detailed and instructive directivity data. This may be accomplished by rotating a musical instrument with sequential azimuthal angle increments under a fixed semicircular array of microphones while recording repeated notes or sequences of notes. The result is a computer-controlled acquisition of hundreds or even thousands of sound pressure measurements over a measurement sphere. The directivity data and corresponding balloon plots may be shown to vary as functions of time or frequency. This paper explores the approach applied to a grand piano with velocity controlled keys played through MIDI communication. Instruments played by live musicians may also be evaluated, although the process requires carefully developed techniques of control, feedback, and compensation to achieve acceptable results. These and other considerations of performing automated, multichannel directivity measurements of musical instruments are detailed in this presentation.

4:00

2pMUb3. Feature space minimization and its affect on head related transfer functions clustering. Areti Andreopoulou and Agnieszka Roginska (Dept. of Music and Perf. Arts Professions, New York Univ., 35 W. 4th St., Ste. 777, New York, NY 10012, aa1510@nyu.edu)

Several approaches have been taken toward data/feature-space reduction in HRIRs, operating either in the time (original, minimum phase, normalized HRIRs) or in the frequency domain (magnitude, log-magnitude, standardized log magnitude HRTFs). Shin and Park (2008) extracted only the response of the pinna (0.23 ms of the original HRIR), Hwang and Park (2007) included also the response of the head and torso (1.5 ms), while Bondu (2006) operated on the first 100 samples of the impulse responses. Other research focuses on employing PCA for minimizing the feature space to 5–12 orthogonal components and their corresponding weights (Langendijk, 2002; Huang, 2009; Hugeng, 2010), while others have managed to isolate directional and non-directional spectral cues of non-individualized HRTFs (Hu, 2008; Diepold, 2010). In previous work, the clustering tendencies of the standardized log-magnitude HRTFs of 110 subjects on the horizontal plane have been demonstrated, by applying k-means on 256-feature filters. In this study, those results are compared to the clustering behavior of data reduced filters by applying the previously mentioned techniques. The extent to which the original clustering tendencies are maintained is used as an evaluation criterion of the appropriateness of each data-reduction technique.

Session 2pNS

Noise: General Topics–Outdoor Noise

Dickson Hingson, Chair
275 S. River Run, #3, Flagstaff, AZ 86001

Chair's Introduction—1:00

Contributed Papers

1:05

2pNS1. Effects of the soil on the noise attenuation of environmental berm barriers. Jorge P. Arenas (Inst. of Acoust., Univ. Austral of Chile, P.O. Box 567, Valdivia, Chile), Jesus Alba, and Romina del Rey (Universidad Politécnica de Valencia, Campus de Gandía, 46730 Valencia, Spain)

Berm mounds are a commonly used technique to reduce the environmental noise levels produced by highways. A berm is a natural noise barrier constructed of soil, stone, rock, or rubble, often landscaped, running along a highway to protect adjacent communities from noise pollution. An earth mound may be constructed using surplus materials at project site, provided there is sufficient land area available for its construction. Therefore, berms are natural environmental barriers, having relative low costs and they are subjectively well perceived by residents. However, exact noise attenuation provided by berms has not been enough explored in the technical literature, as opposed to common barriers made of vertical rigid walls. Although, some highway noise prediction models assign a noise reduction bonus of 3 dB(A) to sound barriers made of earth mounds, experimental assessments have yielded mixed results. Few theoretical reports have studied this particular problem. In this work, numerical analysis using the classical theory of diffraction is performed on a berm made of different types of soil. The model assumes a line source and includes flow resistant data as boundary conditions. By integrating the results, noise attenuation is given in third-octave bands. It is concluded that soil's properties significantly influence the measured results and that this may be one of the causes of varied in-site empirical evidence.

1:20

2pNS2. Grand Canyon National Park Overflights environmental impact statement (EIS): Backcountry impairment under the National Park Services' (NPS) noise standards. Dickson J Hingson (Natl. Parks and Monuments Comm., 275 S River Run, Flagstaff, AZ 86001, dhingson@infowest.com)

The longstanding Grand Canyon overflights noise pollution saga approaches a decision—public comment on the DEIS having concluded in June. Current, longstanding park impairment of soundscape and wilderness character, throughout the park's popular East end backcountry, is readily apparent from detailed quantitative "Location Point" analysis. Per cent time audible and sound level for these, by alternative, and season was displayed with an elegant technique, presented 2009 to INCE/ASA, by Nick Miller. Although the "NPS preferred" draft alternative did not represent significant improvement, the quieter "seasonal use," Alternative "E" fared much better. This derives from seasonal closures, each year, alternating between the two currently used air tour loops. "E's" data indicate that more stringent daily limits on tour flight allocations will be required to avoid major adverse noise impacts continuance in park backcountry, even so. The core business of the park service being to prevent impairment in its wilderness backcountry, the stark findings afford the NPS a clear opportunity to adapt, with a more appropriate implementation alternative. This will be due by spring, 2012, to timely render a record of decision. An recent agreement between NPS and FAA as to relative roles was helpful in clarifying the applicable noise standards.

1:35

2pNS3. Propagation in a realistic outdoor environment. Whitney L. Coyle, Victor W. Sparrow (Graduate Program in Acoust., Penn State Univ., University Park, PA 16802, wlc5061@psu.edu), Bruce Ikelheimer, Micah Downing, Michael James, Kevin Bradley, and Josh Mellon (Blue Ridge Res. and Consulting, 13 1/2 W. Walnut St., Asheville, NC 28801)

A complementary experimental and computational study was conducted to assess variability in realistic outdoor sound propagation environments. Field measurements were conducted in a valley of the Smokey Mountains located in western North Carolina. This location exhibited complex terrain, vegetation, and weather conditions. Continuous and impulsive sources were positioned in multiple, reciprocal locations for the experiment. Receiver locations were spread throughout the area with many significant terrain features. Simultaneous atmospheric measurements were made to measure wind speed and temperature profiles and a level of turbulence. A Green's function parabolic equation model was written and used with matching conditions for comparison. This presentation will give a brief overview of the project and provide preliminary results. [Work supported by Spawar Systems Center Pacific.]

1:50

2pNS4. Model-data comparison for acoustic propagation over water. Sean P. Pecknold, Cristina Tollefsen, and Emma Murowinski (Defence R&D Canada Atlantic, P.O. Box 1012, Dartmouth, Nora Scotia, B2Y 3Z7, Canada, sean.pecknold@drdc-rddc.gc.ca)

Modeling the propagation of sound over water is an important tool for determining the possible effects of noise sources on the environment, such as naval gunfire exercises on bird nesting grounds, or offshore wind turbine disturbance. Temperature, humidity, and wind speed and direction all play an important role in determining acoustic propagation over water. Here, a sound source is mounted on a boat moving up to 2.5 km toward and away from a receiver on another vessel, over the span of several days. The propagation loss measured as a function of range is compared to modeled results based on measured temperature, humidity, wind velocity, and surface roughness, using atmospheric turbulence models to improve prediction capability.

2:05

2pNS5. Application of the equivalent source method to outdoor directional sound sources. Sergey N. Vecherin, D. Keith Wilson (U. S. Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755, Sergey.N.Vecherin@usace.army.mil), and Vladimir E. Ostashev (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 216 UCB, Boulder, CO 80309)

Many outdoor sound sources, such as aircraft and moving ground vehicles, exhibit directional sound radiation patterns that can be

measured in the far field. However, this information is insufficient for specifying a source function that can be used in propagation algorithms. The equivalent source method (ESM) allows one to reconstruct an equivalent distribution of point sources having a given far-field radiation pattern. In this research, the application of the ESM to spatially complex radiation patterns, similar to those of actual helicopters, is studied in detail. Two algorithms for the source reconstruction are developed for arbitrarily complex radiation patterns. The first algorithm reconstructs three-dimensional source distributions that may not, however, be convenient for initializing calculations with parabolic equations. The second algorithm is designed for the two-dimensional parabolic equations and reconstructs strictly vertical source distributions having a given radiation pattern over a limited range of elevation angles. Some practical aspects of the measured data, such as outliers, data incompleteness, and phase loss in sound level measurements are also studied and recommendations are provided for mitigating their adverse effects on source reconstruction.

2:20

2pNS6. Spatio-temporal characteristics of countryside soundscapes in Hong Kong. Kin-che Lam (Dept. of Geography & Resource Management, Chinese Univ. of Hong Kong, Shatin, NT., Hong Kong, kinchelam@cuhk.edu.hk)

Over six hundred 15-min and one hundred 24-h digital sound recordings were undertaken in different parts of the Hong Kong countryside covering different landscape units, different time of day and seasons. Spectrograms were prepared from these recordings and 24-h spectrograms were subdivided into segments with relatively uniform acoustic characteristics. These segments were then analyzed for the sound sources, acoustic and psychoacoustic characteristics. These data were analyzed to determine for their spatial and temporal characteristics. Cluster and discriminant analyses were further undertaken to ascertain how acoustic and sound sources of the countryside vary over space and time. The results indicate that soundscape characteristics are determined primarily by the type of landscape, and then by the diurnal and seasonal characteristics. The implications of these findings on human experience, sampling strategy, and countryside management and planning will be discussed.

TUESDAY AFTERNOON, 1 NOVEMBER 2011 ROYAL PALM 3/4, 1:00 TO 5:45 P.M.

2p TUE. PM

Session 2pPA

Physical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration: Hypersonic Spectroscopy of Microstructured Materials

Kenneth G. Foote, Cochair
Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543

Paul A. Snow, Cochair
Dept. of Physics, Univ. of Bath, Claverton Down, The Avenue, Bath, BA2 7AY, UK

Chair's Introduction—1:00

Invited Papers

1:05

2pPA1. Hypersonic spectroscopy of porous silicon for acoustic devices. Paul A. Snow, Leigh-Anne Thomas, Bernhard Goller, and Gazi N. Aliev (Dept. of Phys., Univ. of Bath, The Ave., Bath, BA2 7AY, UK)

I will review our work on porous silicon (pSi) presenting achievements while highlighting underlying physical questions that remain to be answered. pSi is produced by the electrochemical etching of crystalline silicon. It is typically mesoporous, having pores of 10–30 nm diameter. The etching current density determines the final porosity, the volume fraction of air, with a wide range of porosities, 25%–95%, achievable. For wavelengths much greater than the pore size, pSi gives a tunable effective medium for light and sound waves. We have characterized pSi acoustic properties using transmission spectroscopy with matched transducer pairs working at 0.5–2.5 GHz. The results for velocity, v, have fitted to a general law of $v=v_0(1-\phi)^k$, where v_0 is the velocity in bulk silicon, ϕ is porosity, and k is the fitting parameter. We have investigated the variation of k with the direction of propagation and the etching conditions used to extract the dependence of the elastic constants on porosity. The measurement of velocity has enabled us to produce and characterize pSi Bragg mirrors and rugate filters that have a smoothly varying acoustic impedance. This has demonstrated the potential use of pSi in acousto-optic phoxonic crystal devices that have both phononic and photonic bandgaps.

1:30

2pPA2. Picosecond ultrasonic microscopy of semiconductor nanostructures. Thomas J. Grimsley, Shan Che (Dept. of Phys., Brown Univ., Providence, RI 02912), G. Andrew Antonelli (Novellus Systems, Albany, NY 12203), Arto V. Nurmikko, and Humphrey J. Maris (Brown Univ., Providence, RI 02912)

We describe a new picosecond ultrasonics method for the study of nanostructures. A sound pulse is generated when an ultra-short laser pulse is absorbed in a transducer structure. The sound then propagates across a thin layer of water and is reflected from the surface of the sample being examined. A resonant optical cavity is used to improve the efficiency of optoacoustic detection and generation of the sound. We report on experiments in which sound is reflected from patterned nanostructures. In these experiments, we are able to study the propagation of sound down channels of width as small as 35 nm.

1:55

2pPA3. Probing acoustical, optical, and acousto-optical properties of nanostructured materials by picosecond laser ultrasonics. V. Gusev and P. Ruello (LPEC, UMR-CNRS 6087, Universit du Maine, av. O. Messiaen, 72085 Le Mans, France)

Research results on the characterization of the thin (submicrometers thick) films of the nanostructured materials by the experimental methods of picosecond laser ultrasonics are reviewed. These methods make use of femtosecond lasers to generate and to detect GHz–THz acoustic waves. In this communication, theoretical backgrounds of the fs-laser-based opto-acousto-optic techniques that are used for the evaluation of the material properties are first introduced. Then, the results of the experiments on nanoporous low-k films (for the microelectronics), on nanogranular sol–gel optical coatings (for laser optics), on anodized alumina (for the nanomaterial/nanostructure templates), on synthetic opals, on nanoparticles supra-crystals, and other nanostructured materials are discussed. The emerging opto-acousto-optic technology for the depth-profiling of acoustical, optical, and acousto-optical properties of inhomogeneous transparent films with the nanometers scale spatial resolution is also presented.

2:20

2pPA4. Using light to probe hypersound in porous materials systems. Lance C. Parsons, Jordan Peckham, Anna M. Polomska, and G. Todd Andrews (Dept. of Phys. and Physical Oceanogr., Memorial Univ. of New Foundland, St. John's, NL, A1B 3X7, Canada, tandrews@mun.ca)

An overview of the technique of Brillouin light scattering spectroscopy and its application to the study of hypersound in micro- and mesoporous materials systems will be presented. Particular emphasis will be placed on results obtained from light scattering experiments on porous silicon-based structures. For porous silicon films, it was found that the acoustic phonon velocities and elastic properties depend strongly on the film porosity and morphology. Brillouin studies of porous silicon superlattices with periodicity on the order of the hypersound wavelength reveal that these structures behave as hypersonic phononic crystals, while those with smaller modulation wavelengths act as effective elastic media. New results on localized acoustic modes in porous silicon multilayers will also be discussed. Collectively, these studies provide a detailed picture of hypersound propagation in porous silicon systems and demonstrate the utility of Brillouin spectroscopy for probing acoustic phonon behavior in this challenging class of materials. [This work was supported by the Canada Foundation for Innovation, Memorial University of Newfoundland, and the Natural Sciences and Engineering Research Council of Canada.]

2:45

2pPA5. High frequency soft phononics. George Fytas (Max Planck Inst. for Polymer Res., Ackermannweg 10, 55128 Mainz, Germany, fytas@iesl.forth.gr)

Phononic crystals, the acoustic equivalents of the photonic crystals, are controlled by a larger number of material parameters. The study of hypersonic crystals imposes substantial demand on fabrication and characterization techniques. Colloid and polymer science offer methods to create novel materials that possess periodic variations of density and elastic properties at mesoscopic length scales commensurate with the wave length of hypersonic phonons and hence photons of the visible light. Polymer- and colloid-based phononics is an emerging new field at the interface of soft materials science and condensed matter physics with rich perspectives ahead. The key quantity is the dispersion of high frequency (GHz) acoustic excitations which is nowadays at best measured by high resolution spontaneous Brillouin light scattering. Depending on the components of the nanostructured composite materials, the resolved vibration eigenmodes of the individual particles sensitively depend on the particle architecture and their thermo-mechanical properties [T. Still *et al.*, Nano Lett. **10**, 3194 (2008)]. In periodic structures of polymer based colloids, the dispersion relation $\omega(k)$ between the frequency and the phonon wave vector k has revealed hypersonic phononic band gaps of different nature [T. Still *et al.*, Phys. Rev. Lett. **106**, 175505 (2011)].

3:10–3:25 Break

3:25

2pPA6. Engineering the band structure of one-dimensional hypersonic phononic crystals. Dirk Schneider (Max Planck Inst. for Polymer Res., Ackermannweg 10, 55128 Mainz, Germany, schneider@mpip-mainz.mpg.de), El Houssaine El Boudouti (Universitè Mohamed I, 60000 Oujda, Morocco), Faroha Liaqat, Wolfgang Tremel (Johannes Gutenberg Univ., 55128 Mainz, Germany), Hans-Jürgen Butt (Max Planck Inst. for Polymer Res., 55128 Mainz, Germany), Bahram Djafari-Rouhani (Universitè de Lille 1, 59655 Villeneuve d'Ascq, France), and George Fytas (Univ. of Crete and FORTH, 71110 Heraklion, Greece)

Phononic crystals—the mechanical analogues of photonic crystals—have attracted increasing interest and have been widely studied within the past decade. The phononic dispersion relation at hypersonic frequencies can be directly measured by the powerful non-destructive technique of high resolution spontaneous Brillouin-light-scattering (BLS) [W. Cheng *et al.*, Nature Mater. **2006**, 5, 830]. Due to the vector nature of the elastic wave propagation, theoretical phononic band structures can be uniquely verified at low dimensionality, and hence 1D phononic crystals constitute model systems for fundamental studies. Such hybrid Bragg stacks, composed of alternating layers of silica and poly(methyl methacrylate) (PMMA), respectively, exhibit clear hypersonic phononic band gaps [Gomopoulos *et al.*, Nano Lett. **2010**, 10, 980]. In this paper, we report on the fabrication, characterization, and both experimental and theoretical dispersion diagrams along and normal to the periodicity direction of silica/PMMA multilayers. The width of the gap, the phonon frequencies, and their intensities near the first Brillouin zone are sensitive probes of the longitudinal moduli and elasto-optic constants of the individual layers and structural parameters. Mixing with layer modes under oblique incidence conditions allows access to the shear moduli of the two layers.

Contributed Papers

3:50

2pPA7. A holey structured acoustic metamaterial. J. Zhu (Nanoscale Sci. and Eng. Ctr. (SINAM), 3112 Etcheverry Hall, Univ. of California, Berkeley, CA 94720), J. Christensen, J. Jung (Universidad Autonoma de Madrid, E-28049 Madrid, Spain), L. Martin-Moreno (CSIC-Universidad de Zaragoza, E-50009 Zaragoza, Spain), X. Yin, L. Fok, X. Zhang (Univ. of California, CA 94720), and F. J. Garcia-Vidal (Universidad Autonoma de Madrid, E-28049 Madrid, Spain)

The resolution of acoustic imaging system is restricted by diffraction limit. To beat this limit, early research shows acoustic metamaterials that can manipulate acoustic waves artificially and may act as lens to achieve subwavelength resolution. However, these solutions suffer significant loss therefore lack convincing experimental demonstration. Recent study suggested that arrays of metallic nanorods or nanowire can be used as lens for optical imaging at subwavelength resolution. Similar acoustic hyperlens designs have also been explored, and latest experimental result provided resolution of wavelength/7. Here presented is a holey structured endoscope which supports the transmission of the otherwise-evanescent waves over distances, therefore beating diffraction limit and achieving deep subwavelength imaging. Experimental demonstration shows clear image with feature size of wavelength/50. Such a metamaterial endoscope brings new perspectives to the applications of medical ultrasonography, sonar and ultrasonic non-destructive evaluation.

4:05

2pPA8. Mechanisms of nonlinear saturation in focused acoustic beams of periodic waves and single pulses. Vera Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), Maria Karzova, Mikhail Averiyanov (Moscow State Univ., Moscow 119991, Russia), and Oleg Sapozhnikov (Univ. of Washington, WA 98105)

Physical mechanisms leading to saturation of various acoustic field parameters in nonlinear focused beams of periodic waves and single pulses were investigated numerically. A numerical algorithm based on the KZK equation was used in the simulations. Propagation of an initially harmonic wave and a single pulse (one period of a sine wave) emitted by a focused transducer with Gaussian apodization was modeled. It was shown that in periodic fields, saturation of the peak positive pressure is mainly due to the effect of nonlinear absorption at the shock front. In acoustic fields of single pulses the main mechanism of saturation is the nonlinear refraction. Maximum pressure in the periodic field, achieved at the focus, was found to be higher than that of the single pulse. The total energy of the beam of the periodic wave, however, decreases much faster with the distance from the source as compared to the single pulse focusing. These nonlinear propagation effects propose a possibility to use pulsed beams for more effective delivery of the wave energy to the focal region, and periodic waves—to achieve higher pressure values of at the focus. [Work supported by EB007643, NIH DK43881, DK075090, and RFBR 09-02-01530.]

4:20

2pPA9. A study of the nonlinear effects of air bubbles on the ultrasonic field in water. Christian Vanhille (Universidad Rey Juan Carlos, Tulipan s/n, 28933 Mostoles, Madrid, Spain, christian.vanhille@urjc.es) and Cleofé Campos-Pozuelo (Consejo Superior de Investigaciones Cientificas, 28006 Madrid, Spain)

We consider the propagation of ultrasonic waves in water with air bubbles. On the one hand, a numerical model has been developed to analyze the nonlinear effects of the bubbles at high amplitude, in several configurations (open-field, standing waves, 1–D, 2–D, 3–D, homogeneous bubble density, bubble layers, bubble clouds, bubble generation). On the other hand an experimental setup has been constructed and allows us to study the nonlinear behavior of the inertial cavitation field at high frequency and at high power. In particular, a bubble cloud is formed at a large distance from the sonotrode. [Work is part of the research project DPI2008-01429 funded by the Spanish Ministry of Science and Innovation.]

4:35

2pPA10. Dynamics of bubble clusters in acoustic field. I. S. Akhatov (Dept. of Mech. Eng., North Dakota State Univ., Fargo, ND 58108), E. S Nasibullaeva, Y. V. Volkova (Ctr. for Micro and Nanoscale Dynam. of Dispersed Systems, Bashkir State Univ., Ufa 450074, Russia), and N. A. Gumerov (UMIACS, Univ. of Maryland, College Park, MD 20742)

A bubble cluster composed of gas bubbles of various radii oscillating in an unbounded, slightly compressible viscous liquid under the action of an external acoustic field is considered. The mathematical model describing the dynamics of this bubble cluster is presented. The proposed model is used for an analytical study of small (linear) bubble oscillations in monodisperse and polydisperse clusters, for a numerical investigation of large (nonlinear) bubble oscillations, and for a diffusion stability analysis of gas bubbles in the cluster. The following phenomena have been revealed: (1) synchronization of the collapse phases of bubbles with different radii and (2) collapse intensification for bubbles of one size in the presence of bubbles of other size. These effects are explained by the interaction between bubbles of different radii in the cluster. For a monodisperse (one-fraction) cluster, numerical values were obtained for the initial gas concentrations in the liquid at which bubbles tend to one of two equilibrium states due to the rectified diffusion. It is also found that a polydisperse (two fraction) cluster tends to become a one fraction cluster due to the rectified diffusion. [This research is supported by the Grant of the Ministry of Education and Science of the Russian Federation (G34.31.0040).]

4:50

2pPA11. Soil plate oscillator: Modeling nonlinear mesoscopic elastic behavior and hysteresis in acoustic landmine detection, Part II. Dang V. Duong (Weapons and Systems Eng. Dept., U.S. Naval Acad., Annapolis, MD 21402) and Murray S. Korman (U.S. Naval Acad., Annapolis, MD 21402)

An apparatus (SPO), designed to study flexural vibrations of a soil loaded plate, consists of a thin circular elastic clamped plate (and cylindrical wall) supporting a vertical soil column. A small magnet attached to the center of the plate is driven by a rigid AC coil (located coaxially below the plate) to complete the electrodynamic soil plate oscillator SPO design. The mechanical impedance Z_{mech} (force/particle velocity, at the plate's center) versus frequency is inversely proportional to the electrical motional impedance Z_{mot}. Measurements of Z_{mot} are made using the complex output to input response of a Wheatstone bridge that has an identical coil element in one of its legs. Resonant oscillation measurements (with no soil) before and after a slight point mass loading at the center help determine effective mass, spring, damping and coupling constant parameters of the system. "Tuning curve" behavior of real Z_{mot} and imaginary Z_{mot} at successively higher vibration amplitudes exhibit a decrease "softening" in the resonance and an increase in the quality Q factor. A bilinear hysteresis model [T. K. Caughey, ASME, J. Applied Mech. Trans. 640 (1960)] predicts the tuning curve shape for this nonlinear mesoscopic elastic SPO behavior.

5:05–5:45 Panel Discussion

Session 2pSA

Structural Acoustics and Vibration: Session in Honor of Gideon Maidanik

Richard H. Lyon, Cochair
RH Lyon Corp, 60 Prentiss Ln., Belmont, MA 02478

Joseph W. Dickey, Cochair
3960 Birdsville Rd., Davidsonville, MD 21035

Chair's Introduction—1:15

Invited Papers

1:20 Open microphone—Reminiscences

1:40

2pSA1. Radiation efficiency, impedance, and the acoustics of rattle. Philip Shorter, Vincent Cotoni (ESI Group, 12555 High Bluff Dr., Ste. 250, San Diego, CA 92130), and Robin Langley (Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, United Kingdom)

Rattle issues consistently rank as one of the top consumer complaints in initial quality surveys for many new products. Predicting the acoustics of rattle is complicated by the need to model the vibro-acoustic response of large complex structures across a broad frequency range. The complexity of the analysis can be reduced by making use of standard methods derived almost 50 years ago. In particular, this paper discusses a computationally efficient method for assessing the propensity for rattle in large complex structures. A finite element model is used to predict the probability that impacts will occur when a product is exposed to a particular low frequency random vibro-acoustic environment. The expected contact forces arising from each impact are then estimated by making use of expressions involving the drive point impedances of infinite structures. Finally, the vibration and acoustic radiation associated with the various impacts are predicted and ranked using a SEA model. A number of examples are presented including one which makes use of a radiation efficiency formula due to G. Maidanik. The results are in very good agreement with analytical reference results.

2:00

2pSA2. Controlling the response of an oscillator using a coupled set of satellite oscillators. Ronald G. Hughes (NSWCCD 9500 MacArthur Blvd., West Bethesda, MD 20877) and Gideon Maidanik (Deceased)

The response of a system comprised of multiple dynamic systems is analyzed. The results shown here are for the main or master oscillator in that system. The balance of the dynamic systems is designated as satellite oscillators. Controlling the response of the master oscillator is described in terms of the couplings to the master oscillator, the frequency distribution of the satellite oscillators, the loss factors, and the masses of those oscillators. The frequency distributions and masses of the satellite oscillators are specified via normalizations with respect to the resonance frequency and mass of the master oscillator in order to generalize the approach. It is shown that contrary to reported results by others, there is no requirement to optimize the frequency distribution of the satellite oscillators to maximize the control of the response of the master oscillator. Further it is shown that increasing the loss factor of the satellite oscillators beyond certain values does not bring further benefit in controlling the response of the master oscillator beyond a certain level, in fact, a saturation is reached. We describe the on-set of saturation in terms of the modal overlap parameter.

2:20

2pSA3. Shaping the response of multi-degree-of-freedom mechanical systems. Vyacheslav M. Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA 92606, vryaboy@newport.com)

The paper pays homage to the outstanding contribution by Gideon Maidanik to studying synergic action of multiple add-on oscillators in reducing resonance response of a main structure. Methods and results of constructing multi-degree-of-freedom mechanical systems with required responses in frequency and time domains are presented with application to optimal vibration isolation, impedance matching and shock absorption. It is shown, in particular, that the limiting quality of instantaneous shock isolation, which was usually attributed to highly non-linear, heavily damped or active systems, can be achieved by linear low-damped multi-degree-of-freedom mechanical systems. Such systems can be synthesized as multiple vibration absorbers, stacked oscillators (chains), or more general structures including motion transformation elements.

2:40

2pSA4. Fuzzy structures applied to a vibrating beam. David Feit (Acoust. Society of America, Ste. 1NO1, 2 Huntington Quadrangle, Melville, NY 11757-4502)

Gideon and I were both introduced to the ideas of "Fuzzy Structure Theory" during a visit with Christian Soize while involved in an exchange meeting between the U.S. Navy and the French Navy in 1986. In the years just prior to his passing, he together with Ron Hughes pursued this subject, and that work is also presented in this session. My presentation, a continuation of work that I had originally done with Murray Strasberg, discusses the transient response of a multiple set of fuzzy structures attached to a master structure, either a longitudinally or flexurally vibrating beam, that itself has multiple resonances. The transient response of the subordinate oscillators in certain parameter ranges gives rise to distinct packets of energy traveling with different wave speeds. This phenomenon has not been previously observed, and a tentative explanation is offered.

3:00–3:15 Break

3:15

2pSA5. Thoughts on submarine structural acoustics. Ira Dyer (Dept. of Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA 02139)

The simplest submarine model is a uniform circularly cylindrical shell with flat end caps. This is too simple to be useful, but the recent phenomenal advances in computational structural acoustics, and the availability of test and laboratory facilities, make it feasible to tackle greater complexity in the submarine structure. But the newer numerical computational analyses, and standard experimental approaches, have as their initial outputs large data sets. So, in effect, structural acousticians have traded simpler and less realistic structural models, for more dense data sets and more useful models, a trade bound to be highly positive. In this context, the author has observed, subjectively, that those who have the ability to aggregate their large data sets with physics-centric descriptions do present their results crisply and with deep understanding. Also, they are better prepared to adjudicate alternate interpretations and to suggest further steps to reach more unique conclusions. Accordingly, the author suggests that specialized data filters be researched and developed for use in data interpretation, these to be formulated from questions such as: Where in the submarine structure, and in what wavenumber band, does wavenumber matching needs to be controlled? What is the modal character of an internal stiffening ring, and where should absorptive sinks of resonance peaks be placed? Additional questions are posed and discussed in the paper.

3:35

2pSA6. State transitions in the Duffing resonator excited by narrow band random noise. Richard H. Lyon (60 Prentiss Ln., Belmont MA 02478-2021, rhlyon@lyoncorp.com)

The Dufffing hardening spring resonator has the status of a canonical model for nonlinear vibrations. Over a limited range of excitation frequencies and depending on the degree of nonlinearity, it has three states of response to sinusoidal excitation, one of which is unstable. The system will remain stably in either of the other two states depending on the history of excitation: in a higher energy state for the frequency ascending and in the lower energy state for the frequency descending. When the sinusoidal excitation is replaced by narrow band random excitation, the author showed in a 1961 paper experimentally that the system could make transitions between these two states and argued that the fluctuating phase of the excitation would allow the source to inject or draw energy from the resonator allowing a transition from one state to the other. This presentation develops a dynamic model for the system that allows energy transmission between the source and resonator and an indicator of the state of response based on instantaneous impedance.

3:55

2pSA7. Acoustic radiation pressure, torques, and scatterings: Insight for today from Gideon Maidanik's thesis research. Philip L. Marston, Likun Zhang, and David B. Thiessen (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Renewed interest in the identification of structural contributions to the scattering by complicated objects and the radiation forces and torques on simple objects in complicated acoustic beams makes it timely to review aspects of Maidanik's thesis research [J. Acoust. Soc. Am. 29, 738-742 (1957); 29, 936-940 (1957); 30, 620-623 (1958)]. Maidanik and Westervelt recognized the importance of King's earlier study of the low-frequency radiation forces on rigid movable spheres and extended that work to ka of 10. Hickling and Wang (1966) examined the properties of the scattering by such spheres and more recently Marston [J. Acoust. Soc. Am. 125, 3539-3547 (2009)] found that such a sphere approximates the low ka backscattering by an aluminum sphere in water. It was also found that negative radiation forces were possible for movable rigid spheres in a helicoidal Bessel beam. Recently, subtraction of complex amplitudes for movable semi-rigid targets has been used to isolate elastic contributions in finite-element based computations for finite cylinders [D. B. Thiessen *et al.*, J. Acoust. Soc. Am. 129, 2686 (2011)]. Maidanik's analysis of radiation torques has recently been reconciled with other approaches [L. K. Zhang and P. L. Marston, J. Acoust. Soc. Am. 129, 1679-1680 (2011)]. [Work supported by ONR and NASA.]

4:15

2pSA8. Estimating uncertainty in inverse elasticity with application to quantitative elastography. Paul E. Barbone, Bryan Chue (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu), and Assad A. Oberai (Rensselaer Polytechnic Inst., Troy, NY 12180)

We consider an inverse elasticity problem motivated by medical ultrasound imaging: Given a displacement field measured in a 2D domain, determine the modulus distribution in that domain. An iterative approach to solve the inverse problem can be formulated by repeated solutions of the forward problem. That is, the shear modulus distribution sought is that which predicts a displacement field most consistent with the measured displacement field and any assumed *a priori* knowledge of the modulus distribution. All such inverse problem solutions are subject to uncertainties in the data, however, which results in uncertainties in the predictions. For diagnostic purposes, it is desirable to know the confidence intervals within which the stiffness at a point might reside. The focus of this presentation is

the computation of said confidence intervals. We discuss the formulation of the problem within a Bayesian context. We derive a formal solution for the *a posteriori* probability distribution of the modulus. We prove bounds on uncertainty in terms of the data at the continuous level and discuss the computational solution of the problem at the discrete level.

4:35

2pSA9. Wave approach for the resonances of irregular polygonal membranes. Joseph Dickey (3960 Birdsville, Rd. Davidsonville, MD 21035, Joe@JoeDickey.com)

This study develops a wave or ray technique for determining the resonance frequencies of irregular polygonal membranes. The technique is demonstrated for homogeneous, isotropic, rectangular, and triangular membranes with fixed, free, and mixed boundaries. Where possible, the results are compared with exact calculations. The membrane resonances are calculated using an equivalent string whose length is proportional to the reciprocal of the length of closed paths starting from an arbitrary point within the membrane. Closed paths are ray paths which arrive back at the starting point going in the same direction. The extension of the technique to other irregular polygons and the relationship of the resonance determination in determining the response of the membranes to point excitation are discussed.

4:55

2pSA10. Sound-structure interactions in a Japanese drum. Yun-Fan Hwang (Fanacoustics, Inc., 3024 Rancho La Presa, Carlsbad, CA 92009, yfhwang1@gmail.com) and Hideo Suzuki (A and D Co., Ltd., Kitamoto 364-8585, Japan)

Previous studies of the sound-structure interaction of a Japanese drum conducted by the authors were focused on the vibration of and the coupling between the two membranes attached at both ends of an air-filled hollow wood body which was treated as a rigid cylindrical shell. This is satisfactory for the lower modes where sound is produced primarily by the vibration of membranes. At higher frequencies, the vibration of the wood barrel cannot be ignored. In the current study, the wood barrel is modeled by using conical shell elements. Orthotropic conical shell finite-elements, which include the rotary inertia and transverse shear deformation, have been developed and coded in MATLAB. Experimental verification of the computed results and the effect of wood barrel vibrations on the acoustical characteristics of a drum are discussed. [This paper is dedicated to honor Dr. Gideon Maidanik for his monumental contributions in structural acoustics. The authors would like to thank Miyamoto Unosuke Shouten Co., Ltd., for providing the Japanese drum in this study.]

TUESDAY AFTERNOON, 1 NOVEMBER 2011 — PACIFIC SALON 4/5, 1:00 TO 5:00 P.M.

Session 2pSCa

Speech Communication: Error Analysis in Automatic and Human Speech Recognition

Amit Juneja, Chair
Think A Move, Ltd., 23715 Mercantile Rd., Ste. 100, Beachwood, OH 44122

Chair's Introduction—1:00

Invited Papers

1:05

2pSCa1. Finding schwa: Comparing the results of an automatic aligner with human judgments when identifying schwa in a corpus of spoken French. Peter Milne (Dept. of Linguist., Univ. of Ottawa, 70 Laurier Ave. East, Ottawa, ON K1N 6N5, Canada, pmiln099@gmail.com)

This paper compares the results of an automatic aligner with human judgments when identifying schwa in a natural language corpus of spoken French. All word-final, postconsonantal possible schwa insertion sites in the standardized text portion of investigations conducted in both Qu'ebec and France were both hand coded for schwa and time aligned at the word and phone level by the Penn Phonetics Lab Forced Aligner, modified for use with French. The results of the two methods of coding were statistically compared to determine their level of agreement. Preliminary results show a strong correlation between the two methods. Possible effects due to dialect or phonetic context were investigated using a two-way, between subjects analysis of variance. A mixed design analysis of variance was also conducted. Initial results have found no significant effect due to dialect, but a possible effect due to context. This suggests that the results of automatic alignment are comparable with human judgments in both dialects of French, but results may differ for individual speakers in specific contexts. The advantages of using an automatic aligner that can aid in the collection of larger volumes of natural language data than is possible when working manually is discussed.

1:25

2pSCa2. Improving automatic speech recognition by learning from human errors. Bernd T. Meyer (Int. Comput. Sci. Inst., 1947 Center St., Ste. 600, Berkeley, CA, bmeyer@icsi.berkeley.edu)

This work presents a series of experiments that compare the performance of human speech recognition (HSR) and automatic speech recognition (ASR). The goal of this line of research is to learn from the differences between HSR and ASR and to use this knowledge to incorporate new signal processing strategies from the human auditory system in automatic classifiers. A database with noisy nonsense utterances is used both for HSR and ASR experiments with focus on the influence of intrinsic variation (arising from changes in speaking rate, effort, and style). A standard ASR system is found to reach human performance level only when the signal-to-noise ratio is increased by 15 dB, which can be seen as the human–machine gap for speech recognition on a sub-lexical level. The sources of intrinsic variation are found to severely degrade phoneme recognition scores both in HSR and in ASR. A comparison of utterances produced at different speaking rates indicates that temporal cues are not optimally exploited in ASR, which results in a strong increase of vowel confusions. Alternative feature extraction methods that take into account temporal and spectro-temporal modulations of speech signals are discussed.

1:45

2pSCa3. Automatic and human speech recognition in null grammar. Amit Juneja (Think A Move, Ltd., 23715 Mercantile Rd, Ste. 100, Beachwood, OH 44122)

The accuracy of automatic speech recognition (ASR) systems is generally evaluated using corpora of grammatically sound read speech or natural spontaneous speech. This prohibits an accurate estimation of the performance of the acoustic modeling part of ASR, since the language modeling performance is inherently integrated in the overall performance metric. Even though acoustic modeling accuracy for ASR can be evaluated on these corpora using a null grammar language model, the accuracy cannot be compared with human speech recognition (HSR) since human listeners cannot be asked to ignore grammar. In this work a null grammar speech corpus was collected for comparing HSR and ASR. The corpus was collected in a hemi-anechoic chamber using three different vocabulary sizes—1000, 5000, and 10000—in a quiet environment. Noisy speech files at different signal-to-noise ratios were generated by adding noise at different levels to the quiet speech recordings. Human listeners were employed to transcribe the recordings and their accuracy was compared with an ASR system under different vocabularies and noise levels.

2:05

2pSCa4. Being wrong—Insights from the speech-recognition battleground. Thomas U. Christiansen (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, DK-2800, Kgs. Lyngby, Denmark, tuc@elektro.dtu.dk) and Steven Greenberg (Silicon Speech, Kelseyville, CA 95451, steveng@silicon-speech.com)

Speech-recognition studies rarely use more than a single metric to evaluate recognition performance, usually percent correct (or percent wrong). Such a uni-dimensional evaluation may conceal more than it reveals. An alternative, based on information theory, offers greater insight into brain (and computational) processes associated with human and machine speech recognition. In this presentation, we examine errors associated with phonetic-segment recognition in human listeners and compare them with those committed by automatic speech-recognition (ASR) systems. Consonant errors are analyzed into the phonetic features of VOICING, place—(PLACE) and manner—(MANNER) of—articulation. For both humans and machines, PLACE information is far more vulnerable to distortion/interference than MANNER and VOICING, but is more important for consonant and lexical recognition than the other features. Moreover, PLACE is decoded only after VOICING and MANNER and is more challenging for machines to accurately recognize. The origins of these differences can be traced, in part, to the redundancy with which this information is distributed in the acoustic signal, as well as how the phonetic information is combined across the frequency spectrum. For such reasons, ASR performance could benefit by including phonetic-feature-based information in lexical representations. [Work supported by AFOSR and Technical University of Denmark.]

2:25

2pSCa5. Towards the recovery of targets from coarticulated speech for automatic speech recognition. John-Paul Hosom, Alexander Kain, and Brian Bush (Ctr. for Spoken Lang. Understanding, Oregon Health & Sci. Univ., 20000 NW Walker Rd., Beaverton, OR 97006)

An HMM-based ASR system tested on phoneme recognition of TIMIT (accuracy 74.2%) shows substitution errors covering all distinctive-feature dimensions of vowels: front/back, tense/lax, and high/low. These vowel-to-vowel errors account for about 30% of all substitution errors. These types of errors may be addressed by recovering vowel targets (and, as a by-product, coarticulation functions) during ASR. The current work models observed trajectories using a linear combination of target vectors, one vector per phoneme. A sigmoid function (with parameters for slope and position) models the evolution of the trajectory. In accordance with the Locus theory, if duration is sufficiently short and the rate of change is sufficiently slow, the targets may not be reached. Current data indicate that in clearly articulated speech, the vowel target is often reached, while in conversational speech, the vowel target is often not reached. This difference between speaking styles may explain the difficulty that current ASR systems have in recognizing conversational speech: by not always reaching the vowel target, the observed values for a phoneme have higher variance and increased overlap with other phonemes. By recovering the target values, variance of phonemes within the feature space may be reduced, thereby improving classification accuracy. [Work supported by NSF Grant IIS-0915754.]

2:45

2pSCa6. How knowledge of speech acoustics can improve the robustness of automatic speech recognition. Carol Espy-Wilson (Dept. of Elec. and Comput. Eng., Univ. of Maryland, College Park, MD 20742), Suzanne Boyce (Univ. of Cincinnati, Cincinnati, OH 45267), and Abeer Alwan (UXLA, Los Angeles, CA)

In this talk, we discuss the ways in which the parallel study of speech production and speech perception can help us develop better automatic speech recognition systems. The ultimate goal of speech recognition (recognition of spontaneous speech from any talker in any language) is still elusive due to a high degree of inter- and intra-speaker variability for production of a given sequence of sounds. While the acoustic information required for recognition may be present in the signal, its distribution, strength, and location are consistent and predictable only as a function of lawful changes in speech movements and/or listener perceptions. Understanding speech acoustics from this perspective is vitally important if we are going to achieve our ultimate goal. We will give several examples of lessons learned from studies of speech production and speech perception and how the knowledge gained can inform the engineering of robust ASR systems.

3:05–3:20 Break

3:20

2pSCa7. Recent progress in articulatory modeling for speech recognition. Karen Livescu (Toyota Technol. Inst. at Chicago, 6045 S. Kenwood Ave., Chicago, IL 60637)

The automatic speech recognition research community has experimented with models of speech articulation for several decades, but such models have not yet made it into mainstream recognition systems. The difficulties of adopting articulatory models include their relative complexity and dearth of data, compared to traditional phone-based models and data. This talk will review the current state of articulatory models and will describe one particular approach to incorporating such models in modern speech recognition. In this approach, the articulatory variables are based on the vocal tract variables of articulatory phonology, and the models are represented using dynamic graphical models, a generalization of the more commonly used hidden Markov models. This approach allows the probabilistic modeling of asynchrony between articulators and reduction in articulatory gestures. Results will be presented showing improvements in lexical access using this type of articulatory model with automatically learned context-dependent articulatory feature distributions. Recent efforts to mitigate the data sparseness problem, including manual and automatic transcription, will also be presented.

3:40

2pSCa8. Robust speech recognition with articulatory features using dynamic Bayesian networks. Vikramjit Mitra (Speech Technol. and Res. Lab., SRI Int., 333 Ravenswood Ave., Menlo Park, CA 94025), Hosung Nam (Haskins Labs., New Haven, CT 06511), Carol Espy-Wilson (Univ. of Maryland, College Park, MD 20742), Elliot Saltzman (Boston Univ., Boston, MA 02115), and Louis Goldstein (Univ. of Southern California, Los Angeles, CA 90089)

Previous studies have proposed ways to estimate articulatory information from the acoustic speech signal and have shown that when used with standard cepstral features, they help to improve word recognition performance in noise for a connected digit recognition task. In this paper, I present results from a word recognition and a phone recognition experiments in noise that uses two sets of articulatory representation: continuous (tract variable trajectories) and discrete (articulatory gestures) along with standard mel cepstral features for acoustic modeling. The acoustic model is a dynamic Bayesian network (DBN) that treats the continuous articulatory information as observed and the discrete articulatory presentation as hidden random variables. Our results indicate that the use of articulatory information improved noise robustness for both the word recognition and phone recognition tasks substantially.

4:00

2pSCa9. Semi-supervised learning for speech and audio processing. Mark A. Hasegawa-Johnson, Jui-Ting Huang, and Xiaodan Zhuang (ECE Dept., Univ. of Illinois. Urbana, IL 61801)

Semi-supervised learning requires one to make assumptions about the data. This talk will discuss two different assumptions, and algorithms that instantiate those assumptions, for the tasks of acoustic modeling and pronunciation modeling in automatic speech recognition. First, the acoustic spectra corresponding to different phonemes overlap, but there is a tendency for the instantiations of each phoneme to cluster within a well-defined region of the feature space—-a sort of “soft compactness” assumption. Softly compact distributions can be learned by an algorithm that encourages compactness without strictly requiring it, e.g., by maximizing likelihood of the unlabeled data, or even better, by minimizing its conditional class entropy. Second, the observed phone strings corresponding to coarticulated pronunciations of different words are also, often, indistinguishable, but can be transformed into a representation in which the degree of overlap is substantially reduced. The canonical phonetic pronunciations are transformed into an articulatory domain, possible mispronunciations are predicted based on a compactness criterion in the articulatory domain, and the result is transformed back into the phonetic domain, forming a finite state transducer that is able to effectively use hundreds of alternate pronunciations.

4:20

2pSCa10. Dealing with unknown unknowns in speech. Hynek Hermansky (Ctr. for Lang. and Speech Processing, The Johns Hopkins Univ., Baltimore, MD 21218)

Common belief in speech recognition community is that most significant improvements in performance on a machine come from more training data. Implicit is a tacit assumption that speech to be recognized comes from the same distribution as the speech on which the machine was trained. Problems occur when this assumption is violated. Words that are not in a lexicon of a machine, unexpected distortions of a signal and noises, unknown accents, and other speech peculiarities all create problems for the current ASR. The problem is inherent to machine learning and will not go away unless alternatives to extensive reliance on false beliefs of unchanging world are found. In an automatic recognition of speech, words that are not in the expected lexicon of the machine are typically substituted by some acoustically similar but nevertheless wrong words. Similarly, unexpected noise is typically ignored in human speech communication but causes significant problems to a machine. We discuss a biologically inspired multistream architecture of a speech recognition machine that could alleviate some of the problems with the unexpected acoustic inputs. Some published experimental results are given.

4:40–5:00 Panel Discussion

Session 2pSCb

Speech Communication: Deep Brain Stimulation in Parkinson's Disease: Speech and Nonspeech Outcomes

Emily Q. Wang, Cochair
Communication Disorders and Science, Rush Univ., 1653 W. Congress Pkwy., Chicago, IL 60612

Anders L. Lofqvist, Cochair
Dept. of Logopedics and Phoniatrics, Lund Univ., Lund, S-221 85, Sweden

Charles R. Larson, Cochair
Communication Science Disorders, Northwestern Univ., 2240 N. Campus Dr., Evanston, IL 60208

Chair's Introduction—1:30

Invited Papers

1:35

2pSCb1. Deep brain stimulation in Parkinson's disease: The basics. Leo Verhagen (Dept. of Neurological Sci., Rush Univ. Medical Ctr., 1725 W. Harrison, Chicago, IL 60612, lverhage@rush.edu)

Parkinson's disease (PD) affects over 1×10^6 people in the United States with 50 000 Americans being diagnosed each year. As a neurodegenerative movement disorder, it affects patients' lives through its slow but relentless progression of both motor and non-motor symptoms. Initially, most PD patients receive good benefit from dopaminergic treatment, but over time the symptomatology changes. The goal of this presentation is to discuss the motor features of advanced PD. Treatment of advanced PD typically consists of a combination of pharmaceuticals, but in recent years deep brain stimulation has been increasingly used to complement medical therapy. DBS for PD is indicated in some patients while it may not be the best treatment option for others. Selection criteria, indications, and relative contraindications will be discussed. The procedure will be reviewed and an overview of recent outcomes of DBS studies in PD will be provided. Attention will also be given to potential side effects of this state of the art treatment. Throughout the presentation, video clips will highlight the phenomenology under discussion.

2:05

2pSCb2. Effect of deep brain stimulation on speech in Parkinson's disease: From research lab to clinic. Emily Wang (Dept. of Commun. Disord. & Sci., 1611 W. Harrison, Ste. 530, Chicago, IL 60612, emily_wang@rush.edu)

Bilateral deep brain stimulation (DBS) of the subthalamic nucleus (STN), an evidence-based, effective surgical treatment with increasing popularity, can help patients with advanced Parkinson's disease eliminate or lessen many of the motor symptoms they experience. However, one of its unfortunate adverse effects is that it may worsen existing speech impairment as well as causing new impairment. As a result, the patients may be unable to speak loudly or clearly, or simply unable to initiate speech. The problems often associate with active as well as chronic STN stimulation. Further, there does not seem to be a uniform effect on different speech subsystems of respiration, phonation, and articulation. In this presentation, we will first review studies which have shown differential effects on phonation, articulation, and prosody associated with unilateral vs. bilateral STN DBS. Next, we will discuss the preliminary outcomes of several potential treatment approaches and strategies including the Lee Silverman voice treatment, change of DBS settings, and the altered auditory feedback. Lastly, we will discuss our experience of using the approaches and strategies helping patients with their speech deficits associated with STN DBS in clinical settings. Both mechanisms and limitations will be explored.

2:35–2:50 Break

2:50

2pSCb3. Effect of subthalamic nucleus deep brain stimulation on tremor, rigidity, muscle strength, and movement. Daniel M. Corcos (Dept. of Kinesiology and Nutrition, Univ. of Illinois at Chicago, 1919 West Taylor St., Chicago, IL 60612)

The goal of this presentation is threefold. First, we will review studies in which we have shown the dramatic benefit of subthalamic nucleus (STN) deep brain stimulation (DBS) on bradykinesia, tremor, and rigidity. We will show that limb tremor is normalized, movement speed is increased, muscle activation patterns resemble those of healthy individuals, and rigidity is substantially reduced. Second, we will show that there is no difference between 90 min and greater than 3 months of STN stimulation for both the UPDRS or motor control measures. This finding confirms that the treatment efficacy that is derived from a short time course of stimulation generalizes to the longer time periods of STN stimulation that patients experience in their daily lives. Finally, we will conclude by presenting the effects of five years of continuous STN stimulation on muscle strength and movement speed. We will show that despite the fact that patients become more parkinsonian as measured by the UPDRS, they become stronger and faster at making simple movements. These results will be discussed in the context of models of therapeutic efficacy that are predicated on the idea that STN DBS reduces neuronal noise and thus both facilitates simple movements and the reduction of tremor.

3:20

2pSCb4. Differential effects of deep brain stimulation (DBS) on speech and limb movements in Parkinson's Disease (PD): Clues to basic mechanisms. Howard Poizner (Inst. for Neural Computa., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093)

A puzzling finding has been that STN DBS improves limb motor function, while at the same time providing little benefit to or even markedly worsening speech. In collaboration with Peter Fox's Research Imaging Institute in San Antonio, we combined objective measures of speech with PET imaging and TMS in a PD patient with speech deficits due to STN DBS. Stimulating the left STN produced deteriorated speech together with hyperactivation of left dorsal premotor cortex (PMd), an area known to project to the left STN. TMS to left PMd with DBS off, produced similar speech impairments as DBS stimulation. These findings are consistent with STN DBS antidromically disrupting left PMd activity, thereby causing speech deterioration. In contrast, STN DBS is known to improve limb motor control. We have been examining the effects of STN DBS on reaching to and grasping objects, on reaching to kinesthetically defined 3-D targets, and on inhibiting a pre-programmed action. We have found that bilateral STN DBS improves all of these behaviors, although to varying degrees. Moreover, EEG recordings during response inhibition suggested that the physiological mechanism for the improved behavioral control involves normalization of brain rhythms that may be involved in transferring information within cortico-basal ganglia circuits. [Work supported in part by NIH grant #2 R01 NS036449 and ONR MURI Award No.: N00014-10-1-0072.]

3:50–4:20 Panel Discussion

TUESDAY AFTERNOON, 1 NOVEMBER 2011 — ROYAL PALM 1/2, 1:00 TO 3:15 P.M.

Session 2pSP

Signal Processing in Acoustics: Target Detection and Sonar Related Issues

Jan Dettmer, Cochair
School of Earth and Ocean Sciences, Univ. of Victoria, Victoria, BC V8W 3P6, Canada

Ravi Menon, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

1:00

2pSP1. Mutual interference signal processing for active sonar. Stephen D. Unruh, Jason M. Aughenbaugh, and James M. Gelb (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758)

Active sonar for anti-submarine warfare (ASW) can at times be hampered by interference from the transmissions of other vessels—either friend or foe. For practical reasons, there are typically only a limited number of center-frequencies, bandwidths, and useful waveform types available. We explore the cross-correlation of various frequency modulated waveforms with varying characteristics—particularly bandwidths, pulse lengths, and the degree of frequency overlap. The impact on the ambiguity function is also explored, along with the efficacy of several transmitter and receiver filters. Real world data containing interfering transmissions are presented including examples where basic theoretical effects are revealed. This work represents an early stage of research to ultimately explain subtle nuances in real data and to develop novel signal processing techniques to mitigate interference. [Work sponsored by ONR 321US.]

1:15

2pSP2. Characterization of non-Gaussian, multi-static clutter from a mud volcano field. John R. Preston (ARL, Penn State Univ,, P.O. Box 30, MS 6110D, State College, PA 16804) and Douglas A. Abraham (Causa Sci LLC, P.O. Box 627, Ellicott City, MD 21041)

Sonar clutter is one of the primary limitations to active ASW. This work focuses on statistical analysis of clutter-like returns from some multi-static measurements. Non-Gaussian characterizations of multi-static clutter from a mud volcano field are presented. The received data are taken from either the Five Octave Research Array (FORA) or the NURC triplet array that have been used to collect extensive monostatic and bistatic data in a recent sea trial on the Malta Plateau off Sicily called Clutter 07. This work uses data from that sea trial to characterize non-Gaussian behavior of multi-static clutter from a mud volcano field using pulsed sources in the 800–3500 Hz band. Either the Page test or a maximum likelihood procedure is used to isolate the clutter-like returns before processing. K-distributions with their shape and scale parameters are used to describe non-Gaussian behavior together with the models of Abraham and Lyons to infer physical descriptors from the clutter. The ability to geo-reference key statistical measures of clutter allows CFAR processors to adaptively set thresholds and reduce false alarms. Examples are shown to demonstrate this. Also included are presentations of the shape parameter versus bistatic aspect angle and the cumulative density functions for this parameter. [Work supported by ONR code 321US.]

1:30

2pSP3. Understanding the feedback effect through the physics of the diffraction-based sensitivity kernel for a target in shallow water: A small-scale experimental demonstration. Christian Marandet, Philippe Roux, Patrick La Rizza (ISTerre, BP 53, 38 041 Grenoble Cedex 9, France, christian.marandet@gmail.com), and Barbara Nicolas (GIPSA-LAB, 38 042 St Martin d Hres, France)

Using the feedback effect, we experimentally detect a wavelength-sized target in a shallow ultrasonic waveguide between two source-receiver transducers on acoustic feedback. The waveguide represents a 1-km-long, 50-m-deep ocean acoustic channel at the 1/1000 scale. The feedback phenomenon, or Larsen effect, occurs when a source and a receiver are connected both acoustically through the propagation medium and electrically through an amplifier in such a way that the received signal is simultaneously and

continuously added to the emitted signal. A resonance is obtained when the emitter and the receiver are in phase. This resonance is very sensitive to any change in the medium which makes it a good observable for target detection. In presence of a target in the waveguide, the numerical gain of the feedback effect has to increase in order to compensate the scattering of the acoustic field from the target. In a separate experiment, the scattered field may also be recorded in a transmission configuration from the same couple of emitter/receiver with an impulse as a source signal. A comparison is made between the two different approaches.

1:45

2pSP4. Acoustic diver deterrent in a shallow harbor using time reversal acoustics. Alexander Sutin and Yegor Sinelnikov (Sound Interventions, Inc., 25 Health Sciences Dr., Ste. 201, Stony Brook, NY 11790, ysinelnikov@yahoo.com)

Protection of domestic harbors against surface and underwater threats is an important task of port security. A significant security risk is associated with scuba divers ability to carry explosives. The diver detection sonars have been developed and there is a need to compliment it with low cost acoustic swimmer deterrent. Previous research demonstrated that time reversal acoustic (TRA) system can focus intensive sound using acoustic noise from a diver. This paper discusses the feasibility of applying TRA principles for the focusing of sonar ping-pulse reflected from a diver. The advantages of the TRA focusing system and relevant operating parameters are demonstrated using the model of shallow sea, where propagation of sonar pulse is affected by numerous reflections from surface and bottom. The developed model was used to estimate and compare an effective zone of the diver deterrence taking into consideration the sonar pulse reflections from diver, bottom and surface. The main objective was to estimate if NAVY ship sonar is capable of producing sufficient and spatially localized sound pressure to disorient and divert the diver away. A secondary objective was to ensure that sound pressure level outside the focal zone in the diver proximity remains not harmful to marine life.

2:00

2pSP5. A multi-beam array technique for acoustical imaging. John F. Brady and Dipen N. Sinha (Los Alamos Natl. Lab., MSD429, Los Alamos, NM 87544, jbrady@lanl.gov)

Most acoustical imaging systems rely on phase-steered or multiplexed transducer arrays, requiring complex electronics and powerful data acquisition. This presentation discusses a 1-D array capable of generating multiple simultaneous directional beams in both transmit and receive modes. The array can image over a 50 deg window in a single pulse-echo cycle while requiring as few as four electronic channels for operation. The advantages of this simple approach compared to techniques previously mentioned will be discussed.

2:15

2pSP6. Solution behaviors of multistatic transmission loss equations. Hisashi Shiba (Radio Application Div. NEC Corp., 1-10, Nisshin-Cho, Fuchu, Tokyo 183-8501, Japan, h-shiba@aj.jp.nec.com)

A sonar arrangement is one of the most important problems in the multistatic sonar. Although it is very useful to proceed numerical simulations for arrangements, they usually consume much computing resources and their results are sometimes difficult to grasp physical meanings. On the other hand, it is beneficial to analyze the acoustical field further from the classical sonar equation view point as a quick looking. One of the most important element is the transmission loss in the sonar equation. In the multistatic configurations, the transmission loss is dependent on the direction from a receiver. The transmission loss is usually notified as a function using transmitter parameters. New descriptions are introduced without transmitter parameters in this presentation. Spreading dominant cases produce quartic transmission loss functions of the target distance form the receiver. They are solved analytically; however, the solutions do not constitute Cassini oval in spite of some preceding researches. Absorption dominant cases are also solved analytically. Mixtures of spreading dominant and absorption dominant are not able to be solved analytically. These solution behaviors are easily surveyed by contour maps of the transmission loss. These maps are useful for sonar arrangements.

2:30

2pSP7. Time-varying filter estimation for the deconvolution of environmental reverberation from active sonar returns. Kevin D. LePage and Ryan Goldhahn (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, SP Italy)

The estimation and removal of the time-varying two-way impulse response to environmental scatterers from broadband reverberation data is considered for increasing the signal-to-noise ratio of sonar returns from targets in the water column. Spectrograms of simulated and real reverberation time series data from active sonars in the mid-frequency range show strong evidence of interference patterns which give clues to the number of important paths to environmental scatterers as well as their depth in the water column. In this talk we consider the estimation of a time dependent deconvolution filter for the removal of these environmental reverberation returns from active sonar data. Issues regarding the degrees of freedom required for the efficient implementation of this filter and the stability of these estimates are considered. Simulation results are shown which demonstrate the potential gain of using this approach to partially null the impact of environmental scatterers in active sonar data.

2:45

2pSP8. Near-field blind deconvolution in a reverberant environment. Shima H. Abadi, Eric S. Haapaniemi, Andrew J. Femminineo, Laura M. Williamson, and David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48109-2133)

Artificial time reversal (ATR) is a passive technique for blind deconvolution in an unknown multipath environment that relies on generic features of underwater sound fields. ATR has been found to be effective when the source is far from the receiving array and the receiving array properly resolves propagating modes or ray-path arrival angles throughout the bandwidth of the source signal. This presentation describes the results of an experimental investigation into ATR's performance for a near-field source in a highly reverberant environment. The experiments were conducted with nominally 0.1 ms pulses at frequencies from 20 kHz to 150 kHz in a 1.0-m-deep and 1.07-m-diameter cylindrical water tank having a reverberation time of $\sim$10 ms using a single sound projector and a linear receiving array of 16 hydrophones. The correlation coefficient between the original and the ATR-reconstructed signals is presented as a function of receiving-array and broadcast-signal characteristics and compared to equivalent signal reconstruction results from spherical-wave delay-and-sum beamforming. The intended application of this research lies in determining the acoustic signature of cavitation bubbles and other hydroacoustic sound sources in hydrodynamic test facilities. [Sponsored by ONR and by NAVSEA through the Naval Engineering Education Center.]

3:00

2pSP9. The influence of suppression of side lobes on the range of passive sonars. Zvonimir Milosic (MORH Sarajevska 7, IROS Ilica 156 b, Zagreb, zvonimir.milosic@morh.hr)

This paper presents a universal procedure of the de-embedding of mathematical functions of ratios of maximal ranges depending on the level of suppression of side lobes at the directivity pattern of passive sonar antenna. This dependence is founded on the specially made and named as "idealized model" of measured directivity pattern of a sonar antenna with the model of elliptical cross section of the main lattice perpendicular on acoustic axes. Using the given mathematical expressions of directivity index and dependence of suppression of side lobes at the mentioned idealized model of directivity pattern characteristics with equations of hydrolocations and their conditions, there are relatively simple expressions of ratio of maximum ranges in form of defined function mz. Using the given mathematical function mz there is possible to control the value of all important parameters for the de-embedding of the sonar ranges in its specification. In accordance with given model, there are point out the importance of ratio of ranges of maximum value of level of suppression of side lobes at directivity pattern characteristic with elliptical cross section of main lattice of antenna. Mainly, the level of suppression of side lobes is less than -50 dB at contemporary passive sonars today.

Session 2pUW

Underwater Acoustics and Acoustical Oceanography: Theory and Practical Applications for Bottom Loss II

Nicholas P. Chotiros, Cochair
Applied Research Lab., Univ. of Texas, P.O. Box 8029, Austin, TX 78713

Martin Siderius, Cochair
Electrical and Computer Engineering, Portland State Univ., 1900 S.W. 4th Ave., Portland, OR 97201

Roger W. Meredith, Cochair
U.S. Oceanographic Office, Stennis Space Center, MS 39529

Invited Papers

1:00

2pUW1. High frequency bottom loss database generation at the naval oceanographic office. Jacob George, David Harvey, and Jorge Novarini (Naval Oceanogr. Office, 1002 Balch Blvd., Stennis Space Ctr., MS)

The Naval Oceanographic Office (NAVOCEANO) generates the low- and high-frequency bottom loss databases (LFBL and HFBL). In support of these upgrades, NAVOCEANO carries out survey operations to make acoustic and geophysical measurements. From these measurements, bottom loss parameters are extracted via numerical inversions. The HFBL database describes acoustic bottom loss over the range 1.5-4 kHz and is based on a set of nine bottom-loss curves derived from the Marine Geological Survey carried out by NAVOCEANO (1965–1968). The optimum curve at a given location is derived from a process of brute force inversion using measured transmission loss (TL) data. Curves are interpolated to the nearest 1/10th; they are then input to the CASS model to determine which curve optimizes the fit between the CASS prediction and the measured TL data. Optimal curves are determined independently at each of the 1/3 octave frequencies over the HFBL range. Studies of the derived curve values in a recent upgrade area have indicated very high variability. This variability does not tend to correlate with bottom sediment type, and the variability tends to be highest in the higher frequency bands. An implication is that scattering caused by roughness at the water/sediment interface may be driving variability.

1:20

2pUW2. Prediction of marine fine-grain sediment states: Determinants of mine burial and acoustic impedance. Richard H. Bennett (SEAPROBE, Inc., 501 Pine St., Picayune, MS 39466), Conrad W. Curry, Roger W. Meredith (Stennis Space Ctr., MS 39529), and Richard W. Faas (Univ. of Southern MS, MS 39529)

The predicted depth of mines buried in marine muds is generally based on estimates of sediment shear strength (often unreliable). Conversely, sediment states of marine muds are water-dependent, defined empirically by the Atterberg Limits (liquid limit and plastic limit), and allow the sediment to be described as having fluid-like, plastic-like, or semi-solid consistency. When the natural water content and the liquid limit of normally and unconsolidated marine muds are approximately equal at depth below the seafloor, the mud at greater depth is considered to no longer behave as fluid-like, but plastic-like. This relationship provides a predictable conservative minimum mine burial penetration depth. Mine burial depths at two sites were shown to closely agree with predicted burial depths based on the natural water contents and the liquid limits (Bennett *et al*., SEAPROBE, Inc., Technical Report Number SI-0004-01, p., 89, 2004, funded by ONR). Prediction of selected sediment physical properties using acoustic impedance as a function of depth below seafloor may provide a method to estimate and evaluate sediment states. Comparison of subbottom natural water contents with a database showing known liquid limits for different types of marine muds should make possible prediction of conservative, minimum, mine burial depths.

1:40

2pUW3. Experience with geoacoustic inversion of transmission loss. Paul J. Vidmar (SAIC, 4001 N. Fairfax Dr., Arlington, VA 22203)

This presentation will discuss recent experience with geoacoustic inversion of transmission loss (TL) data from 50 to 5000 Hz from both deep and shallow water regions. For shallow water, a multi-layered geoacoustic profile with range dependent layer thicknesses is used. Layer thickness is not an inversion parameter but is known from ancillary data such as seismic profiling and chirp sonar. Issues related to range dependent sound speed profiles (SVPs) will be discussed as will observation of the time spread of multipath arrivals. For deep water, a single layer geoacoustic profile is assumed. Quality of inversion is assessed using comparison of measured and modeled TL and the estimation error. The estimation error is the dependence of the total rms error—summed over range, frequency, and receivers—on the values of a geoacoustic parameter within the bounds used to constrain inversion. The estimation error identifies geoacoustic parameters that are well constrained or poorly constrained by the TL data. Inversion was carried out using a genetic algorithm model [D. Harvey *et al*., Oceans '02 MTS/IEEE, **1**, 358–362 (2002)]. Acoustic and environmental data were provided by the Naval Oceanographic Office. [Work funded by the Ocean Bottom Characterization Initiative through PEO C4I, PWM 120.]

2:00

2pUW4. Relating volume scattering from the seafloor to dispersion and attenuation within the seafloor. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Heterogeneities in ocean sediments can produce significant scattering of sound from the seafloor, particularly in soft sediments. First-order volume perturbation models for scattering from the seafloor into the water ignore the effect of this scattering on propagation and dispersion within the sediment. The energy that is scattered, however, contributes to the attenuation of the penetrating sound. This increase in attenuation, and its effect on dispersion, is modeled by applying perturbation theory to sound propagation through a fluid sediment. This allows the application and predictions of this propagation model to be connected directly to previous perturbation approximations of scattering from the sediment. Several sediments for which volume scattering has been previously studied are revisited in the context of sound propagation. The implications of this loss mechanism within the sediment for the scattering of sound from the sediment are also considered. [Work supported by the Office of Naval Research.]

Contributed Papers

2:20

2pUW5. Comparison of parabolic equation and coupled mode solutions to seismo-acoustic problems. Scott D. Frank (Dept. of Mathematics, Marist College, 3399 North Ave., Poughkeepsie NY 12601, scott.frank@marist.edu), Robert I. Odom (Univ. of Washington, Seattle, WA 98105), Minkyu Park (Korea Polar Res. Inst., Incheon, 406-840, Korea), and Jon M. Collis (Colorado School of Mines, Golden, CO 80401)

Parabolic equation methods that use the single scattering approximation and improved rotated coordinate methods generate accurate and efficient solutions for range-dependent underwater acoustic problems with elastic sediments. Recently these methods have demonstrated the conversion of acoustic energy in a fluid into shear propagation in an underlying elastic layer. Elastic coupled mode theory can also be used in a range-dependent environment and provides an accurate description of the conversion between fluid and elastic modes during propagation. The current parabolic equation approach will be augmented with an elastic version of the self-starter that includes both compressional and shear wave energy. Results from these two approaches will be compared for acoustic and elastic sources in both range-independent and range-dependent underwater environments. Propagation at the fluid–solid interface will be examined as a possible mechanism for the conversion of elastic layer shear energy into acoustic energy in the water column. An elastic source is used to demonstrate that sources of this type can transmit substantial acoustic energy into the water column. [Work supported by ONR.]

2:35

2pUW6. Initial assessment of combustive sound source arrays as airgun alternatives for Arctic under-ice seismic exploration. Juan I. Arvelo, >Jerrold Dietz (Appl. Phys. Lab., The Johns Hopkins Univ. 11100 Johns Hopkins Rd., Laurel, MD 20723-6099), Andrew R. McNeese, Jason D. Sagers, and Preston S. Wilson (The Univ. of Texas at Austin, Austin, TX 78712-0292)

Combustive sound source arrays consisting of submersible combustion chambers filled with a hydrogen/oxygen mixture are employed to assess their effectiveness for seismic exploration applications. The combustive mixture is ignited via spark and radiates acoustic pulses capable of undersea deep sub-bottom sediment penetration. Since electrolytic cells may be employed to generate the hydrogen/oxygen mixture from surrounding seawater, this source is an attractive alternative to airgun arrays for Arctic seismic exploration from under-ice platforms. Combustive sound source array configurations were tested in a central Virginia basin with hydrophones deployed in a line at another nearby basin. Seismic reflections are compared against nearby geologic cross sections of the central Virginia seismic zone. [Funding provided via UAF sub-award under NOAA Grant NA09NOS4000262 and ONR.]

2:50–3:05 Break

3:05

2pUW7. Efficient parabolic equation modeling with shear (SCRAM). Richard L. Campbell, Kevin D. Heaney (Ocean Acoust. and Instrumentation Systems, Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, rlcamp.pdx@gmail.com), Alec J. Duncan, and Robert D. McCauley (Curtin Univ., Western Australia)

A novel parabolic equation model has been developed based upon Michael Collins RAM algorithm. By separating the PE solver from the fixed range-depth slice geometry of RAM, efficiency is significantly improved for full-field $\times$ propagation to points on a plan-view grid or along an arbitrary vessel track. The code is written in C with the ability to fully leverage modern many-core computers. In this paper, the SCRAM model is introduced, based on the PE propagator from Collins RAMS variant, extending CRAM to environments with sediments supporting shear wave propagation. Comparison with range-independent wave-integral solutions will be made. Specific application to the problem of propagation over continental shelves with calcerenite seabeds, such as those observed in the seas off the coast of Australia will be examined. Model-data comparisons for measurements taken near Dogan, Australia, will be made.

3:20

2pUW8. The depth dependence of earthquake T-phases at an ocean acoustic observatory. Ralph A. Stephen, S. Thompson Bolmer (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543-1542), Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0225), James A. Mercer (Univ. of Washington, Seattle, WA 98105-6698), and Bruce M. Howe (Univ. of Hawaii at Manoa, Honolulu, HI 96822)

T-phases are earthquake signals that have propagated, at least partially, in the ocean sound channel. T-phase hydrophone networks detect much smaller earthquakes over basin scales than land-based networks and they detect many more earthquakes than comparable regional scale seismic land networks. Furthermore, since T-phases travel at lower velocities than seismic phases, they result in much more precise locations of events given the same timing accuracy. T-phases are typically spread over 10's of seconds, and a common problem, however, is precisely identifying the arrival time of an event. T-phase stations usually consist of single hydrophones moored near the sound channel axis and the depth dependence of the T-phase envelope and frequency content is rarely studied. In the North Pacific Ocean, from 2004 to 2005, ambient noise and earthquakes were observed at an ocean acoustic observatory consisting of a vertical hydrophone array (from about 750 m above the seafloor to 375 m from the surface) and three co-located ocean bottom seismometers. This data set provides a unique opportunity to observe earthquake signals and their characteristics throughout the water column and to provide ground-truth to theoretical predictions on the excitation and propagation mechanisms of T-phases. In at least one case, a T-phases from a distant earthquake was readily observed even at the seafloor, well-below the conjugate depth.

3:35

2pUW9. Time-evolving T-phase arrival structure using simultaneous recordings by large-aperture horizontal and vertical line arrays in PhilSea09. Simon E Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, sfreeman@ucsd.edu), Gerald L. D'Spain (Scripps Inst. of Oceanogr., San Diego, CA 92106), Ralph Steven (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Kevin D. Heany (Oasis Inc., Lexington, MA 02421), Arthur Baggeroer (MIT, Cambridge, MA 02139), Peter Worcester (Scripps Inst. of Oceanogr., La Jolla, CA 92037), Jim Mercer (Univ. of Washington, Seattle, WA), Stephen Lynch (Scripps Inst. of Oceanogr., San Diego, CA 92106), and Jim Murray (Oasis Inc., Lexington, MA 02421)

A number of models have been proposed to explain the mechanisms by which seismic phases couple to the deep ocean sound channel in order to create water-borne acoustic tertiary (T) phases. Beamforming conducted on simultaneous recordings by large-aperture horizontal towed and vertical moored line arrays during PhilSea09 shows the temporal evolution of a T-phase arrival consistent with the down-slope modal conversion/propagation model. Towed array calibration is conducted using ship-deployed, controlled multi-tone acoustic sources. Conventional, minimum variance distortionless response, white noise constrained, and dominant mode rejection beamformers are compared in their ability to minimize bias and variance in estimating the azimuthal arrival directions of signals from both the controlled source and the seismic phases recorded by the horizontal array. Horizontal array beamformer-derived azimuth and time-of-arrival range estimates from P, S, and T-phase arrivals at towed and moored receivers indicate the event occurred in a region with appropriate bathymetric relief for down-slope conversion/propagation. The seismic event in question was not recorded by the USGS/NEIC seismometer network. This study thus further showcases the highly sensitive capabilities of in-water hydrophones and the effect of array gain to characterize high-frequency (5–50 Hz) seismic events. [Work supported by the Office of Naval Research.]

3:50

2pUW10. Seismic tremor event intervals from dual-frequency coherence. LeRoy M. Dorman (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA, ldorman@ucsd.edu), Susan Y. Schwartz (Earth & Planetary Sci., UCSC, Santa Cruz, CA 95064), and Michael Tryon (SIO, UCSD, 92093-0220)

Slip occurring at plate boundaries creates seismic tremor as well as "normal" earthquakes. This nonvolcanic tremor appears to consist of swarms of low-frequency earthquakes which lack impulsive P and S arrivals. Tremor is accompanied by slip observed by GPS and can show anomalies in fluid flow. The seismic radiation resembles continuous microseismic noise more than discrete events. We report dual-frequency coherence (DFC) calculations on tremor and normal microseismic background noise observed on Ocean-Bottom Seismographs and land seismic stations around the Nicoya Peninsula, Costa Rica. Both the OBS and land tremor signals show a banded pattern in DFC that is absent in normal noise. The similarity in the DFC patterns between OBS and land tremor signals suggests a common source, eliminating the possibility that DFC is a property of the OBS or seafloor environment. Banded DFC patterns can be generated by repeated events with a repeat time equal to the reciprocal of the offset frequency between bands. If, as is becoming widely accepted, nonvolcanic tremor consists of swarms of low frequency earthquakes (LFEs), DFC analysis may help to reveal LFE periodicities or intervals.

4:05–5:05 Panel Discussion

TUESDAY AFTERNOON, 1 NOVEMBER 2011 — CRESCENT, 1:00 TO 2:00 P.M.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

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

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

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

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, November 1, 2011.

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

Meeting of Accredited Standards Committee (ASC) S12 Noise

W. J. Murphy, Chair, ASC S12
NIOSH, 4676 Columbia Pkwy., Mail Stop C27, Cincinnati, OH 45226

R. D. Hellweg, Vice Chair, ASC S12
Hellweg Acoustics, 13 Pine Tree Rd., Wellesley, MA 02482

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

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note—that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, November 1, 2011.

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

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

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

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

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

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

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Tuesday are as follows:

Acoustical Oceanography	Pacific Salon 2
Architectural Acoustics	Sunrise
Engineering Acoustics	Pacific Salon 6/7
Musical Acoustics	Towne
Physical Acoustics	Royal Palm 3/4
Psychological and Physiological Acoustics	Esquire
Structural Acoustics and Vibration	Sunset