Session 2aAAa

Architectural Acoustics and Psychological and Physiological Acoustics: Architectural Acoustics and Audio II

K. Anthony Hoover, Cochair
McKay Conant Hoover, Inc., 5655 Lindero Canyon Rd., Westlake Village, CA 91362

Alexander U. Case, Cochair
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Invited Papers

8:00

2aAAa1. When an idea goes in a different direction from that expected. Thomas Plsek (Berklee College of Music, MS-1140 BRASS, 1140 Boylston St., Boston, MA 02215, tplsek@berklee.edu)

Quite a few years ago when airlines were using the simple headsets with the rubber tubes, it was realized that they could be coupled to a trombone practice mute to make a system that one could use to practice without disturbing others. As it happened, the idea did not for various reasons work as expected. It was realized that a new instrument could be created by placing the headphones on someone other than the performer. By doing so, it was realized that the acoustic feedback normally received by the performer would be eliminated, the performance space and its acoustic qualities was made irrelevant, and the listener was given a unique perspective shared with no one. Over the past few years, this instrument was used to perform for numerous people. This paper explores and summarizes those adventures.

8:20

2aAAa2. Don't get caught with your mics open. Deborah J. Britton (K2 Audio, LLC 4900 Pearl East Circle, Ste. 201E, Boulder, CO 80301)

Designing sound systems for legislative facilities often presents numerous challenges. From creating intelligible sound reinforcement and broadcast feeds in highly ornate, reverberant spaces to using digital signal processing to ensure that "off-the-record" side conversations are not made public, many different elements contribute to the final design. This presentation discusses a few of these specific challenges and how they were overcome.

8:40

2aAAa3. Methods for separating harmonic instruments from a monaural mix. Mert Bay and James W. Beauchamp (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801, mertbay@illinois.edu)

During the past decade, solutions to the problem of musical sound source separation have become more evident. Potential applications include sound editing, enhanced spatialization, music-minus-one, karaoke, music classification/identification, music transcription, and computational musicology. Our current approach is to restrict the input signal to a mix of a limited number of instruments, each comprised of harmonic partials, with known F0 contours. (These can be obtained either by audio-to-midi alignment or multiple-F0 estimation.) Since harmonic frequencies of known F0s are easily predicted, binary mask separation is robust except for frequency regions where harmonics of different instruments collide. Three methods for repairing collisions are compared: (1) F0-informed non-negative matrix inversion using instrument spectral libraries; (2) least-squares estimate of collision frequencies based on a multiple sinusoidal model; and (3) common amplitude modulation (CAM) [Li *et al.*, IEEE Trans. Audio, Speech, Lang., Process. **17**(7), 2009]. Separation examples using these methods will be demonstrated.

9:00

2aAAa4. Would a rose sound as sweet. Sam Ortallono (MediaTech Inst., 3324 Walnut Bend Ln., Houston, TX 77042)

Would a rose sound as sweet? In this presentation, we will explore perception of individual microphones. The same vocalist will be recorded with five different microphones. Volunteers will be presented with two sets of tracks to evaluate. The first set of tracks will be labeled only with letters, microphone A, B, C, D, and E. The second set of tracks will be labeled with the names of the microphones. Then the subjects will be asked to rate subjective qualities of both sets. Each person will be their own control by judging one set blind and one with the knowledge of the name of the microphone. Will preconceived notions of a microphonex's reputation change the outcomes?

9:20

2aAAa5. Binaural room acoustics. William M. Hartmann, Brad Rakerd, and Ryan A. Cook (4208 BPS Bldg., Michigan State Univ., East Lansing, MI 48824)

Room acoustics concerns the geometrical and materials properties of rooms as they determine measurable sound field parameters that are perceptually important to listeners. For example, the volume and absorption in a room determine the reverberation time, which is important to the comprehension of speech and the enjoyment of music. Binaural room acoustics concerns the properties of rooms as they affect measurable interaural parameters, important to the human binaural perceptual system. For example, room properties affect the short-term interaural cross-correlation, which is important to the perception of apparent source width. Our recent work emphasizes room effects on steady-state interaural level and phase differences, important to sound localization in the horizontal plane. Particularly, we have sought to give mathematical meaning to a "binaural critical distance." If a sound source and a listener are separated by less than the binaural critical distance, there is good probability that the interaural differences correctly indicate whether the source is on the listener's left or right. Experimentally, we have focused on sine tones in the range 200–1200 Hz, but we expect the results to be more generally applicable. [Work supported by the AFOSR (grant 11NL002) and by the NSF REU program.]

9:40

2aAAa6. Why reverberation affects different individuals differently. Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., 677 Beacon St., Boston, MA 02215)

Any self-respecting architectural acoustician understands that good reflected sound enhances all sorts of important sound attributes. He or she undoubtedly also understands that reflected energy distorts the fine spectrotemporal structure of sound reaching a listener's ears. But what are the perceptual consequences of such minor acoustic distortions of sound detail, and what is the significance of this distortion when designing listening spaces? In psychophysics, recent experiments show that fine spectrotemporal structure is very important for understanding the content of complex, natural sounds, like speech, especially in the presence of background noise or competing sound. This talk will review why spectrotemporal structure is so important for verbal communication in everyday settings. Building from behavioral and physiological measures, this review will explore why even modest amounts of reverberant energy can dramatically impair the ability to communicate for some individuals, even when it causes no such ill effects in others. Understanding such effects is critical when designing listening spaces in which verbal communication is a key activity, like in a classroom.

10:00

2aAAa7. Verb, reverb, and re-reverb. Jim Anderson (Clive Davis Inst. of Recorded Music, New York Univ., 194 Mercer St., New York, NY 10012)

In a contemporary recording, artificial ambience, reverberation of varying length and degree, is added to compliment the various elements in a recording. This mixing condiment comes in many flavors: live chamber, plate, and digital. The application of reverberation is an art, as well as science, and its treatment can be a matter of taste, good, or bad. While it can be difficult to achieve a natural effect with one reverb in a recording, parallel reverbs can be used effectively. As a live performance takes place in a single reverberant hall, its recorded counterpart may take advantage of the application of different multiple reverbs.

10:20–10:35 Break

10:35

2aAAa8. Highly dynamic reverb—The recording studio always has what the performance hall sometimes wants. Alex Case (Sound Recording Tech., Univ. of Mass Lowell, 35 Wilder St., Lowell, MA 01854, alex_case@uml.edu)

The orchestra wishes to perform in a space whose acoustics offers flattering reverberation. That reverb is generally unchanging, from the beginning to the end of the performance. And that reverb must be "one size fits all," as every instrument in the band is offered the same acoustic treatment. Not so in recorded music. When reverb is generated acoustically, mechanically, and digitally and introduced into the music by way of a mixing console or digital audio workstation, under the watchful ear of the creative recording engineer, highly variable reverb becomes possible. Momentary reverb, dynamic reverb, multiple contrasting reverbs, and still more variations not easily realized in a physical space are found common in popular music recordings, demonstrated and discussed here.

10:55

2aAAa9. Stonehenge-like auditory illusion evoked by interference pattern. Steven J. Waller (Rock Art Acoust., 5415-8 Lake Murray Blvd., La Mesa, CA 91942, wallersj@yahoo.com)

Blindfolded participants hearing an interference pattern attributed the dead zones to the presence of acoustic obstructions in an arrangement reminiscent of Stonehenge. Without prior mention of interference or Stonehenge, participants were brought blindfolded to an open field with two English flutes atop thin 1 m tall wooden stands 2 m apart, connected via tubing to an air pump, giving a sustained pitch of C#5 (1 100 Hz). One by one, participants were led blindfolded in a 7.6 m radius circle around the pair of flutes. Afterward, faced away from the setup, they were instructed to remove blindfolds and independently "draw and describe what you thought was between you and the noise." Although there actually were no physical obstacles, the resulting drawings from all three participants represented arrangements of objects having many characteristics in common with Stonehenge; verbal descriptions included "pillars," "solid objects," "openings," "tall vertical slats." These pilot results demonstrate regions of low sound intensity due to destructive interference of sound waves from musical instruments can be misperceived as an auditory illusion of acoustic shadows cast by a ring of large obstructions. Measurements of acoustic shadows radiating out from Stonehenge are consistent with the hypothesis that interference patterns served as blueprints for the design.

Contributed Papers

11:15

2aAAa10. A new generation of time variant electro-acoustic enhancement systems. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

In 1991, Dr. David Griesinger presented a paper at the 90th AES convention [preprint 3014] describing a new method for improved electronic acoustic enhancement, and hundreds of these systems have been installed throughout the world. This paper describes a new approach to both hardware and software that comprises a third generation system. The new system retains the essential features of multichannel time variance with low pitch alteration, but updates the algorithms to utilize the superior DSP power of modern computers. Multiple reverberators, equalization, and matrix distribution can be combined into a single, reliable, and easily replaceable piece of hardware. Overall system gain and equalization can be calibrated automatically anytime; parameters in the hall have changed. The new algorithm combines the efficiency of infinite impulse response filters with direct convolution to produce a time-variant structure with extremely low coloration. The algorithm also includes a novel 32 band antifeedback circuit that further reduces coloration when the system is installed in a room. Unlike much of the competition, this system architecture allows the strength of the reverberation to be adjusted independently of the reverberation time, which makes it possible to simultaneously optimize both clarity and reverberance.

11:30

2aAAa11. Motion simulation in the environment for auditory research. Braxton B. Boren (Dept. of Music and Performing Arts, New York Univ., 34 W. 4th St., New York, NY 10012) and Mark A. Ericson (Army Res. Lab., Aberdeen Proving Ground, MD 21005)

Virtual sound source motion has been implemented in the Army Research Laboratory's Environment for Auditory Research, which contains a 57-channel spherical loudspeaker array located in an anechoic chamber. Using the Psychophysics Toolbox Version 3 low-latency PortAudio API, 57 channels of streaming audio are dynamically updated in real-time using MATLAB for signal processing. Both distance-based amplitude panning (DBAP) and vector base amplitude panning (VBAP) have been implemented in MATLAB for controlling source motion. Sources are defined on a given path, such as a circle, ellipse, or the "dog bonex"; flight pattern often used in aviation. While DBAP works convincingly for virtual sources located on the sphere defined by the loudspeaker array, VBAP is needed to position sources outside the array. Source motion paths are defined parametrically with respect to time, and playback buffers update the panned position every 11.6 ms. Based on the source's instantaneous distance, diffuse-field or free-field amplitude attenuation is added in MATLAB as well air absorption filtering. This system will be used for a variety of audio simulations and auditory experiments.

11:45

2aAAa12. Whales and poets—Two tales of techniques for adapting, enhancing, and fictionalizing acoustic traits. Carl P. Giegold, Kevin T. Watson, Molly K. Norris, and Scott D. Pfeiffer (Threshold Acoust. LLC, 53 West Jackson Blvd. Ste. 815, Chicago, IL 60604)

Enhancement techniques are employed in two vastly divergent venues in Chicago with the purpose of removing barriers to communication. In the first case, the marine environment scaled to house Beluga whales presents its own acoustical obstacle. In the second, the presence of a microphone can be an obstacle as seen by a poet with a message to share. The Poetry Foundation implemented a voice lift system in a sonorous environment that supports the voice or light amplification, but requires assistance to extend the reach of the unamplified talker. Natural acoustic design, design of traditional public address systems, and the introduction of enhancement techniques are all outlined for these two case studies to illustrate how the roles intersect. Both environments are live in their un-enhanced state. The unusual aspect of these projects is the use of enhancement to increase intelligibility within a naturally reverberant space.

TUESDAY MORNING, 1 NOVEMBER 2011 SUNSET, 8:00 A.M. TO 12:00 NOON

Session 2aAAb

Architectural Acoustics, Noise, and Signal Processing in Acoustics: Accuracy of Absorption, Scattering, and Diffusion Coefficients

Peter D'Antonio, Chair
RPG Diffusor Systems, Inc., 651C Commerce Dr., Upper Marlboro, MD 20774

Chair's Introduction—8:00

Invited Papers

8:05

2aAAb1. New developments in the measurement and calibration of absorption, scattering, and diffusion coefficients. Peter D'Antonio and Brian Rife (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

Round robin experiments have confirmed that current methods to measure the random incidence absorption coefficient are inaccurate and not reproducible from lab to lab. As a result, various improvements to ISO 354 are being considered. Among them include measures to prevent the coefficient from exceeding unity and achieve reproducibility, using calibration. Since all measurement equipment is calibrated, it seems prudent to calibrate the rev room as well. We will describe approaches used at RPG to replace suspended clouds with boundary diffusors, the use of Eyring instead of Sabine and calibration, using a non-diaphragmatic reflective sample, with the same area, thickness and perimeter as the absorber, as a "zero" or low absorption reference. The new rotating table ISO 17497-1 scattering coefficient apparatus, using dual source/simultaneous impulse response measurements will be described. Since the technique utilizes subtraction of the reflective table as a reference, some of the problems associated with the rev room method are mitigated and reproducibility is achieved. Finally, the new diffusion coefficient goniometer, utilizing 32 simultaneous impulse response measurements, will be described. As with the scattering coefficient, the normalized diffusion coefficient benefits from the use of a reflective calibration reference, thus removing edge diffraction and other effects.

8:25

2aAAb2. The sound absorption measurement according to ISO 354. Martijn Vercammen and Margriet Lautenbach (Peut, P.O. Box 66, Mook ZH-6585, Netherlands, m.vercammen@mook.peutz.nl)

Sound absorption measurements of building materials such as sound absorbing ceilings and other products are performed in a reverberation chamber according to ISO 354. It is known that the interlaboratory reproducibility of these measurements is not very well. At this moment, the differences of results between laboratories are much larger than can be accepted, from a practical point of view for predictions as well as from a jurisdictional point of view. An ISO working group has started to investigate possibilities to improve the method. Due to the insufficient diffuse sound field in a reverberation chamber with test sample, the shape of the reverberation room and the placing of diffusers will influence the result. A round robin research containing ten laboratories is performed to get information on the spread and if it is possible to reduce this by correcting for the mean free path or by application of a reference material. Additional measurements are performed to improve the measurement conditions such as measurements with volume diffusers. Possible improvements of ISO 354 will be presented. These consist of a procedure to qualify laboratories based on the statistical variation of the reverberation time and based on the results of a reference absorber.

8:45

2aAAb3. Diffusivity of diffusers in the reverberation room. Margriet Lautenbach (Peutz BV, Paletsingel 2, P.O. Box 696, 2700 AR Zoetermeer, m.lautenbach@zoetermeer.peutz.nl)

The random incidence absorption coefficient is measured in a reverberation room according to the ISO354 or ASTM C423-09. According to these standards, the diffusivity of a reverberation room is usually obtained with panel diffusers. Besides the fundamental problem that a reverberation room with a highly absorptive specimen is not diffuse, these panel diffusers introduce a number of uncertainties like the acoustical effective volume and the total boundary surface of the reverberation room. This might be one of the causes that some laboratories are structurally able to measure absorption coefficients larger than 1, even if the volume of the specimen, edge absorption, and the absorption of the surface covered by the specimen are taken into account. To reduce the difference in measurement results between different laboratories, the possible use of volume diffusers instead of panel diffusers is investigated. The following criteria are investigated to substantiate the hypothesis that volume diffusers lead to better results: (1) Deviation between microphone-source positions. (2) Comparison to maximum relative standard deviation (ASTM). (3) Comparison to theoretical variance. (4) Influence of place of specimen. The investigations have been performed in a 1:10 scale model. The results are presented in this paper.

9:05

2aAAb4. Measuring absorption: Bad methods and worse assumptions. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., SDP\#8, Elma, WA 98541, audio_ron@msn.com)

Ever since Sabine described the use of absorptive materials to affect reverberation time, the acoustic community has been trying to quantify this effect. Standards like C-423 and ISO-354 have been developed to aid in that process. These standards, describing coefficients ranging from 0.0 to 1.0, have been inadequate to fully describe the actual absorption of the tested materials. It is not uncommon to find actual measurements that result in coefficients exceeding the number 1.0. When this happens, assumptions are made that other properties of the material are not taken into account by the coefficient. The present standards also suffer from inadequacies of methodology. They do not measure the full range of absorptive qualities. This paper describes the incorrect assumptions about absorption and measuring it as well as illustrates the incorrect methodologies that are used in these standards. Because of the flaws in the assumptions and the inherent defects in the methodologies, this author believes the current standards need to be replaced by new standards. Replacement instead of correction is preferable because of the extremely large database of materials already measured and the difficulty explaining how new measurements based upon corrections would be comparable to old measurements.

9:25

2aAAb5. On the reproducibility of measuring random incidence sound absorption. Anthony Nash (Charles M. Salter Assoc., 130 Sutter St., San Francisco, CA 94104, anthony.nash@cmsalter.com)

For over 50 years, the American Society for Testing and Materials (ASTM International) has promulgated a method for the laboratory testing of random-incidence sound absorption coefficients (Test Method C423). This test method falls under the purview of ASTM Committee E33 (Environmental Acoustics). In 1999, the protocols in this test method became significantly more stringent with the goal of improving inter-laboratory "reproducibility" (i.e., quantitative differences among laboratories when testing the same specimen). ASTM calls such an inter-laboratory test a "round robin"; its outcome is an array of computed precision (i.e., uncertainty) values. This paper presents results from several "round robin" evaluations and discusses some possible causes for the range of values. If time permits, the fine points of the test protocols in C423 will be described and compared to those in ISO 354.

9:45

2aAAb6. Energy-based measurements in reverberation chambers. Timothy W. Leishman, Buye Xu, Scott D. Sommerfeldt, and Nicholas J. Eyring II (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, twleishman@byu.edu)

Standardized reverberation chamber measurements, including those of absorption coefficients, scattering coefficients, and sound power, rely on the acquisition and processing of squared acoustic pressure (or potential energy density) from many sound field locations. Kinetic energy density, total energy density, and the newly defined generalized energy density exhibit greater spatial uniformity in

enclosed fields (especially at lower frequencies), which makes them more favorable quantities for these measurements. They produce greater accuracy or require fewer source-receiver positions to produce acceptable levels of accuracy. This paper explores their use and provides several analytical and experimental results to demonstrate their benefits.

10:05–10:30 Break

10:30

2aAAb7. Error factors in the measurement of the scattering coefficient in full and small scales. Tetsuya Sakuma and Hyojin Lee (Grad. Sch. of Frontier Sci., Univ. of Tokyo, 5-1-5 Kashiwanoha, Kashiwa 277-8563, Japan, sakuma@k.u-tokyo.ac.jp)

The reverberation room method for measuring the random-incidence scattering coefficient was standardized by ISO 17497-1 in 2004, and now the amendment is being discussed on the specimen mounting and the turntable speed. Regarding the former point, destructive diffraction from an uneven perimeter of a specimen is an error factor to overestimate the scattering coefficient; however, it can be suppressed by setting a border around the turntable. Regarding the latter point, special attention is needed to the combination of the turntable speed and the signal period of MLS in the impulse response measurement. Basically, the turntable speed should be limited by an angular step of 3° to 6° for one signal, thus 60 to 120 signals are required for one revolution. In practice, in order to suppress the time variance, a shorter signal period is preferred as far as the measured reverberation time is guaranteed. However, a best choice of the signal period depends on reverberation rooms in full and small scales, additionally depending on frequency bands. Some data measured in full and small scales demonstrate the effect of the combination on measurement accuracy.

10:50

2aAAb8. Uncertainty factors in determining the random-incidence scattering coefficient. Markus Müller-Trapet and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr 50, D-52066 Aachen, Germany, mmt@akustik.rwth-aachen.de)

When determining the random-incidence scattering coefficient with measurements that have been carried out according to ISO 17497-1, several quantities from four different measurements have to be combined to finally produce the sought-after coefficient. These include the reverberation time and atmospheric conditions to account for air absorption. As in most measurement situations, one has to keep in mind that uncertainties have affected the measurements, so that an absolute statement about the scattering coefficient might not be possible. This work will give an insight as to how each of the mentioned measured parameters and their uncertainty will affect the evaluation of the scattering coefficient. The "Guide to the Expression of Uncertainty in Measurement" will be employed to investigate the effect of both systematic and random errors. Within the limits of applicability, intervals for the necessary accuracy of the measured parameters that lead to an acceptable uncertainty in the final result will be given.

11:10

2aAAb9. Diffusion and scattering: An improved method of measuring both at the same time. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd. SDP#8, Elma, WA 98541, audio_ron@msn.com)

ISO-17497-1 and ISO-17497-2 describe methods of measuring the properties of diffusers and scattering devices. The result of these measurements is a description of the quality of the diffuser and the quantity of energy scattered. This paper will describe an improved method of measurement that enables both properties being simultaneously measured using the same procedure. The hope is that this method will produce a simplified database mimicking the directivity measurements of sound sources for use in simulation programs and in comparing various diffuser designs.

Contributed Papers

11:30

2aAAb10. Predicting scattering at the mid-frequency range. Carl R. Hart and Siu-Kit Lau (Durham School of Architectural Eng. and Construction, Univ. of Nebraska—Lincoln, 1110 S. 67th St., Omaha, NE 68182, carl.hart@huskers.unl.edu)

Predicting scattering characteristics of isolated surfaces is problematic in the mid-frequency range, due to high computational costs. In the mid-frequency range, techniques such as the finite element method or the boundary element method require extremely fine mesh structures, leading to the solution of very large matrices. A gap exists for predicting acoustic scattering in the mid-frequency range with low computational costs. A geometric acoustic time domain method, incorporating diffraction, is proposed to bridge the gap in current prediction methods. The geometric acoustic method utilized is adaptive beam tracing. Adaptive beam tracing is advantageous, compared to other geometric methods, since it does not generate aliasing errors and faulty image sources, which are shortcomings of ray tracing and the image source method, respectively. The scattering characteristics of a periodic rigid geometry are investigated. Relationships between geometry dimensions and frequency of excitation are studied. Furthermore, the importance of accounting for single diffraction or multiple diffractions will be discussed. The spatial, temporal, and frequency characteristics of the scattering geometry are quantified.

11:45

2aAAb11. Repeatability and reproducibility concepts in acoustical laboratories. John LoVerde and Wayland Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

There are many categories of variability in acoustical laboratory testing. The repeatability of acoustical testing is usually defined based on repeated testing of the same assembly over a short time period. Other variability, such as rebuilding an assembly in the same lab or building a nominally identical assembly in a different lab, is generally termed reproducibility. A previously undocumented variability is what may be termed "long term repeatability," the repeated testing of the identical specimen in the same laboratory with the same equipment and operator, but over an extended time period. The authors recently had the opportunity to measure the long term repeatability of a wood joist floor/ceiling assembly that remained in the same test chamber at an accredited acoustical laboratory over 50 days. The results are compared with previously known variability, and the implications for manufacturers and designers are discussed.

Session 2aAB

Animal Bioacoustics: General Topics in Animal Bioacoustics I

John A. Hildebrand, Chair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

8:15

2aAB1. Bird's singing codification using quick response barcode. Hugo Fernando Jácome Andrade (Dept. of Sound & Acoust. Eng., Univ. of the Americas, E12-41 Colimes St. and Granados Ave., Quito, Pichincha, hjacome@udla.edu.ec) and David Parra Puente (Univ. Catholic of Ecuador, Quito, Pichincha)

This work responds to the necessity to objectively describe bird's songs in scientific publications and field guides. Nevertheless, it can also be extended to the study of other animal like marine mammals, bats, humans, and insects. The project consists on a computer application development in MAX/MSP language, that extracts coefficients from an audio signal, those coefficients will characterize the signal, and are codified by QR tool (Quick Response Barcode), created by the Japanese company Denso-Wave in 1994. The following stage of the project, that is in developing process, is the programming of the de-codifier application, that rebuild the audio signal from those numbers, to bring back the signal to the human hearing field, as final stage of the project, it will be developed a similar application with the same intentions for mobile phones with the built-in QR reading tool.

8:30

2aAB2. Correlation detection of bat echolocation calls. Mark D. Skowronski (Dept. Speech Lang. and Hearing Sci., Univ. of Florida, Gainesville, FL 32611, markskow@hotmail.com)

In an experiment comparing detection of echolocation calls between humans and automated methods, an optimal linear detector (matched filter) out-performed humans by 5 dB and model-based automated methods by 9 dB [Skowronski and Fenton, J. Acoust. Soc. Am. **125**, 513–521 (2009)]. While optimal linear performance cannot be achieved in practice, near-optimal performance may be reached if the detection filter nearly matches the target calls. Bat calls from a species are fairly stereotypic, so designing a test call (or bank of test calls) for a correlation detector is feasible. Model-based detectors collect information over short analysis windows and piece that information together to form a detection decision, a bottom-up strategy that is good at collecting local information but poor at arranging local information to form a global detection decision. By contrast, a correlation detector operates at one time scale, the global call duration, and may be considered a top-down strategy of detection. The strengths and weakness of a correlation detector are discussed, including designing filters for a given species, the tradeoff between filter bank size (computational cost) and detector sensitivity, species classification, and the effect of filter bank size on false positive rate.

8:45

2aAB3. Automated extraction and classification of contours in humpback whale vocalizations. Helen H. Ou (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, ohui@pdx.edu), Whitlow W. L. Au, Marc O. Lammers (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kaneohe, HI 96744), Adam A. Pack (Univ. of Hawaii at Hilo, Hilo, HI 96720), and Lisa M. Zurk (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR 97201)

Humpback whales produce songs which consist of a sequence of short, continuous sounds known as units. This paper introduces an automated algorithm to extract the unit contours. An unsupervised classification is developed to provide a set of distinct units of the singing group. The analysis is performed on the vocalization spectrograms, which are normalized and interpolated into a squared time-frequency image. Unit contours are detected using two edge detection filters capturing sharp changes in the image intensities. The algorithm generates a group of rectangular image segments each containing a single contour unit, with the pixels outside the contour edge lines set to zero. The contours are compared with one another to identify distinct units. The comparison is quantified using parameters including the contour pixel intensity correlation, contour area, frequency range, and frequency of the peak pixel. A pairwise comparison provides a coarse division of classes, where each class is then represented by a candidate unit. The candidate units are compared with one another, and the ones with low similarity are advanced to the final set. The algorithm has been tested on humpback whale songs obtained during the winter season in Hawaiian waters in 2002 and 2003.

9:00

2aAB4. Noise reduction for better detection of beaked whale clicks. Yang Lu, Holger Klinck, and David M. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, luya@onid.orst.edu)

We seek to improve the signal-to-noise (SNR) ratio of clicks recorded from Blainville's beaked whales (*Mesoplodon densirostris*). The proposed filter is based on subspace principles and projects the noisy speech vector onto signal and noise subspaces. An estimate of the clean signal is made by retaining only the components in the signal subspace. To test the filter, a detector using the filter output has been designed for the detection of beaked whale calls. Through simulations, the proposed detector is shown to be capable of detecting most of the desired clicks but is not able to differentiate other co-existing species such as Risso's dolphins and pilot whales' that is, it efficiently detects all clicks. By combining the proposed detector with the energy ratio mapping algorithm (ERMA; Klinck and Mellinger (in press), which measures energy differences between different species, higher detection accuracy for beaked whale clicks can be achieved. The filter can also be used to improve the SNR of other marine mammal acoustic signals.

9:15

2aAB5. Spatial orientation of different frequencies within the echolocation beam of a *Tursiops truncatus* and *Pseudorca crassidens*. Stuart Ibsen (Dept. of Bioengineering, Moores Cancer Ctr., Univ. of California San Diego, 3855 Health Sci. Dr. # 0815, La Jolla, CA 92093-0815), Paul Nachtigall (Univ. of Hawaii, Kailua, HI 96734), Jacqueline Krause-Nehring (Alfred Wegener Inst. for Polar and Marine Res., Am Handelshafen 12, 27570 Bremerhaven, Germany), Laura Kloepper, and Marlee Breese (Univ. of Hawaii, Kailua, HI 96734)

A 2D array of hydrophones was created to determine the spatial distribution of frequencies within the echolocation beam of a *Tursiops truncatus*. This *Tursiops* was shown previously to only pay attention to frequencies between 29 and 42 kHz while echolocating. It was found that the 30 kHz frequency was tightly focused, and the spatial location of the

focus was consistently pointed toward the target. At 50 kHz, there was less focusing and less precision in being pointed at the target. At 100 kHz, the focus was often completely lost and was not pointed at the target. This indicates that this dolphin was actively focusing the frequencies it paid attention to toward the target, while the frequencies not paid attention to were left unfocused and undirected. This focusing was probably achieved through morphological manipulations of the melon and nasal air sacs. This explains earlier observations of how the dolphin achieved consistent frequency content only in the 0–42 kHz range with simultaneous variability outside this range in echolocation clicks recorded with a single on-axis hydrophone. Similar results were observed for a *Pseudorca crassidens*, while performing similar target discrimination tasks. [Office of Naval Research Grant No 0014-08-1-1160 to P.E. Nachtigall supported this work.]

9:30

2aAB6. Depth and range tracking of sperm whales in the Gulf of Alaska using a two-element vertical array, satellite and bioacoustic tags. Delphine Mathias, Aaron Thode (Scripps Inst. of Oceanogr., Marine Physical Lab., 9500 Gilman Dr., La Jolla, CA 92037-0238 delphine.mathias@gmail.com), Jan Straley (Univ. of Alaska Southeast, Sitka, AK 99835), and Russel Andrews (School of Fisheries and Ocean Sci., Univ. of Alaska Fairbanks, AK 99775)

A two-element vertical array was deployed between August, 15 and 17 2010, on the continental slope off Southeast Alaska, in 1200 m water depth. The instruments were attached to a longline fishing anchorline, deployed at 300 m depth, close to the sound-speed minimum of the deep water profile. The anchorline also served as a decoy, attracting seven depredating sperm whales to the area. Three animals were tagged with a satellite tag and one of them was tagged with both a satellite and bioacoustic "BProbe" tag. Both tags recorded dive depth information. Relative arrival times of surface- and bottom-reflected paths are used to estimate animal range and depth on a single hydrophone, and compared with tagging results. The two-element array is then used to estimate vertical arrival angles of the direct and surface-reflected paths to determine whether range and depth localization can occur without the use of bottom multipath. This data will be useful in determining whether long-range tracking of sperm whales is possible using a single compact instrument deployment. Potential applications include observing what ranges whales are willing to travel to depredate. [Work conducted under the SEASWAP program, supported by the National Oceanic and Atmospheric Administration and the North Pacific Research Board.]

9:45

2aAB7. Matched-field processing and modal filtered range estimates of bowhead whale calls detected in the Alaskan Beaufort Sea. Aaron M Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238, athode@ucsd.edu)

In 2010, a 15-element autonomous vertical array was deployed roughly 35 km north of Kaktovik alongside a distributed array of directional autonomous seafloor acoustic recorders (DASARs). Matched-field processing and geoacoustic inversion techniques were used to extract the range, depth, bottom sound speed profile, density, and attenuation from one close-range whale call. The inversion localized the call at 1.2 km range and 44 m depth in 55 m deep water. The inverted propagation model was used to derive the group and phase velocities of the normal modes in the region surrounding the array. The vertical array spanned sufficient aperture in the water column to permit isolation of the first and second mode arrivals from any given call. A range- and frequency-dependent phase shift was applied to each modal arrival to remove geometric dispersion effects. The modeled range that time-aligned the modal arrivals was selected as the range estimate. The modal filtering technique is demonstrated on additional whale calls produced at 17.3 km and 35 km range from the vertical array, with the range estimates independently confirmed by triangulating bearings of call detections on surrounding DASARs. [Work supported by the North Pacific Research Board, Shell Exploration and Production Company, and Greeneridge Sciences Incorporated.]

10:00

2aAB8. Bearing estimation and 2D localization of bowhead whale calls using directional autonomous seafloor acoustic recorders. Gerald L. D'Spain, Heidi A. Batchelor, Simon E. Freeman (Marine Physical Lab, Scripps Inst. Oceano., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu), Katherine H. Kim, Charles R. Greene, Jr., Susanna B. Blackwell (Greeneridge Sci., Inc., Santa Barbara, CA), and A. Michael Macrander (Shell Exploration and Production Co., Houston, TX 77079)

Passive acoustic monitoring of the summer/fall westerly bowhead whale migration in the Beaufort Sea has been conducted by Greeneridge Sciences, sponsored by SEPCO, every year since 2006. The directional autonomous seafloor acoustic recorder (DASAR) packages used in this effort each contain three acoustic sensors that simultaneously measure the two horizontal components of acoustic particle motion and acoustic pressure. A variety of data-adaptive beamforming methods have been applied to a selected subset of these data to examine direction-of-arrival estimation performance, including quantifying bearing bias and variance. A principal component analysis (PCA)-type eigenanalysis of the sensor data cross spectral matrix is used to decompose the received field into orthogonal components having different particle motion polarization and energy flux properties. Appropriate manipulation of these components provides high resolution directional estimates of the primary arriving energy while maintaining robustness to uncertainties in sensor calibration and acoustic propagation conditions. Representative results of beamforming as well as 2D localization performance will be presented. Application of data-adaptive beamforming techniques to bulk processing of these large monitoring data sets will be discussed. [Work supported by Shell Exploration and Production Company (SEPCO).]

10:15–10:30 Break

10:30

2aAB9. A chimpanzee responds best to sine-wave speech with formant tones 1 and 2 present. Lisa A. Heimbauer, Michael J. Beran, and Michael J. Owren (Dept. of Psych., The Lang. Res. Ctr., Georgia State Univ., P. O. Box 5010, Atlanta, GA 30302)

A seminal study by Remez and colleagues [R. E. Remez, *et al*., Science, **212**, 947–949 (1981)] demonstrated that listeners were more successful in identifying sine-wave speech when the first two (T12) or all three formant tones (T123) were present than when either were absent (forms T13 and T23). To determine whether a language-trained chimpanzee (Panzee) with the ability to recognize English words in sine-wave form [L. A. Heimbauer, *et al*., Curr. Biol., doi:10.1016/j.cub.2011.06.007 (2011)] would respond similarly, she and 13 humans were tested with words synthesized in the four forms used by Remez *et al*. Indeed, for each species, perception of speech was significantly better when the first and second tones were both present. Panzee's performance suggests that she is attending to the same spectro-temporal features of sine-wave speech that are critical to humans. The outcomes further indicate that basic capabilities involved in speech perception could have been present in latent form in the common ancestor of humans and chimpanzees. [Work supported by NICHD.]

10:45

2aAB10. Structure of gray whale calls of San Ignacio Lagoon and their distribution in the water column. Anaid Lpez-Urban, Aaron Thode, Carmen Bazúa-Durán, Melania Guerra, and Jorge Urbán-Ramírez (Univ de Baja California Sur, Carretera al Sur Km 5.5, La Paz, Baja California Sur, Mexico)

During the winter gray whales congregate in San Ignacio Lagoon, Baja California, in order to breed and give birth. The Lagoon population can be roughly divided into two demographic groups: mothers with calves, and single animals. Here the acoustic behavior of gray whales wintering in the lagoon is studied by using bioacoustics tags to determine potential relationships between call type and call structure, relative calling frequency, and position in the water column among demographic groups. Between 2008 and 2010, 24 tags (Bio-Probe) recording tags were attached to gray whales in San Ignacio Lagoon. From 1591 minutes of recordings, 1250 calls were identified and classified into five call types: conga (S1), quejid, purr, croac, and ronroneo. The last call has not been previously reported for this species.

Conga calls were the most common type (88% of calls recorded) and was produced by both demographic groups. Ronroneo calls (5%) were mainly produced by single whales. Differences in call parameters (Call duration, minimum, maximum, and low-maximum frequency, number of pulses, and number of harmonics) were only determined for conga calls, for which statistically significant differences were found between demographic groups. Different call types tend to be produced at different depths.

11:00

2aAB11. N3 call types produced long-term by a killer whale of the northern resident community under controlled conditions: Characteristics, variation, and behavioral context. Juliette S. Nash (Dept. of Mar. & Env. Sci., Univ. of San Diego, 5998 Alcala Park, San Diego, CA 92110, julietten-11@sandiego.edu) and Ann E. Bowles (Hubbs - SeaWorld Res. Inst., San Diego, CA 92109)

A-Clan of the Northern Resident killer whale (*Orcinus orca*) community produces a dialect of discrete stereotyped vocalizations, including N3, a call found in all pods of the clan. Ford (1984) reported that N3 is produced almost exclusively in low-arousal states. A 45-yr old female (BC-F1) collected from the A23 matriline of A5 pod and held under controlled conditions has produced this call throughout her adult life. Her calls were collected in 1985 at Marineland of the Pacific (14.25 h of samples) and every few years from 1987 to 2010 at Sea World San Diego (252 h). Analysis of the data show that BC-F1 uses two stereotyped variations of the N3 call. While temporally and structurally similar, there is a positive and negative frequency inflection distinguishing them. BC-F1 delivers N3 with the positive-inflected call preceding the negative. The N3 call occurs in low-arousal states, but also in high-arousal states, including caller interruption, breaking-up of synchronous swimming, and percussive behavior. N3 is infrequently produced during affiliative synchronous swimming and call-matching bouts. Thus, the call is not simply an indication of the resting state but transitions from one state to another. [Research supported by the author's organizations with in-kind support from SeaWorld San Diego.]

11:15

2aAB12. Scalable distributed ultrasonic microphone array. Tórur Andreassen, Annemarie Surlykke (Dept. of Biology, Univ. of S. Denmark, Campusvej 55, 5230 Odense M, Denmark, thor@biology.sdu.dk), and John Hallam (Maersk Mc-Kinney Moeller Inst., Univ. of S. Denmark, 5230 Odense M, Denmark)

A modular approach to recording airborne ultrasound will be presented, the solution is based as much as possible, on retail components and open source software. The modular design, where each module has approx. 4 microphones, allows modules to be combined to extend the coverage area or get higher recording resolution. The system has been designed to have no inherent scalability limits, i.e., only limited by data storage available and the number of microphones you can get a hold of. Current implementation has been used for recording in field experiments with bats, both short-term (minutes) on BCI Panama and for long-term recording (months) in Denmark. The generic nature of the design and implementation allows us to easily replace old components with new technology as it becomes available, both hardware and software. This means we reap the benefits achieved by the electronics industry and also gain any software improvements/bugfixes by a simple Internet upgrade of the module software. Modular design introduces some extra complexity, but this is only an issue when real-time processing the data, or with regards to synchronization. Although said complexity also has advantages, one of them being the possibility of parallel processing.

11:30

2aAB13. Bat recording under controlled conditions: A replicable chamber for comparable results. Eduardo Romero Vivas, Patricia Cortés Calva, and Braulio León López (Centro de Investigaciones Biológicas del Noroeste, S.C., CIBNOR, Mar Bermejo 195 Col. Playa Palo de Santa Rita, La Paz, B.C.S. 23090, Mexico, evivas@cibnor.mx)

In ecology, bats have become a major subject of study. Some pollinate and disperse the seeds of many tropical plants; some help to control insect populations; a few affect livestock by sucking blood, but in all cases bats are indispensable links in ecosystems. Bats use of sound waves make echolocation calls monitoring a powerful tool for distribution, census, and present studies. This technique depends on having a reliable database of sounds for species identification purposes. Although there are several databases of echolocation sounds; the signal emitted by a bat depends on environmental conditions (vegetation, weather conditions), biotic factors (prey size, movement, defensive measures), and the specific task (seek, flee, pursue, evade, wandering, obstacle avoiding). This variability makes comparisons among databases difficult. A portable chamber made of common materials for *in situ* recording under controlled conditions is presented. The use of the chamber is proposed as a tool which might help to generate databases that could be reliable compared. Nine species of desert bats from the northwest region of Mexico were recorder using the chamber under controlled conditions and compared with field recordings. The performance of the chamber and the utility of the database generated for filed identification are presented.

11:45

2aAB14. Development of dolphin-speaker. Yuka Mishima, Keiichi Uchida, Kazuo Amakasu, Yoshinori Miyamoto (Tokyo Univ. of Marine Sci. and Technol., 4-5-7, Konan, Minato-ku, Tokyo 1088477, Japan), and Toyoki Sasakura (Fusion Incorporation, 1-1-1-806, Daiba, Minato-ku, Tokyo 1350091, Japan, sasakura@fusion-jp.biz)

Dolphin speaks broadband sound of ten octave ranges. The vocalization of dolphin sound is categorized into three types, whistle, burst pulse, and echolocation clicks. The frequency range of whistle is below 20 kHz, while that of echolocation clicks and burst pulse is from about several kHz to 150 kHz. Whistle and burst pulse are used social communication. Whistle is used identifying each other and burst pulse may contain emotive factor. Few burst pulse researches are conducted compared to whistle. Until now, burst pulse researches are mainly collecting the dolphin sound using hydrophone but not speaking their sound from human side. The main reason why is no speaking from human side is no speaker of broadband frequency range for burst pulse. The newly developed Dolphin-Speaker has the broadband frequency response from 15 to 150 kHz. The Dolphin-Speaker is developed by advanced technology using multilayer piezoelectric device. We introduce the frequency characteristics of Dolphin-Speaker and the spoken sound by Dolphin-Speaker. We compared the original burst pulse of dolphin and the playback burst pulse by the Dolphin-Speaker. In near future, we would try to playback the burst pulse to dolphin and observe the behavior of the dolphin.

Session 2aEA

Engineering Acoustics and Structural Acoustics and Vibration: Periodic Structures

Andrew J. Hull, Chair
Naval Undersea Warfare Center, Code 8212, Newport, RI 02841

Invited Papers

8:40

2aEA1. Periodic metal structures for acoustic wave control. Andrew Norris (Dept. of Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854)

Passive control of acoustic waves, such as steering energy around an object (acoustic cloaking), requires non-standard materials. Either the density or the bulk modulus or both must be anisotropic. This explains why the phenomenon is not observed in nature. The focus of the talk is the design of metamaterials that have the desired anisotropic stiffness, with isotropic density. This is achieved by periodic metal lattices with sub wavelength cell size. The simplest example is "Metal Water" which is a foam-like periodic structure with the wave speed and density of water, and low shear modulus. The small shear is important in order to get the material to behave acoustically like water but is large enough to provide structural stability. When "deformed" the structure becomes anisotropic with the desired wave steering properties. The simple idea behind this class of acoustical metamaterial has simple consequence in design of cloaking devices, including the concept of conservation of cloaking space. These ideas will be explained and illustrated through scattering simulations.

9:00

2aEA2. Acoustic behavior of magnetorheological fluids in magnetic fields. Thomas R. Howarth, Frank Fratantonio, Jeffrey E. Boisvert, Anthony Bruno (NAVSEA Div. Newport, 1176 Howell St., Newport, RI 02841-1703), Clyde L. Scandrett (Naval Postgrad. School, Monterey, CA), William M. Wynn, and Philip S. Davis (NAVSEA Div. Panama City, Panama City Beach, FL)

Acoustic metamaterials are being considered for periodic structures where specific microscopic material properties can be tailored to alter macroscopic acoustic fields. One type of acoustic metamaterial being considered is an active fluid known as magnetorheological (MR) fluids. MR fluids contain magnetic particles dispersed within a host fluid where its viscoelastic behavior is controllable by varying the magnetic field intensity. A series of acoustic experiments has recently been conducted at the National High Magnetic Field Laboratory in Tallahassee, Florida. The acoustic sound speed of MR fluids was measured as functions of applied magnetic field strength, normal and orthogonal field orientations, and acoustic frequency. This presentation will discuss MR fluids, measurement methodology, and preliminary results. [Work supported by NAVSEA Division Newport ILIR.]

9:20

2aEA3. Three-dimensional acoustic scattering by layered media: A novel surface formulation with operator expansions implementation. David Nicholls (Dept. of Math, Stat, and CS, Univ. of Illinois-Chicago, 851 S. Morgan St., Chicago, IL 60607, nicholls@math.uic.edu)

The scattering of acoustic waves by irregular structures plays an important role in a wide range of problems of scientific and technological interest. This talk focuses upon the rapid and highly accurate numerical approximation of solutions of Helmholtz equations coupled across irregular periodic interfaces meant to model acoustic waves incident upon a multi-layered medium. A novel surface formulation for the problem is described in terms of boundary integral operators (Dirichlet–Neumann operators), and a boundary perturbation methodology (the method of operator expansions) is proposed for its numerical simulation. The method requires only the discretization of the layer interfaces (so that the number of unknowns is an order of magnitude smaller than volumetric approaches), while it avoids not only the need for specialized quadrature rules but also the dense linear systems characteristic of boundary integral/element methods. The approach is a generalization to multiple layers of Malcolm & Nicholls' Operator Expansions algorithm for dielectric structures with two layers. As with this precursor, this approach is efficient and spectrally accurate.

9:40

2aEA4. Level repulsion states and cavity modes excited by evanescent waves in sonic crystals waveguides. Vicent Romero García (Instituto Universitario para la Gestión Integrada de zonas Costeras, Universidad Politécnica de Valencia, Gandia 46730, Spain), Luis Miguel Garcia-Raffi (Instituto Universitario de Matemática Pura y Aplicada, Universidad Politécnica de Valencia, Spain), Jérôme Vasseur, Anne Christine Hladky-Hennion (Inst. d'Électronique, Microéléctronique and Nanotechnologie, U.M.R. CNRS 8520, France), and Juan Vicente Sanchez-Prez (Centro de Tecnologías Fsicas: Acstica, Materiales y Astrofísica, Universidad Politécnica de Valencia, Spain)

The relevance of the evanescent modes in sonic crystals is theoretically and experimentally reported in this work. The complex bands structure, $k(\omega)$, calculated using the extended plane wave expansion reveals the presence of evanescent modes in these systems, never predicted by the traditional usual numerical $\omega(\vec{k})$ methods. The interpretation of the evanescent modes introduces novel

explanations of the deaf bands as well as of the level repulsion states in antisymmetric periodic systems. In this work we observe that in the ranges of frequencies, where a deaf band is traditionally predicted, an evanescent mode with the excitable symmetry appears changing drastically the transmission properties. On the other hand, the simplicity of the sonic crystals in which only the longitudinal polarization can be excited is used here to interpret, without loss of generality, the level repulsion between symmetric and antisymmetric bands in sonic crystals as the presence of an evanescent mode connecting both repelled bands. These evanescent modes explain both the attenuation produced in this range of frequencies and the transfer of symmetry from one band to the other. The experimental evidence of the level repulsion and the evanescent coupling are in very good agreement with the theoretical predictions.

10:00–10:20 Break

10:20

2aEA5. Elastic response of a cylinder containing longitudinal stiffeners. Andrew J. Hull (Autonomous and Defensive Systems Dept., Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, andrew.hull@navy.mil)

This talk develops a 3-D analytical model of a cylinder that contains a longitudinal stiffener. The model begins with the equations of motion for a fully elastic solid that produces displacement fields with unknown wave propagation coefficients. These are inserted into stress and displacement equations at the cylinder boundaries and at the location of the stiffener. Orthogonalization of these equations produces an infinite number of indexed algebraic equations that can be truncated and incorporated into a global matrix equation. Solving this equation yields the solution to the wave propagation coefficients and allows the systems' displacements and stresses to be calculated. The model is verified by comparison of the results of a plane strain analysis example to a solution generated using finite element theory. A 3-D example problem is formulated, and the displacement results are illustrated. The inclusion of multiple stiffeners is discussed.

10:40

2aEA6. Computationally efficient finite element method for predicting wave propagation in periodic structures. Vincent Cotoni, Phil Shorter (ESI Group. 12555 High Bluff Dr., San Diego, CA 92130), and Julio Cordioli (Federal Univ. of Santa Catarina, 88040-970 Florianopolis SC, Brazil)

A generic numerical method for predicting the wave propagation in structures with two-dimensional periodicity is presented. The method is based on a combination of Finite Elements and periodic structure theory. A unit cell of the periodic structure is described with finite elements, and a Craig-Bampton reduction is applied to reduce the number of degrees of freedom. Periodic boundary conditions are then applied and the waves propagating in the structure are obtained by solving an algebraic eigenvalue problem. A number of analytical expressions are then used to derive the vibro-acoustic properties of the finite or infinite periodic structure. The method was recently extended to account for heavy fluid loading and material with frequency-dependent properties (typical of acoustic treatments). A number of examples are presented to validate the formulation and demonstrate the possible use of the method for design.

Contributed Papers

11:00

2aEA7. Reflection reduction by three-dimensional and two-dimensional phononic crystal slabs. Sven M. Ivansson (Swedish Defence Res. Agency, 16490 Stockholm, Sweden, sven.ivansson@foi.se)

A thin rubber coating with scatterer inclusions in a periodic lattice can redistribute sound energy, normally incident on a steel plate in water, in the lateral direction. The scattered energy can be absorbed by the rubber material and the reflection amplitude in the water can be reduced significantly. Coatings with different scatterer material types are here compared: air-filled cavities, high-density inclusions, and high-density inclusions coated by soft silicone rubber (which have attracted much interest in recent phononic crystal research). For each material type, scatterers of spherical (in a doubly periodic lattice) or cylindrical (in a lattice with a single period) shape are considered. Each coating type is optimized by differential evolution, varying a number of material and geometrical parameters to minimize the maximum reflectance within a certain frequency band. The layer multiple-scattering method is used as forward model. Good broad-band reflectance reduction is achieved with cavities (monopole scattering), but even better results are obtained with the coated high-density inclusions (dipole scattering). Combined with mixing scatterers of different sizes, the cylindrical shape, with scatterers in a lattice with a single period, is very powerful. The sensitivity of the performance to different parameters, as well as the incidence angle, is illustrated.

11:15

2aEA8. The appearance and use of Bragg scattering effects when sound is perpendicularly incident on a periodic structure. Jingfei Liu and Nico Declercq (Lab. for Ultrasonic Nondestruct. Eval., Georgia Tech Lorraine, 2 rue Marconi, Metz 57070, France)

In the study of diffraction spectra for sound incident on periodic structures, it has often been thought that different diffraction orders can be observed and studied separately. This assumption was probably a result of geometrical considerations where a bounded beam, incident and diffracted, is considered as a straight line. It has been widely used to study different phenomena such as surface wave generation and its physical relation to Wood anomalies. In this paper, we develop a geometrical study based on the finite beam width of transducer. The study reveals that measurement of the zero order reflected beam is always accompanied by the back-scattered beams of higher (Bragg) diffraction orders. Extensive measurements have been performed that precisely show the appearance of such waves in the diffraction spectra. Furthermore, the study reveals how the presence of such waves can be used for nondestructive characterization of corrugated surfaces, a technique much easier to interpret than earlier techniques based on Wood anomalies physically caused by surface wave generation. The most important feature is that the introduced technique can be applied for normal incidence and not necessarily in situations where surface waves can be physically generated. [Thanks to the French Centre National de Recherche Scientifique (CNRS)and the Conseil Rgional de Lorraine (CRL) for their financial support.]

11:30

2aEA9. Influence of periodicity on the reflection properties of finite periodic media. Alejandro Cebrecos, Vicent Romero-García, Rubén Picó, Vctor Snchez-Morcillo (Instituto Universitario para la Gestión Integrada de zonas Costeras, Universidad Politécnica de Valencia, Gandia 46730, Spain), Luis Miguel Garcia-Raffi (Instituto Universitario de Matemática Pura y Aplicada, Universidad Politécnica de Valencia, Spain), Juan Vicente Snchez-Prez (Universidad Politécnica de Valencia, Spain), and Kestutis Staliunas (Universitat Politécnica de Catalunya, Barcelona, Spain)

The dispersion relation of periodic systems, the well-known bands structure, gives the information about the propagating modes inside the media. These bands reveal both propagating and non-propagating ranges of frequencies. It is well known that the transmission properties, with relation to the propagating modes, can be characterized by both the bands structure and the equifrequency contours. Oppositely, in finite systems, one should consider the reflection properties on the interface defined by the host and the periodic media in order to know the spatial distribution of the reflected field. In this work, we analyze the reflected acoustic field by a periodic system. An experimental set up is proposed in this work to analyze the reflection properties of a periodic media. From the exploitation of the properties of both the interface and the inner periodicity of the crystal, fundamental and applied questions can be discussed using this media.

TUESDAY MORNING, 1 NOVEMBER 2011 TOWNE/ESQUIRE, 7:55 A.M. TO 12:00 NOON

Session 2aEDa

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

Kent L. Gee, Cochair
Dept. of Physics and Astronomy, Brigham Young Univ., N243 Eyring Science Center, Provo, UT 84602

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

Chair's Introduction—7:55

Invited Papers

8:00

2aEDa1. An advanced version of the vibrating string lab. Andrew C. Morrison (Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, achmorrison@gmail.com)

The vibrating string laboratory is a classic of undergraduate physics courses all over. At the introductory level, a common way of setting up a vibrating string lab is with a mechanical shaker and a non-magnetic string. The analysis usually done in the introductory laboratory usually assumes the vibrating string is a linear system. Real vibrating strings are non-linear and are an appropriate choice for exploring non-linear systems in a lab beyond the introductory level. Variations on the introductory lab and pedagogic approaches which make the lab appropriate for advanced undergraduate labs will be presented.

8:20

2aEDa2. Three approaches to understanding sound radiation from a tuning fork. Daniel O. Ludwigsen (Phys. Dept., Kettering Univ., 1700 University Ave., Flint, MI 48504, dludwigs@kettering.edu) and Daniel A. Russell (The Penn State Univ., University Park, PA 16802)

The simple tuning fork has a remarkable pattern of radiated sound. Measuring and modeling this radiation is the ultimate objective of a module designed for our senior-level laboratory course in acoustics. The primary audience for this course is composed of majors in applied physics and engineering physics, as well as engineering students interested in the acoustics minor, mostly electrical and mechanical engineering majors. To complement the blend of students, the course features a careful mix of theory and analytical modeling, computational modeling, and testing. These approaches run through a series of activities emphasizing practical knowledge, including calibrating microphones, validating a finite element model, measuring sound pressure level to produce polar plots of directivity, and using monopoles, dipoles, and quadrupoles to model and predict behavior of several common sources of sound. With these tools, students examine the sound radiated from a tuning fork driven near its lowest resonance frequency. They collect data to create far field and near field directivity plots and use an intensity probe to map 2-D vector intensity in the plane perpendicular to the tines. These are compared with a quadrupole model, as well as the results of a finite element model of their own creation.

8:40

2aEDa3. High speed video of commonly studied oscillating systems. Edward J. Tucholski (Phys. Dept., US Naval Acad., 572C Holloway Rd., Annapolis, MD 21402)

To allow student visualization of vibrating systems, laboratories and demonstrations have been cleverly designed to include stroboscopes, sand, and cork dust. Sensors in the laboratory are often single point type and must be moved to fully visualize the pattern of a vibrating system. Animations fill in the blanks but are still not real. Standard video recording would be the perfect data gathering and visualization tool except the frame rate is too slow for the most commonly studied oscillating systems. Reasonably priced high speed video cameras provide an alternative. In this paper, simple one and two dimensional vibrating systems (e.g., strings, bars, and membranes) are examined with a camera capable of frame rates up to 16 000 frames per second. Video analysis software is used to strip data from the recordings. More complex oscillating systems are also examined to demonstrate the power of this laboratory tool.

9:00

2aEDa4. Optical imaging of bubbles. Tom Matula (APL, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@apl.washington.edu)

Slow-motion imaging provides a useful way to image various phenomena, from ballistics to biomechanics to fluid dynamics. Such images lead to discoveries and better understanding of concepts. Our lab uses optical imaging to visualize bubbles under various conditions. The results of some of these images have resulted in paradigm shifts. For example, the famous “Crum bubble” showed clearly the formation and propagation of a liquid jet through a bubble. That image has led to numerous publications attempting to model jetting, and the liquid jet concept is now used extensively to explain various phenomena, including pitting of hardened materials and the destruction of kidney stones. In this paper, I will discuss typical imaging setups and provide examples of imaging bubbles under different conditions. Topics will include high-speed imaging, strobe imaging, and shock wave imaging. Phenomena will include sonoluminescence, snapping shrimp, and ultrasound contrast agents. [Work is supported by NIH NIBIB.]

9:20

2aEDa5. Low cost sound level meters for education and outreach. Ralph T. Muehleisen (Decision and Information Sci. Div. Argonne Natl. Lab., Argonne, IL 60439, rmuehleisen@anl.gov) and Andrew C. Morrison (Joliet Junior College, Joliet, IL 60431)

It used to be that when you wanted a low cost sound level meter for teaching and outreach, you could purchase a Radio Shack 33-2050 analog sound level meter. These meters were fairly accurate as well as rugged and could be used as a microphone with preamp in a pinch. Now that the Radio Shack meter is no longer for sale what other options are available? In recent years a plethora of meters and software apps have become readily available at a low cost. In this presentation, the authors will discuss some of the qualities one should look for in a low cost meter to be used for outreach and education and suggest some equipment that meets these needs.

9:40

2aEDa6. Tablet tools for teaching acoustics. Benjamin M. Faber (Faber Acoust., LLC, 654 Stonebrook Ln., Santaquin, UT 84655, ben@faberacoustical.com)

Recent advances in tablet computing technology have in some ways made tablet devices, such as the iPad, an attractive and viable alternative to the traditional notebook computer in and out of the classroom. Tablets are not only smaller and lighter than notebook computers, but typically employ capacitive touchscreen technology, which enables an unprecedented level of interactivity between user and device. The media-centric nature of the current crop of mobile devices also makes them potentially useful as teaching aids, both for classroom demonstration as well as for hands-on experimentation. The tablet’s utility is further enhanced by wireless and/or mobile Internet connectivity, as well as an increasing amount of available third party software. Potential uses of a tablet computer in teaching acoustics will be discussed.

10:00–10:20 Break

10:20

2aEDa7. Application of active-learning techniques to enhance student-based learning objectives. Tracianne B. Neilsen and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602)

Research in physics education has indicated that the traditional lecture-style class is not the most efficient way to teach science courses at the university level. Current best-teaching practices focus on creating an active-learning environment and emphasize the role of the students in the learning process. Several of the recommended techniques have recently been applied to Brigham Young University’s acoustics courses. Adjustments have been built on a foundation of establishing student-based learning outcomes and attempting to align these objectives with assessments and course activities. Improvements have been made to nearly every aspect of the courses including use of class time, assessment materials, and time the students spend out of the classroom. The progress made in bringing two of the courses, specifically an introductory, descriptive acoustics course for a general audience and a junior level introduction to acoustics course for majors, is described. Many of the principles can be similarly applied to acoustics education at other academic levels. Suggestions are made for those seeking to modernize courses at their institutions.

10:40

2aEDa8. Structural vibration studies using finite element analysis. Uwe J. Hansen (Dept. of Chemistry & Phys., Indiana State Univ., Terre Haute, IN 47809)

Bending wave propagation on a two dimensional structure is usually governed by a partial differential equation the solutions of which represent traveling waves. Imposing boundary conditions generally limits the solutions to standing waves representing the normal modes of vibration of the structure. These deflection shapes can be studied experimentally using holographic interferometry or computer aided modal analysis. Finite element analysis (FEA) provides a tool for calculating such normal modes. Input parameters include geometric variables of the structure along with elastic constants, boundary conditions, and a mesh grid which serves as a calculation basis for the iterative computer calculations. Full functioned FEA programs are generally beyond the means of most University educational laboratories. Among more accessible programs with somewhat limited capabilities is ANSYSED. The bulk of the time allotted for this paper will be spent to demonstrate the operation of this program by doing a detailed calculation of normal modes of a square plate clamped on one edge and also illustrating calculated results for a hand-bell.

11:00

2aEDa9. Using COMSOL multiphysics software to investigate advanced acoustic problems. Andrew A. Piacsek (Dept. of Phys., Central Wash. Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu) and Ralph T. Muehleisen (Argonne Natl. Lab, Argonne, IL 60439)

Numerical simulations provide a valuable tool for students to investigate complicated behavior found in many applications of acoustics. Computational "experiments" can be conducted quickly for a large number of parameter values, enabling students to visualize abstract quantities and to grasp causal relationships not easily recognized when looking at equations. The COMSOL finite-element multiphysics software package provides an integrated workspace in which the user defines a problem, meshes the geometry, and plots the solution(s). A brief overview of COMSOL will be presented, along with three examples of how it can be used to model advanced acoustics problems often encountered by students. The example problems involve musical acoustics, fluid-loaded shell vibrations, and flow resistance in porous materials. Also discussed will be the educational benefit of examining how choices made setting up the model can affect the integrity of the solution.

11:20

2aEDa10. Computer modeling in graduate level underwater acoustics. Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, misakson@arlut.utexas.edu)

The more a student can become part of the teaching process, the more engaged he is. In a laboratory setting, this involves physically implementing the learned theories. However, for upper level graduate courses, laboratory experiments are often unrealistic. For example, in an underwater acoustics class, it is impossible for a student to physically realize ray bending in the ocean without a large-scale experiment. However, computer modeling can bridge the gap. In this example, students write their own simulation of underwater propagation based on concepts learned in the lecture. Through the computer simulation, students manipulate the environment to determine the effects on acoustic propagation. As the course progresses, interface and target scattering and array and signal processing are added to the propagation code, leaving the student with a comprehensive ocean acoustics modeling toolkit.

11:40

2aEDa11. Measurements of electrical transfer characteristics with soundcards as classroom activity. Stephan Paul (Undergrad. Prog. Acous. Eng., Federal Univ. of Santa Maria, DECC, CT, UFSM, Av. Roraima 1000, Camobi, 97105-900 RS, Santa Maria, stephan.paul@eac.ufsm.br) and Pascal Dietrich (RWTH Aachen Univ., Neustrasse 50, 52066 Aachen, Germany)

Measurements of transfer functions and impulse responses of different types of systems are an important part in acoustics and should be an integral part in acoustics education. Nevertheless, usually expensive equipment is required to carry out such measurements making it difficult to realize hands-on measurement classes. Contrarily the transfer characteristics of electrical systems can be measured much easier. Electrical systems and acoustics meet in electro-acoustic equipment such as soundcards, which can be found as onboard and external devices and in a wide variety. Input and output of soundcards can be connected by different types of transmission systems that might use different types of electrical circuits adding a variety of transfer characteristics. The inherent transfer characteristics of the given soundcard can be analyzed and compensated before assessing the transfer characteristics of different electrical circuits (black boxes). Thus, the experimental measurement of soundcards together with different black-boxes connected to them can be a good means to perform comprehensive measurements of IR and FRF. Using a MATLAB based measurement software, a measurement set-up has been developed to be used by students in a classroom experience. This contribution discusses the set-up, the results of the different scenarios measured and the feedback of the students.

TUESDAY MORNING, 1 NOVEMBER 2011 SAN DIEGO, 10:30 A.M. TO 1:00 P.M.

Session 2aEDb

Education in Acoustics: Halloween Hands-On Acoustics Demonstrations for Middle-School Students

Joseph R. Gladden, Cochair
Dept. of Physics, National Center for Physical Acoustics, Univ. of Mississippi, 108 Lewis Hall, University, MS 38677

Murray S. Korman
Dept. of Physics, U.S. Naval Academy, Annapolis, MD 21402

Acoustics has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomenon. In this session we will be organizing a set of "Hands-On" demonstrations for a group of middle school students from the San Diego area. The goal is to foster curiosity and excitement in science and acoustics at this critical stage in the students' educational development and is part of the larger "Listen Up" education outreach effort by the ASA. Each station will be manned by an experienced acoustician who will help the students understand the principle being illustrated in each demo. Demos will be run in parallel, so students can break into small groups to encourage questions and allow students to spend more time on demos that interest them. The session will have a "Halloween" theme with decorations and costumes! Any acousticians wanting to participate in this fun event should email Josh Gladden (jgladden@olemiss.edu) or Murray Korman (korman@usna.edu).

Session 2aMU

Musical Acoustics: Physical Models For Sound Synthesis I

Edgar J. Berdahl, Chair
Dept. of Music, Stanford Univ., Stanford, CA 94305

Chair's Introduction—8:30

Invited Papers

8:35

2aMU1. Synthesis of new bowed-string sounds using physical models. Philippe Manoury (UCSD Dept. of Music, CPMC 354, 9500 Gilman Dr., MC 0099, La Jolla, CA 92093-0099)

With a virtual bowed-string model, natural aspects of the sound and sound production, such as bow pressure, position, and velocity, control the realization of new synthetic sounds. It is the best way to resolve many problems in synthetic music involving phrasing, transitions, irregularity, and regularity. But what happens when you ask the model to simulate bow movements which are quite impossible for a human being? For example, to produce vertically an exaggerated pressure at the same time as an exaggerated slow motion of the bow across a string? That is even more interesting than to reproduce a simply natural acoustical sound.

9:05

2aMU2. Modeling of a violin input admittance by direct positioning of second-order resonators. Esteban Maestre (CCRMA—Dept. of Music, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305, esteban@ccrma.stanford.edu), Gary P. Scavone (CIRMMT—Schulich School of Music, McGill Univ., Montreal, QC H3A 1E3, Canada), and Julius O. Smith (CCRMA—Dept. of Music, Stanford Univ., Stanford, CA 94305)

When approaching violin sound synthesis, the theoretical advantage of body modeling by finite difference or finite element paradigms comes from parameterizing the model by geometry and material properties. However, difficulties in representing the complexity of physical phenomena taking place have kept such approaches from raising more success, due especially to limited modeling accuracy and high computational cost. Conversely, the design of digital filters from admittance measurements, although generally offering a less meaningful parameterization, represents a more affordable technique as it provides significant fidelity at lower computational cost. Within digital filter formulation applied to the violin, modal representations can be considered as among the most physically pertinent, since vibration modes defining the timbre signature are in general observable from admittance measurements. This work introduces a technique for designing violin passive admittances by direct, non-uniform positioning of second-order resonators. Starting from admittance measurements, second-order resonator parameters are designed so that desired modes are modeled from frequency-domain data. Positve real models providing significant accuracy at low order are obtained from second-order resonator parameter fits. As an example, a two-dimensional input admittance is designed from measurements, so that it can be used for a digital waveguide model to include a control for bowing angle.

9:35

2aMU3. An overview of the afterglow phenomena compensation. Carlos Spa (Dept. of Math., Univ. Técnica Federico Santa María Av., Vicuña Mackenna 3939, San Joaquin Santiago de Chile, Chile), Jose Escolano (Telecommunication Eng. Dept., Univ. of Jaén), Toni Mateos (Barcelona Media Innovation Ctr.), and Adan Garriga (Int. Ctr. of Numerical Methods in Eng.)

In many fields such musical and room acoustics, wave propagation is usually simulated by using discrete-time numerical methods. Since 3-D simulations often show an excessive computational cost, proposals exist for extrapolating results from 3-D simulations conducted in equivalent 2-D scenarios. Among the most significant limitations of such extrapolation techniques, recently it has been pointed out the propagation of a point-like impulse in 2-D exhibits the so-called afterglow effect, which leads to obtaining non-null field values after the arrival of the wavefront. In order to compensate for this effect, a filtering process is proposed to overcome this limitation, not only in free space but also on closed spaces. Therefore, three dimensions-like impulse responses can be efficiently computed from 2-D simulations in the context of complex geometries. Recent results using pseudo-spectral time-domain, finite-difference time-domain, and digital waveguide mesh methods demonstrate the potential of this method to extrapolate impulse response from 2-D simulations with similar properties to those coming from 3-D.

10:05–10:20 Break

10:20

2aMU4. Synthesis of harpsichord pluck tones using a physical plectrum and string models. Chao-Yu J. Perng (Dept. of Phys., Stanford Univ., 382 Via Pueblo Mall, Stanford, CA 94305, perng@stanford.edu), Julius O. Smith, and Thomas D. Rossing (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ.)

The harpsichord is a plucked string keyboard instrument with a distinct sound, and previous work has been done to synthesize it [V. Valimaki *et al*., EURASIP J. Appl. Signal. **7**, 934–948 (2004)]. However, these excitation signals are extracted through recorded tones and are not physical models in the true sense. A physical model of the harpsichord plectrum and its interface with the digital waveguide string model has been proposed by us [Chao-Yu J. Perng *et al*., J. Acoust. Soc. Am. **128**, 2309(A) (2010)], and a revised model accounting for the plectrum tip was presented subsequently [Chao-Yu J. Perng *et al*., J. Acoust. Soc. Am. **129**, 2543(A), (2011)]. In this paper, we will demonstrate results of the synthesized tones using our physical plectrum-string interaction model. Changing certain physical parameters alters the synthesized tones, and they are discussed and explored. Lastly, a simple physical model of the harpsichord lute-stop is presented.

10:50

2aMU5. Modalys, a physical modeling synthesizer: More than twenty years of researches, developments, and musical uses. Rene Emile Causse, Joel Bensoam, and Nicholas Ellis (IRCAM, UMR CNRS 9912, Musical acoustics, 1 Place IGOR Stravinsky, 75004 Paris, France)

In the early 1990s, the software was initially created as an open environment with the purpose to serve as a virtual instrument maker workshop. Indeed, the modal formalism offers several interesting advantages, among them the uniform description of numerous mechanical or acoustical systems allowing easy hybridizations. Wildest virtual instruments may well be imagined, played, and manipulated. As there are very few acoustics volumes or mechanical structures tractable by an exact analytical solution, numerical methods, such as FEM, have been programmed for complex geometry for both structures and fluids. The usages are now extending from the virtual reproduction of existing instruments to industrial prototyping. This diversification as the necessity to further improve performance made it necessary to rethink many parts of the software, from the core synthesizer to the numerous interfaces: textual, MAX/MSP, OPENMUSIC, and MATLAB. To control physical modeling synthesis, it is necessary to specify the physical, low-level details of how to play an instrument. Consequently, several recent studies have focused on the study of instrumental gesture and its modeling, thereby increasing the realism of the synthesis. This presentation will review the current possibilities of the software which will be illustrated by examples. Work in progress will also be presented.

11:20

2aMU6. Sound synthesis and musical composition with the physical modeling formalism CORDIS-ANIMA. Claude Cadoz (ACROE & ICA Lab., Grenoble Inst. of Technol., 46 Ave. Flix Viallet, Grenoble, F38000, Claude.Cadoz@imag.fr), Nicolas Castagne (ICA Lab., Grenoble Inst. of Technol., 46 Av. Flix Viallet, Grenoble F38000), and Olivier Tache (ACROE, Grenoble Inst. of Technol., 46 Av. Flix Viallet, Grenoble F38000)

The two historical starting points of Computer Music, in 1957, were digital sound synthesis, founded by Max Mathews (Bell Labs), and automatic composition, founded by Lejaren Hiller and Leonard Isaacson (University of Illinois). Digital sound synthesis and computer aided musical composition then developed and are today essential components of computer music. As said by Jean-Claude Risset, sound synthesis allowed to compose the sound itself. At the end of the 1970s, a new paradigm appeared, no more based on the synthesis of the sound signal, but on the simulation of the physical objects that produce the sound. The CORDIS-ANIMA language, created by Claude Cadoz and his colleagues Annie Luciani and Jean-Loup Florens at the ACROE lab, in Grenoble (France), is one of the most important representatives of this approach. We will present this language, which allows simulating the various components of a musical instrument, but also complex orchestras and dynamic objects with macro temporal behaviors. We will then demonstrate how this language enables linking sound creation and musical composition within a single and global paradigm using the GENESIS environment, created at ACROE for musical applications of CORDIS-ANIMA, and we will play extracts of musical pieces composed with GENESIS.

Session 2aNS

Noise: Noise Impacts in Quiet Residential Communities

Richard D. Horonjeff, Chair
Consultant in Acoustics and Noise Control, 81 Liberty Square Rd., #20-B, Boxborough, MA 01719

Chair's Introduction—8:10

Invited Papers

8:15

2aNS1. Examination of valued attributes of low sound level environments. Richard D. Horonjeff (81 Liberty Square Rd. #20-B, Boxborough, MA 01719) and Herb J. Singleton, Jr. (P.O. Box 90842, Springfield, MA 01139)

The introduction of new noise sources into rural environments presents a number of unique community response challenges not typically encountered in urban and suburban environments. Low ambient sound levels combined with intermittently (rather than continuously) audible anthropogenic ambient contributors create unique soundscapes. Residents appear to value many attributes of these soundscapes and wish to preserve and protect them. Application of existing standards and guidelines (quantitative or nuisance-based) often does not meet resident expectations of protection. This paper examines a number of the valued attributes. It then asks questions about noise policy, resident expectations, adequacy of existing standards, source and ambient noise quantification, and the masking of valued attributes by a new noise source. It concludes with a list of issues that bear close examination in order to set future noise impact guidelines for low-ambient environments.

8:35

2aNS2. Challenges in selecting rural project locations under different types of noise regulations. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

It is sometimes desirable to locate certain types of projects in very rural areas, not only because of noise issues, but also because some may need to be close to electrical grid transmission lines, natural gas pipelines, transportation corridors, open space or special topographical features. Examples of such projects include mining operations, military training facilities, power plants, firing ranges, experimental proving grounds, and rescue dog training facilities, among many others. Whenever planners of such specialty facilities identify an appropriate location, invariably there are noise standards that must also be met before construction and operations can be permitted by the respective local authorities. While some noise requirements are quite specific, others are purposefully vague. Some noise regulations are just copied from other municipalities, while others are carefully crafted to provide for the desired community ambiance. However, a major problem can occur when typical community noise standards are applied to rural locations. This paper describes very different types of projects in quiet rural settings and their difficulties complying with the various types of typical municipal noise ordinances. Some suggestions are given to avoid the obvious noise regulation difficulties.

8:55

2aNS3. Case study: A quiet rural community in a commercial/industrial zone. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

A small New Hampshire town allowed a community to be built in a commercial zone, immediately adjacent to an industrial zone that is home to a long-dormant quarry. When the quarry owner applied for a permit to resume quarrying operations, the nearby residents objected on the grounds of noise impacts and other environmental concerns. A failure to consider noise as an element of community planning put the town in a difficult position. Should residents living in an commercial zone be afforded the same privilege of living in a quiet environment than those in residential zones? Should the industry be held to more stringent standards because of this non-conforming use? Both sides of this issue will be discussed.

9:15

2aNS4. Can a statewide noise rule cover both urban and rural areas? David Braslau (David Braslau Assoc., Inc., 1313 5th St. SE Ste. 322, Minneapolis, MN 55414)

Minnesota adopted a statewide noise rule in 1974. Statistical descriptors L10 and L50 for daytime and nighttime periods were established, along with a state law that no local government can adopt more stringent standards. While these are land use receiving standards, the rule is applied to noise generated by individual sources. The rule has worked well in urban environments but control of existing and new sources in rural environments has been problematical, both for generators and receptors of noise. The intent of the rule is to provide criteria that potential new development can take into account when proposing and designing new facilities, but in rural areas with very low ambient levels, the rule allows very large increases in sound level against which receptors have no recourse. Local governments have taken other approaches to control new sound sources by proposing acceptable increases over ambient noise level at receiving land uses or simply denying permits to new noise sources. Sources such as wind turbines are particularly difficult to address, since these are commonly placed in quiet rural areas. The paper discusses some problems and attempted solutions to allow new noise sources in rural areas without compromising quiet rural environments.

9:35

2aNS5. The sound of quietness. How to design sites that are perceived as quiet? Andre Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de) and Brigitte Schulte-Fortkamp (TU Berlin, Berlin 10587, Germany)

The EC Noise Directive 2002/49, related to the assessment and the management of environmental noise, demands "to preserve environmental noise quality where it is good." In addition, one major issue of the directive concerns the creation of quiet zones in urban areas. This ambitious goal of creating "acoustically green" and quiet areas in cities cannot be achieved without an interdisciplinary approach, which holistically reflects the human perception of acoustical sceneries. Sound sources constitute a soundscape and semantic attributions to sources become intrinsic properties of the respective acoustic signals. This means that humans cannot distinguish between the physical and the interpreted events. Psychological and social aspects have a sustainable impact on evaluations and must be considered to create quiet areas, which are perceived as quiet and recreative. Thus to understand residents' and visitors' reactions to environmental noise, it is imperative to investigate their expectations and experiences besides considering noise metrics and acoustical indicators. The paper will present outcomes of recent surveys, where quiet areas in Aachen, Berlin, and Leipzig and their impacts on residents and visitors were scrutinized. The analyses have been conducted with special focus on Soundscape design.

9:55

2aNS6. Individual expectations and noise policy in quiet residential settings. Nancy S. Timmerman P. E. (Consultant in Acoust. and Noise Control, 25 Upton St., Boston, MA 02118-1609, nancy.timmerman@alum.mit.edu)

Consultants in the noise field tend to encounter the more sensitive individuals. These people may feel entitled to not hear anyone else. This paper will characterize the quiet acoustical setting, and consider the expectations of the more normal individuals. It quiet settings, it is as easy to be heard as to hear. The author, whose family owned a cottage in the Manistee National Forest in Michigan, will recount some personal acoustical observations from being there in the summer. This paper will also examine noise policies for these settings and look at "dB above ambient" as a criterion, since it is in use in Massachusetts, where this consultant practices. Illustrations will also be taken from clients of the author in western Massachusetts and in Maryland.

10:15–10:30 Break

Contributed Papers

10:30

2aNS7. Proving a facility complies with noise regulations. Mark V. Giglio (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA 01776, mvg@cavtocci.com)

This paper examines the difficulties associated with proving compliance with environmental noise regulations. Topics include determining pre-existing background sound, post-construction measurements during facility operations, and balancing corporate rights with neighborhood expectations. When a new facility is built in a quiet neighborhood, noise complaints are commonplace. The new facility is required to prove compliance with the local noise regulation. Compliance measurements must be made when the facility cannot be shut down and ambient sound levels are unknown. A statistical analysis of the pre-existing background sound often suggests the probability of proving compliance is minimal. In addition, bias toward/against one party in the dispute can have significant ramifications.

10:45

2aNS8. A three-dimensional noise mapping study for heterogeneous traffic conditions. Ramachandraiah Alur and Kalaiselvi Ramasamy (Dept. of Civil Eng., Indian Inst. of Technol., Chennai, India)

Characteristics of noise in terms of frequency, noise level, vary considerably with respect to heterogeneous traffic. Urban traffic noise characteristics in the cities of a like India are slightly varied in the sense that the composition of the traffic is heterogeneous. A heterogeneous traffic stream composes of vehicles having different speeds, sizes, and operating characteristics. While studying the feasibility of noise reduction in some of the areas around a heterogeneous traffic stream, a three-dimensional (3-D) noise mapping study has been attempted with the help of Arc GIS, Arc Scene along with field measurements. The methodology involves to build 3-D noise models to analyze the 3-D occurrence of noise pollution. Specifically in this work a 3-D acoustical model has been developed for a local area. Subsequently, the observation points around the buildings have been determined and the noise levels have been calculated using the authors' regression model. The noise mapping parameters such as Ld, LN, and Lden have been calculated incorporating the geometrical features of the roads and varying heights of the buildings, heterogeneity, and prominent honking conditions. This work leads to the prediction of noise levels in front of building facades both in horizontal and vertical directions.

11:00

2aNS9. Energy-based method for near real-time modeling of sound field in complex urban environments. Stephanie M. Pasareanu, Marcel C. Remillieux, and Ricardo A. Burdisso (Mech. Eng. Dept., Virginia Tech, 142 Durham Hall, Blacksburg, VA 24061, psteph7@vt.edu)

Prediction of the sound field in large urban environments has been limited thus far by the computational heaviness of standard numerical methods such as boundary element (BE) or finite-difference time-domain methods. Recently, a considerable amount of work has been devoted to developing energy-based methods for this application, and results have shown the potential to compete with conventional methods. However, these developments have been limited to 2-D models, and no real description of the phenomena at issue has been exposed. Here, the mathematical theory of diffusion is used to predict the sound field in complete 3-D complex urban environments. A 3-D diffusion equation is implemented by means of a simple finite-difference scheme and applied to two different types of urban configurations. This modeling approach is validated against BE and geometrical acoustic solutions, showing a good overall agreement. The role played by diffraction near the source is discussed, and suggestions are made on the possibility to accurately predict the sound field in complex urban environments in near real time simulations.

11:15

2aNS10. Experimental and numerical study on the propagation of impulsive sound around buildings and induced structural loading. Marcel C. Remillieux, Joseph M. Corcoran, T. Ryan Haac, Ricardo A. Burdisso (Vib. and Acoust. Labs., Mech. Eng. Dept., Virginia Tech, Blacksburg, VA 24061, mremilli@vt.edu), and U. Peter Svensson (Acoust. Res. Ctr., Dept. of Electron. and Telecommunications, Norwegian Univ. of Sci. and Technol., NO-7491 Trondheim, Norway)

Propagation of impulsive sound around buildings and induced structural loading are investigated experimentally and numerically. Experiments were conducted on a rectangular building at Virginia Tech using sonic booms generated by an explosive technique. Assuming linear-acoustic propagation and acoustically rigid surfaces, these experiments were simulated with a three-dimensional numerical model, in the context of geometrical acoustics, by combining the image source method for the reflected field (specular reflections) with the Biot-Tolstoy-Medwin method for the diffracted field. This numerical model is validated against a boundary element solution and against experimental data, showing a good overall agreement. Some of the key advantages of this modeling approach for this application are pointed out such as the ability to model three-dimensional domains over a wide frequency range and also to decompose the sound field into direct, reflected, and diffracted components, thus allowing a better understanding of the sound-propagation mechanisms. Finally, this validated numerical model is used to investigate sound propagation around a cluster of six rectangular buildings, for a range of elevated source positions.

11:30

2aNS11. Sound transmission modeling for residential buildings using finite elements. Beom Soo Kim (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg. University Park, PA 16802, buk104@psu.edu) and Victor W. Sparrow (The Penn State University, 201 Appl. Sci. Bldg., University Park, PA 16802)

Noise transmission into residential building structures has been studied in order to understand the outdoor-to-indoor transmission characteristics. This work is motivated by the need to model the transmission of subsonic aircraft noise into homes. As a first step, finite element (FE) models of a simplified residential house developed using both MSC MD Nastran and FFT Actran were compared to check the validity of the modeling techniques as well as to analyze their applicability for low frequencies. Fluid structure interaction was considered to include the presence of the enclosed acoustic fluid. Model relevance was also checked with impulsive load measured data supplied by NASA in 2007. The FE model of a detailed individual room was developed to show interior pressure responses. [Work supported by FAA.]

11:45

2aNS12. Investigation of the sound distribution in street canyons with non-parallel building Façades. Kaj Erik Piippo and Shiu-keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong)

Sound propagation in street canyons have been a topic of interest for decades. In a congested city such as Hong Kong, street canyons with large height to width ratios are a common sight. Many prediction models have been developed over the years, but the problem is still of interest because of the complexity due to the multiple reflections and varying surface conditions in real life cases. In this paper, a 1:4 scale model experiment of a street canyon, 4 m long and 2 m high, is presented where one of the facades is inclined. The reference case is with two parallel facades, which is compared with cases when one facade is tilted 80°, 70°, and 60°. The measurements were conducted in an anechoic chamber and a line source array generating white noise was used for simulating traffic noise. The sound distribution patterns were studied on the reference wall and the inclined wall. It was observed that the sound strength and the reverberation decrease rapidly when inclining the opposite wall. For low and mid frequencies, the sound reduction was more significant closer to the top of the inclined facade, while the opposite could be seen for the vertical facade.

TUESDAY MORNING, 1 NOVEMBER 2011 ROYAL PALM 3/4, 7:30 A.M. TO 12:00 NOON

Session 2aPA

Physical Acoustics: Physical Acoustics of Extreme Environments, Condensed Matter, and More

Veerle M. Keppens, Cochair
Materials Science and Engineering, Univ. of Tennessee, Dougherty Hall, Knoxville, TN 37996

Cristian Pantea, Cochair
Sensors and Electrochemical, Los Alamos National Lab., Los Alamos, NM 87545

Chair's Introduction—7:30

Invited Papers

7:35

2aPA1. Resonant ultrasound techniques for measurements from 500 mK to 700 K. Jonathan B. Betts, Albert Migliori (NHMFL-PFF Los Alamos Natl. Lab., Los Alamos, NM 87507), Arkady Shehter (NHMFL, 1800 E. Paul Dirac Dr., Tallahassee, FL 32310-3706), Victor R. Fanelli, and Fedor F. Balakirev (NHMFL-PFF Los Alamos Natl. Lab. Los Alamos, NM 87507)

Measuring elastic moduli using resonant ultrasound spectroscopy (RUS) over a very wide temperature range requires very specialized hardware. The authors have developed both this hardware and some interesting software to help make measurements in these very challenging conditions. Solutions to these problems will be presented, along with results from some recent RUS experiments carried out at the National High Magnetic Field Laboratory in Los Alamos. [Work was supported by the National Science Foundation, State of Florida, and the U.S. Dept. of Energy.]

8:00

2aPA2. Resonant ultrasound spectroscopy at high temperatures and pressures: Palladium hydride near the tri-critical point. Joseph R. Gladden III and Rasheed Adebisi (Dept. of Phys. & NCPA, 108 Lewis Hall, Univ. of Mississippi, University, MS 38677, jgladden@olemiss.edu)

Much fundamental quantum and critical phenomenon physics has been learned from precision elastic constant and attenuation measurements at low temperatures. Less attention has been paid to phenomenon in the high temperature regime outside of specialized areas such as geophysics. However, interesting fundamental and applied physics does not stop at room temperature. In this talk, I will review methods developed for applying resonant ultrasound spectroscopy at high temperature (<1300 K) and high hydrostatic pressure (<140 atm). One of the phenomenon we have studied using this apparatus is palladium hydride near the tri-critical point in the temperature, pressure phase diagram. The results from these measurements have validated a recent theory by Schwarz and Khachaturyan (2006) regarding the hysteresis in metal hydride systems which predicts a strong softening in the elastic moduli.

8:25

2aPA3. Phonon-driven electron localization in delta-plutonium. Albert Migliori (Natl. High Magnetic Field Lab., Los Alamos Natl. Lab., Los Alamos, NM 87545)

Three remarkable properties of delta-Pu, accurately measured, lead to exceptionally strong constraints on *ab-initio* electronic structure models and suggest a new route must be taken to understand the localization of electronic states. The three properties that lead to this are the thermal expansion coefficient, the temperature dependence of the elastic moduli, and the absolute values of the elastic moduli. Both neutron diffraction and dilatometer measurements of the thermal expansion reveal a broad temperature range where thermal expansion is zero, and where small changes in Ga concentration can make thermal expansion vary from small and positive to small and negative. Measurements of the bulk modulus of these alloys over the same temperature range reveal extreme softening on warming. How can the bulk modulus, which is the curvature of the energy with respect to volume, change when volume does not? This is the central property that is not encompassed by present electronic structure models. We discuss theory and RUS measurements to understand better what must be true.

8:50

2aPA4. Acoustic velocity measurements on solids and liquids under extreme conditions of pressure and temperature. Baosheng Li and Wei Liu (Mineral Phys. Inst., Stony Brook Univ., Stony Brook, NY 11794)

Elastic properties of materials under extreme conditions of pressure and temperature are of great interests to researchers in many disciplines. In the last decade, acoustic velocity measurements using an improved ultrasonic interferometry method have been developed in both MA-6 and MA-8 types of multi-anvil high pressure apparatus. By placing the piezoelectric transducer (lithium niobate, 10 deg Y-cut) in a stress-free location and using extended delayline, high S/N acoustic signal can be maintained, while sample is under high pressure and temperature. By combining with synchrotron X-radiation, measurements of sound velocities using ultrasonic interferometry, crystal structure and unit cell parameters using X-ray diffraction, and sample strain (length) using X-radiographic imaging, can be made simultaneously, all *in-situ* at a high pressure and temperature, enabling a pressure-standard-free characterization of solid and liquid materials to 25 GPa and 1800 K. Results on ceramic and metallic materials from recent experiments will be presented to show velocities as a function of pressure and temperature, absolute pressure determination, equation of state for glass, and the application to liquids. Other new developments, such as controlling sample stress state at high P and T, the study of composites, and materials undergoing phase transformations will also be reviewed [Work sponsored by NSF and DOE/NNSA.]

9:15

2aPA5. Acoustics of metals under extreme conditions by laser-ultrasonics in diamond anvils cell. Frédéric Decremps (UPMC, 4 place Jussieu Paris 75005, France), Laurent Belliard, Bernard Perrin, and Michel Gauthier (UPMC, 4 place Jussieu, Paris 75005, France)

Major progress on ultrafast acoustics instrumentation and diamond anvils design during the last 2 yr now allows detailed elastic and visco-elastic studies under extreme conditions and on a wide variety of systems. I will here mainly review the state of the art of the recent development of a method combining the time-resolved picosecond optical technique and a diamond anvil cell to measure sound velocity Decremps *et al*., [Phys. Rev. Lett. **100**, 3550 (2008)]; Decremps *et al*., [Rev. Sci. Instrum. **80**, 73902 (2009)]. Contrary to other groups which currently scope with these problems mostly using large facilities, we propose here entirely new and novel technique to measure the sound velocity of solid and liquid under high pressure and high temperatures. I will illustrate these possibilities by a number of recent studies on crystalline, polycrystalline, and liquid metals. Prospects will be discussed.

Contributed Papers

9:40

2aPA6. Acoustic microscopy investigation of superconducting materials. T. Tahraoui, S. Debboub, Y. Boumaïza, and A. Boudour (Faculty of Sci., Dept. of Phys., Lab. LEAM, Badji Mokhtar Univ. ANNABA B.P.12, 23000, ANNABA, Algeria)

The present work has as subject the study of the ultrasonic attenuation in some superconductor materials. This study is based on the simulation of the acoustic signal obtained by the reflection acoustic microscope upon the exploration of a coated or uncoated material. The examination of the simulated signal has permitted the determination of the variation of the reflection coefficient with respect to the incidence angle of the exciting acoustic wave. We have also determined the elastic constants, the velocities of different modes of propagation, the acoustic attenuation of the Rayleigh mode as a function of temperature from the reflection coefficient, and the acoustic signature.

9:55

2aPA7. Weakly nonlinear simple waves in Hertzian chains. B. Edward McDonald and David Calvo (Naval Res. Lab., Washington, DC 20375, ed.mcdonald@nrl.navy.mil)

The discrete system of equations for a granular chain consisting of a large number of spheres interacting via the hertz force is cast as an effective

medium. In the long wavelength limit, the second order equation of motion for the effective medium possess a subset of simple waves obeying a first order equation of reduced nonlinear index. Simple waves are those in which knowledge of one dependent variable determines all the rest. For a given initial strain, the simple wave solution prescribes initial mass velocity. Strain and velocity profiles from the first order equation are used as initial conditions in simulations for the second order discrete system. Results for viscous and inviscid shock formation compare very well between the second order system and the reduced first order equation. Second order simulation of colliding waves reveals the ability of waves to pass through each other, with a phase advance accruing during the collision process. Results may be related to explosions in granular media. [Work supported by the Office of Naval Research.]

10:10–10:30 Break

10:30

2aPA8. Cylindrical bubble dynamics. Yurii A. Ilinskii, Todd A. Hay, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

Bubbles generated between closely spaced parallel surfaces are approximately cylindrical. Here, cylindrical bubble dynamics are compared with those of spherical bubbles. In the linear approximation, the product of bubble radius and natural frequency of the monopole mode in water is $\sim$1m/s for a cylindrical bubble versus $\sim$3m/s for a spherical bubble. Radiation damping of a cylindrical bubble is an order of magnitude greater than for a spherical bubble, and for cylindrical bubbles larger than $\sim$1 μm it is the dominant loss mechanism, whereas spherical bubbles must be larger than $\sim$ 1 mm before radiation losses dominate. Also in contrast with spherical bubbles, the inertial load on a cylindrical bubble in an unbounded incompressible liquid is infinite and prohibits the existence of a monopole mode. Derivation of a dynamical equation for cylindrical bubbles in Rayleigh-Plesset form therefore requires the incompressible liquid to be of finite extent. The radial extent needed to impose the appropriate inertial load is found by comparison with results for a compressible liquid to be approximately 200 bubble radii. The resulting dynamical equation was further augmented to account for bubble deformation due to coupling of surface modes through nonlinearity. [Work supported by NIH DK070618 and EB011603.]

10:45

2aPA9. Simulations of self-organization of bubbles in acoustic fields in three dimensions. Nail A. Gumerov (UMIACS, Univ. of Maryland, 115 A.V.Williams Bldg., College Park, MD 20742, gumerov@umiacs.umd.edu), Iskander S. Akhatov (North Dakota State Univ., Fargo, ND 58108), Yekaterina V. Volkova, and Uliana O. Agisheva (Bashkir State Univ., Ufa, 450074, Russia)

Self-organization of bubbles (structure formation) and related self-action of the acoustic field is a strongly non-linear phenomenon, which is observed, e.g., in acoustic cavitation and sonochemical reactors. The phenomenon is a manifestation of a two-way field-particle interaction, when the bubbles change their position and sizes due to the Bjerknes and other forces and rectified diffusion, and affect the acoustic properties of the medium, which leads to restructuring of the acoustic field. The models of this phenomenon available in literature are revised, and a model based on the spatio-temporal averaging, which includes the above mentioned effects, is derived. Based on this model, a 3-D pseudospectral code for simulation of bubble self-organization is developed and tested. Simulations show interesting spatio-temporal behavior of the bubbles and the acoustic field, which is initialized as a standing wave but may drastically change its structure due to bubble dynamics (especially near the resonance). Bubble pattern formation is sensitive to many parameters including parameters of the acoustic field, bubble initial size, number density, spatial distribution, and ambient conditions. The effects found in the result of simulations are discussed. [This research is supported by the Grant of the Ministry of Education and Science of the Russian Federation (G34.31.0040).]

11:00

2aPA10. Sonic boom structural response simulation of an historic structure. Michael M. James, J. Micah Downing (Blue Ridge Res. and Consulting, LLC, 15 W. Walnut St., Ste. C, Asheville, NC 28801, Michael.James@BlueRidgeRes.com), and Joel M. Garrelick (Appl. Physical Sci., Inc., Lexington, MA 02421)

The potential for structural damage from sonic booms can be assessed by using stationary sound sources. Such sources simulate sonic booms as incident, linear, and plane acoustic waves. In theory, the source simulates the pressure field on a structure arising from this incident wave, including influences from the surrounding acoustic environment. In practice, compromises must be made in the placement of the acoustical source and its orientation to the structure and the resulting change in the expected incident angle of a sonic boom. In this effort, we evaluated multiple structural elements of Fort Jefferson in the Dry Tortugas, Fl which ranged from structurally sound to fragile. The structural concern was damage to the more fragile elements from sonic booms generated in the surrounding airspace. The methodologies and results will be discussed. [Work sponsored by Naval Facilities Command, Southeast.]

11:15

2aPA11. Acoustic oscillations in the early universe. David Knobles (KSA, P.O. Box 27200, Austin, TX 78731)

Acoustic oscillations in the baryonic-photon plasma leave an imprint on both the cosmic microwave background radiation (CMBR) and the power spectrum of the mass distribution in the universe. This effect is well known and is currently applied to determine the value for quantities such as the percentage of dark matter and dark energy in the universe. This study examines the contributions of higher order terms in the equations for the radiation-baryonic acoustic oscillations prior to radiation decoupling from matter. The energy and momentum conservation equations are combined with the Einstein field equations for a Robertson Walker metric containing the perturbations. An attempt is made to consider the effect of the higher order perturbations on the observed power spectrum and on the anisotropy of the CMBR. The sensitivity of these higher order terms to adiabatic, instead of non-adiabatic, boundary conditions on the fluctuations outside the particle horizon is considered. [Work supported by KSA.]

11:30

2aPA12. Acoustic levitator for contactless transport and mixing of droplets in air. Nada Bjelobrk, Majid Nabavi, and Dimos Poulikakos (Dept. of Mech. and Process Eng., Inst. of Energy Technol., Lab. of Thermodynamics in Emerging Technologies, ETH Zurich, CH-8092, Zurich, Switzerland, nadab@ethz.ch)

Acoustic levitation is a simple and cost-effective method for investigation of small amounts of any kind of material. Standard levitators focus the standing acoustic wave only along its traveling direction. Therefore, levitated objects (entrapped in the potential minima) are separated by potential maxima which cannot be overcome. Thorough design of the emitter plate and consequently its vibration pattern allows a more complex acoustic field between the emitter and reflector, establishing multiple potential minima perpendicular to the wave propagation direction in the H1-mode. The number of minima is regulated by changing the emitter-reflector distance H. Thus, multiple solid or liquid samples can be simultaneously levitated and rapidly transported over several centimeters by controlling H. A piezoelectrically driven droplet generator was used to eject pL droplets into the acoustic field and coalesce them to a 1–2 μL sized droplet in the preferred potential minimum. Two suspended droplets were transported and mixed when two potential minima were merged into one by increasing H. During pattern transition, high pressure fluctuation might increase droplet Bond number beyond its critical value and cause droplet atomization. This was prevented by continuous adjustment of acoustic power.

11:45

2aPA13. Effects of heat conduction in wall on thermoacoustic wave propagation in a gas-filled, channel subject to temperature gradient. Nobumasa Sugimoto and Hiroaki Hyodo (Dept. of Mech. Sci. Graduate School of Eng. Sci., Univ. of Osaka, Toyonaka, Osaka 560-8531, Japan, sugimoto@me.es.osaka-u.ac.jp)

This paper examines effects of heat conduction on acoustic wave propagation in a gas-filled, channel subject to temperature gradient axially and extending infinitely. Within the narrow-tube approximation in the sense that a typical axial length is much longer than a channel width, the system of linearized equations for the gas supplemented by the equation for heat conduction in the solid wall is reduced to a thermoacoustic-wave equation for excess pressure uniform over the cross-section. This is the one-dimensional equation taking account of the wall friction and the heat flux at the wall surfaces given in the form of hereditary integrals. This equation is derived rigorously under the narrow-tube approximation, and is valid for any form of disturbances. The effects of heat conduction appear in the form proportional to the square root of the product of the ratio of the heat capacity per volume of the gas to that of the solid, and the ratio of the thermal conductivity of the gas to that of the solid. Although the product is very small usually, which endorses validity in neglect of the heat conduction, it is revealed that there are situations in which its effects are enhanced, depending on geometry and materials.

TUESDAY MORNING, 1 NOVEMBER 2011 ROYAL PALM 5/6, 8:00 TO 11:35 A.M.

Session 2aSA

Structural Acoustics and Vibration and Architectural Acoustics: Extraction of Information from Correlations of Random Vibrations

Earl G. Williams, Cochair
Naval Research Lab., Acoustics Div., 4555 Overlook Ave., SW, Washington, D.C. 20375

Karim G. Sabra, Cochair
Mechanical Engineering, Georgia Inst. of Technology, 771 Ferst Dr., NW, Atlanta, GA 30332

Invited Papers

8:00

2aSA1. Extracting the earth response from noise and complex earthquake data. Roel Snieder, Nori Nakata, Kees Wapenaar, and Evert Slob (Ctr. for Wave Phenomena, Colorado School of Mines, CO 80401 rsnieder@mines.edu)

Theory shows that when random noise is generated in proportion to the local dissipation rate, one can extract the response of a system from correlations of field measurements of such noise. Such a condition of equilibrium is rarely satisfied for elastic waves of the earth. Yet the earth response can be extracted from measured noise. We show examples of the retrieval of P-waves and of S-waves. The latter type of waves are retrieved from traffic noise in an urban environment. Since the traffic noise is excited by localized sources (road and railroads), it displays strong amplitude variations. In order to compensate for such amplitude variations, we use cross-coherence rather than cross-correlations for the data processing. We also analyze complicated waveforms excited by earthquakes to create maps of the shear-wave velocity in Japan and show that the shear wave velocity changes with time; this velocity drops throughout northeastern Japan with about 5% after the recent Tohoku-Oki earthquake. Our measurements show that the shallow subsurface in Japan weakens after the earthquake over a region about 1 200 km wide.

8:25

2aSA2. Green's function retrieval from noise by multidimensional deconvolution. Kees Wapenaar, Joost van der Neut, Evert Slob (Dept. of Geotechnol., Delft Univ. of Technol., Stevinweg 1, 2628CN Delft, The Netherlands, c.p.a.wapenaar@tudelft.nl), and Roel Snieder (Colorado School of Mines, Golden, CO 80401-1887)

The correlation of noise at two receivers is approximately proportional to the Green's function between these receivers. The approximation is accurate when the medium is lossless and the noise field is equipartitioned. These assumptions are in practice often violated: the medium of interest is often illuminated from one side only, the sources may be irregularly distributed and losses may be significant. For those situations, the correlation function is proportional to a Green's function with a blurred source. The source blurring is quantified by a so-called point-spread function which, like the correlation function, can be derived from the observed data (i.e., without the need to know the actual sources and the medium). The blurred source can be focused by multidimensionally deconvolving the correlation function for the point-spread function. We illustrate the correlation and deconvolution methods with several examples and discuss the advantages and limitations of both methods.

8:50

2aSA3. The potential for extracting the electromagnetic earth response from uncorrelated noise. Evert Slob, Kees Wapenaar (Dept. of Geotechnology, Delft Univ. of Technol., Stevinweg 1, 2628 CN, Delft, Netherlands, e.c.slob@tudelft.nl), and Roel Snieder (Ctr. for Wave Phenomena, Colorado School of Mines, Golden, CO 80401-1887)

Thermal electromagnetic radiation from an absorbing medium allows for extracting the response of this medium. In the earth, thermal noise is usually very weak and other forms of electromagnetic noise prevail. Noise below radio frequencies is generated in the atmosphere, ionosphere, and magnetosphere, while cosmic noise generates electromagnetic waves above radio frequencies that reach the earth's surface. The noise requirements are discussed for energy and Lagrangian forms of earth response extraction by correlation and multi-dimensional deconvolution. It is shown that if the uncorrelated noise sources are located outside the earth, only the earth response to incident electromagnetic waves can be extracted. Potential applications are found for ground-penetrating radar, and these are illustrated with numerical examples. For coupled seismic waves and electromagnetic fields in fluid-filled porous media, uncorrelated seismic noise is shown to be sufficient to approximately extract the electro-seismic earth response.

9:15

2aSA4. Imaging and monitoring with ambient vibrations: A review. Larose Eric (ISTERRE, CNRS, & UJF, BP 53, 38041 Grenoble cedex 9, France)

The principle of passive imaging and reconstructing the Green functions by means of correlating ambient vibrations or noise will be reviewed. Some basic processing procedures for optimizing the convergence of the correlations, along with the role of multiple scattering, will also be presented. Monitoring with ambient noise constitutes a different goal that relies on different assumptions on the background noise structure. Similarities and differences between the imaging and the monitoring approaches will be addressed.

9:40

2aSA5. Correlation processing of ocean noise. W.A. Kuperman (Marine Physical Lab. of the Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238)

Correlation processing of ocean ambient noise has been a topic of growing interest. While theory confirms the efficacy of this procedure, experimental confirmation has been limited to a few data sets. The basic issue of the potential utility of this procedure is the time needed to build up the relevant cross-correlation peaks that are diagnostic of the ocean environment. This time is a function of frequency/bandwidth, the ocean environment and noise structure/distribution, sensor separation and, if employed, the array configuration. After reviewing some basic background, a selection of experimental results for noise originating from ships, surface generated sources, and geophysical sources is presented.

10:05–10:15 Break

10:15

2aSA6. Using cross-correlations of ambient vibrations for passive structural health monitoring of a high-speed naval ship. Karim G. Sabra (School of Mech. Eng. Georgia Inst. of Technol., 771 Ferst Dr., NW Atlanta, GA 30332-0405)

Previous studies have used the cross-correlation of ambient vibrations (CAVs) technique to estimate the impulse response (or Green's function) between passive sensors for passive imaging purposes in various engineering applications. The technique (CAV) relies on extracting deterministic coherent time signatures from the noise cross-correlation function computed between passive sensors, without the use of controlled active sources. Provided that the ambient structure-borne noise field remains stable, these resulting coherent waveforms obtained from CAV can then be used for structural monitoring even if they differ from the actual impulse response between the passive sensors. This article presents experimental CAV results using low-frequency random vibration data (>50 Hz) collected on an all-aluminum naval vessel (the HSV-2 Swift) operating at high speed (up to 40 kn) during high sea states. The primary excitation sources were strong wave impact loadings and rotating machinery vibrations. The consistency of the CAV results is established by extracting similar coherent arrivals from ambient vibrations between the pairs of strain gages, symmetrically located across the ship's centerline. The influence of the ship's operating conditions on the stability of the peak coherent arrival time, during the 7 days trial, is also discussed. [Sponsored by ONR, N00014-09-1-0440.]

10:40

2aSA7. Extracting information from an array of sensors in a diffuse noise field using random matrix theory. Ravi Menon, Peter Gerstoft, and William Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA 92093)

Isotropic noise fields are often used to model several practical diffuse noise fields. For an array of equidistant sensors in such a noise field, the cross-spectral density matrix (CSDM) of the array is a Toeplitz sinc matrix. Here, the eigenvalues of the CSDM for ideal isotropic noise fields are first derived for infinite arrays. The eigenvalues have close connections with classical array processing concepts such as the invisible region in frequency-wavenumber space (region where there is no propagating energy, but a spectrum can be calculated). Random matrix theory deals with eigenvalue distributions of random matrices and its concepts are applied here by modeling the array snapshot vectors as zero-mean, unit variance Gaussian random variables, with a sinc covariance matrix. Using the Stieltjes transform, the eigenvalues of the ideal CSDM are related to those of the sample CSDM, and an analytical solution for the distribution of the eigenvalues of the sample CSDM is obtained. At frequencies where the array is spatially undersampled, increasing the number of observations results in the noise masquerading as a signal, which could lead to erroneous signal detections. We demonstrate how knowing and understanding the eigenvalue distribution helps improve the extraction of information from ocean ambient noise.

Contributed Papers

11:05

2aSA8. Analyzing structure-borne sound by using the forced vibro-acoustic components. Logesh Kumar Natarajan and Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, logesh@wayne.edu)

At the 158th meeting of the ASA, a numerical study on analyzing structure-borne sound radiation using the forced vibro-acoustic components (F-VACs) established by the Helmholtz equation least squares (HELS) method was presented [Natarajan *et al*., JASA, **126**, 2244 (2009)]. The present paper presents an experimental case study of this approach on a baffled square plate subjected to a point force excitation using random signals. The radiated acoustic pressures are measured by a planar array of microphones at a very close distance to the plate surface. These input data are used to determine the F-VACs of the plate obtained by decomposing the transfer functions into orthogonal space via singular value decomposition. These F-VACs are correlated to sound radiation through sound power calculations. Meanwhile, the F-VACs are expanded in terms of the natural modes of the plate, so the dominant F-VACs that are directly responsible for sound radiation can be correlated to the natural modes of the plate. Using this F-VAC analysis, one can identify the critical natural modes of a structure, which by themselves have no direct relationships to sound radiation. Based on the F-VAC analysis results, the plate is modified and tested to examine its effectiveness in reducing structure-borne radiation.

11:20

2aSA9. An improved method of structural-borne random vibration mass attenuation. Arya Majed, Ed Henkel (Appl. Structural Dynam., Inc., P.O. Box 272428, Houston, TX 77277), and Ali Kolaini (NASA/Jet Propulsion Lab., California Inst. of Technol., Pasadena, CA 91125)

In 2009, the NASA Engineering and Safety Center (NESC) Vibroacoustics Working Group identified vibroacoustic environment predictions as one of the highest risk areas for new launch vehicle programs. Specifically, the working group identified the need for improved random vibration mass attenuation prediction methods as key to mitigating this risk. This paper derives a multi drive point, multi axis, interface sized equation between the unloaded and loaded drive point random vibration accelerations and cross correlations. The method's development and associated acoustic test validation program were a collaborative effort between ASD, NESC, and NASA/JPL. The derivation utilizes the methods of modal synthesis and random vibration averaging and precludes the use of any simplifying structural interaction assumptions besides the interfaces behaving linearly (no interface deadbands). Key to the method's improved accuracy is its built-in mechanism for combining the contributions of the different unloaded drive-point accelerations and associated cross-correlations. The method reduces to the time-tested method of Norton–Thevenin for the single drive-point case. Predictions are compared to measurements from NASA/JPL acoustic chamber test of a panel loaded by a component. It is shown that the method accurately captures the major spectral characteristics of attenuations and amplifications resulting in improved mass loaded environment predictions.

TUESDAY MORNING, 1 NOVEMBER 2011 PACIFIC SALON 4/5, 7:55 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication and Psychological and Physiological Acoustics: Fifty Years of Invariant/Variant Speech Features: Where Do We Stand?

Jont B. Allen, Cochair
Beckman Inst., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801

Peggy B. Nelson, Cochair
Dept. of Speech, Languages, Hearing Sciences, Univ. of Minneosta, 164 Pillsbury Dr., S.E., Minneapolis, MN 55455

Sandra Gordon-Salant, Cochair
Dept. of Hearing and Speech Science, Univ. of Maryland, College Park, MD 20742

Chair's Introduction—7:55

Invited Papers

8:00

2aSC1. What causes the confusion patterns in speech perception? Feipeng Li (Dept. of Biomedical Eng., Johns Hopkins Univ., 108 W 39 St., 33, Baltimore, MD 21210) and Jont Allen (Beckman Inst., Univ. of Illinois at Champaign-Urbana, Urbana, IL 61801)

It is widely believed that speech contains redundant cues which make it robust in noise. We have discovered that natural speech often contains conflicting cues that cause significant confusion patterns in speech perception. To demonstrate the impact of conflicting cues on the identification of consonant sounds, we selected four plosives /ta, ka, da, ga/ and four fricatives /si, Si, zi, Zi/ that show significant confusion with other sounds from a female high-performance talker in the LDC database and derived a set of test stimuli by removing the dominant cue and manipulating the conflicting cue. It was found that the consonants always morph to their competing sounds, once the dominant cue is unavailable. The confusion can be strengthened by enhancing the conflicting cue. These results demonstrate that the identification of consonant is dependent on a careful balancing between the dominant cue and the conflicting cues. Such prior knowledge about perceptual cues can be used to manipulate the perception of consonants, as demonstrated by the selected examples of nonsense syllables, meaningful words, and sentence.

8:20

2aSC2. Speak, memory—Wherefore art thou, invariance? Steven Greenberg (Silicon Speech, 4683 Hawaina Way, Kelseyville, CA 95451, steveng@silicon-speech.com)

Spoken language is highly variable, reflecting factors of environmental (e.g., acoustic-background noise, reverberation), linguistic (e.g., speaking-style), and idiosyncratic (e.g., voice-quality) origin. Despite such variability, listeners rarely experience difficulty understanding speech. What brain mechanisms underlie this perceptual resilience, and where does the invariance reside (if anywhere) that enables the signal to be reliably decoded and understood? A theoretical framework—DejaNets—is described for how the brain may go from "sound to meaning." Key is speech representations in memory, crucial for the parsing, analysis, and interpretation of sensory signals. The acoustic waveform is viewed as inherently ambiguous, its interpretation dependent on combining data streams, some sensory (e.g., visual-speech cues), others internal, derived from memory and knowledge schema. This interpretative process is mediated by a hierarchical network of neural oscillators spanning a broad range of time constants (ca. 15–2 000 ms), consistent with the time course and temporal structure of spoken language. They reflect data-fetching, parsing, and pattern-matching involved in decoding and interpreting the speech signal. DejaNets accounts for many (otherwise) paradoxical and mysterious properties of spoken language including categorical perception, the McGurk effect, phonemic restoration, semantic context, and robustness/sensitivity to variation in pronunciation, speaking rate and the ambient acoustic environment. [Work supported by AFOSR.]

8:40

2aSC3. Invariant acoustic cues of consonants in a vowel context. Riya Singh and Jont B. Allen (Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801)

The classic JASA papers by French and Steinberg (1947), Fletcher and Galt (1950), Miller and Nicely (1955), and Furui (1986) provided us with detailed CV+VC confusions due to masking noise and bandwidth and temporal truncations. FS47 and FG50 led to the succinctly summarizing articulation index (AI), while MN55 first introduced information-theory. Allen and his students have repeated these classic experiments and analyzed the error patterns for large numbers of individual utterances [http://hear.beckman.illinois.edu/wiki/Main/Publications], and showed that the averaging of scores removes critical details. Without such averaging, consonant scores are binary, suggesting invariant features used by the auditory system to decode consonants in isolated CV. Masking a binary feature causes the consonant error to jump from zero to chance (within some small subgroup of sounds), with an entropy determined by conflicting cues, typically present in naturally spoken sounds. These same invariant features are also used when decoding sentences having varying degrees of context. A precise knowledge of acoustic features has allowed us to reverse engineer Fletcher's error-product rule (FG50), providing deep insight into the workings of the AI. Applications of this knowledge is being applied to a better understanding of the huge individual differences in hearing impaired ears and machine recognition of consonants.

9:00

2aSC4. The perception of phonetic features and acoustic cues by impaired listeners. Matthew B. Winn, Monita Chatterjee (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD 20742, mwinn1@.umd.edu), and William J. Idsardi (Dept. of Linguist., Univ of Maryland, College Park, MD 20742)

The search for invariant acoustic cues in the perception of speech features has been an under-utilized blessing to those interested in individuals with hearing impairment. While multiple co-varying acoustic cues complicate the search for invariance, they provide plentiful opportunity for impaired listeners to perceive contrasts that would otherwise be lost when salient information is compromised by hearing loss or noise. Classic confusion matrices and information transfer analyses account for phonetic features perceived, but do not always reveal the acoustic cues that drive these perceptions; it is rarely acknowledged that difficult listening conditions that are likely to result in a re-prioritization of acoustic cues. In this presentation, we discuss some completed and ongoing work suggesting that impaired listeners (or simulated impaired listeners) don't merely show different amounts of success but exhibit different listening strategies (i.e., cue-weighting strategies) when perceiving phonetic features. This is similar to findings related to developing language learners and adult learners of a second language. Implications are discussed here for cochlear implant users, listeners with high-frequency hearing loss, and listeners in background noise in tasks that measure the use of cues in both the auditory and visual domains.

9:20

2aSC5. Individual variance of hearing-impaired consonant perception. Woojae Han (Dept. of Speech and Hearing Sci. Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, woojaehan@gmail.com) and Jont Allen (Univ. of Illinois at Urbana-Champaign)

Individuals with sensorineural hearing loss (SNHL) are prescribed hearing aids (HAs), based on the results of clinical measurements. Although the HAs do help these individuals communicate in quiet surroundings, many listeners have complained that their HAs do not provide enough benefit to facilitate understanding of normal speech. We argue that the current clinical measurements, which interpret the result as a mean score (e.g., pure-tone average, speech recognition threshold, AI-gram, etc.), do not deliver sufficient information about the characteristics of a SNHL listener's impairment when hearing speech, resulting in a poorly fitting HA. We confirm how reliably this consonant-vowel (CV) test could measure a SNHL listener's consonant loss using only zero-error utterances (in normal hearing listeners) and having a statistically suitable number of presentations in CVs, in order to characterize unique SNHL consonant loss. As noise increased, the percentage of error and confusions of target consonants increased. Although some consonants showed significantly higher errors and resulted in more confusion than others, SNHL ears have a very different consonant perception/error, which may not be either measured or analyzed by the use of average scores. Comparison between the two (separated) phases of the study supports a good internal consistency for all SNHL ears.

9:40

2aSC6. Dynamic spectral structure really does support vowel recognition. Joanna H. Lowenstein and Susan Nittrouer (Dept. of Otolaryngol., The Ohio State Univ., 915 Olentangy River Rd., 4th Fl, Columbus, OH 43212, lowenstein.6@osu.edu)

Traditional approaches to the study of speech perception manipulate cues that are spectrally and temporally discrete, assuming these cues define phonetic segments. Alternatively, dynamic structure across longer signal stretches may support recognition of linguistic units. Strange and colleagues [Strange *et al*. J. Acoust. Soc. Am. **74**, 695–705 (1983)] seemed to support this view by showing that adults can identify vowels in CVCs even when large portions of the syllable centers are missing, but dynamic structure at the margins is preserved. However, adults may just be "filling in the missing parts," having learned how structure at the margins covaries with structure in the middle. To test these alternatives, adults and 7-yr-olds labeled sine-wave, and vocoded versions of silent-center stimuli. The former preserves dynamic structure; the latter obliterates it. If listeners simply fill in missing parts, adults should perform equally well with sine-wave and vocoded stimuli, and children should perform equally poorly with both. Instead, all listeners performed better with sine-wave stimuli. These outcomes provide further support for the perspective that speech perception is facilitated by broader and longer acoustic structure than that represented by notions of the acoustic cue. [Work supported by NIDCD Grant DC-00633.]

9:55

2aSC7. Deriving invariance by integrating cues computed relative to expectations. Allard Jongman (Dept. of Linguist., Univ. of Kansas, Lawrence, KS 66045, jongman@ku.edu) and Bob McMurray (Univ. of Iowa, Iowa City, IA 52240)

Invariance and variability are notions that have stimulated much research on the acoustics and perception of speech. While some invariance must exist to explain how listeners generally derive the same message from spoken input, the question remains at which level invariance occurs. The theory of acoustic invariance explicitly states that the invariance is found in the acoustic signal. While some phonemic distinctions do seem to be represented by invariant acoustic properties, it is clear that many acoustic cues are context-dependent. Normalization or compensation processes for coping with specific sources of variability have been proposed as solutions to this context-dependency. We examined acoustic properties for a large corpus of American English fricatives. We argue that the lack of invariance is not a major obstacle and demonstrate that it can be overcome by the use of multiple acoustic cues and a relatively simple compensation scheme by which cues are recoded relative to expectations derived from context (vowel quality and speaker). A comparison of this approach (C-CuRE, Computing Cues Relative to Expectations) [McMurray and Jongman, Psych. Rev. **118**, 219–246 (2011)] to acoustic invariance and exemplar approaches suggests that only C-CuRE achieved a similar accuracy to listeners and showed the same effects of context.

10:10–10:25 Break

10:25

2aSC8. Invariants for phones or phonemes?: Only literacy leads us to expect them. Robert Port (Dept.s of Ling'cs and Cognit. Sci, Indiana Univ, Bloomington, IN 47405, port@indiana.edu)

This special session asks about "invariants," but *invariants in what domain under what transformations?* If it is acoustic invariants for phones or phonemes, i.e., letter-sized units for spelling words, which implies the same acoustic cues for all contexts where a phone or phoneme is used in transcription. This clear impossibility is why Liberman turned to "motor invariants." Yet acoustic invariants for phones are what is required for perceptual categorization. Phones (and phonemes) must be rejected as a linguistic memory code due to, at least, (a) evidence of indexical information in speech memory, (b) lack of evidence of discrete jumps for feature changes, and (c) evidence of real-time patterns in language (see Port and Leary, 2005, Lang; Port, 2011, Eco Psych). Our strong intuitions of abstract and letter-like linguistic memory result from our lifelong training with an alphabet, not from data. Memory for speech must incorporate rich detail and continuous (not serial) time and also supports some degree of abstraction for generalization to new contexts. Linguistic units like distinctive features, phones, and phonemes are merely categories, not symbol tokens. Categories are cultural conventions regarding events "considered the same" by some community (like *tree* and *game*) and do not necessarily have any invariant properties.

10:40

2aSC9. In defense of discrete features. Khalil Iskarous (Haskins Labs., 300 George St., New Haven, CT)

A major challenge for discrete feature theories of speech perception is that the acoustic signal is continuously changing within and between segments at a high rate. These theories must explain how continuously changing spectral patterns are perceived as discrete and independent features. This problem does not disappear after auditory transformations of the acoustic signal, since temporal and frequency resolution, despite being coarse, still preserve most of the signal dynamics. In this work, it is shown that if articulatory features are extracted from the continuously changing acoustic signal, assuming that the signal is the output of a lossless vocal tract terminated in a unit resistance, i.e., a Darlington configuration (Iskarous, Journal of Phonetics, 2010), it is possible to obtain a discretely changing place of articulation feature from the continuous signal change. This will be illustrated using articulatory inversion of VV and CV transitions from the X-ray microbeam database for five male and five female participants. This work supports theories of discrete and independent perceptual features, as assumed for instance by Miller and Nicely (1955), if the acoustic signal is interpreted in terms of the vocal tract actions that cause it (Goldstein and Fowler, 2003).

10:55

2aSC10. On the non-acoustic nature of acoustic invariants. D. H. Whalen (Haskins Labs, 300 George St., Ste. 900, New Haven, CT 06511, and City U New York, whalen@haskins.yale.edu)

Various proposals have attempted to put speech targets into invariant "acoustic" form, sometimes with an additional transformation into "auditory" space. These targets, however, are not strictly acoustic. Targets for vowels, for example, are transformed so that vocal tract length differences (between talkers, especially across men, women, and children) are taken into account. Such a transformation makes the resultant targets combinations of articulatory and acoustic information. The auditory transformation improves automatic speech recognition, but the theoretical underpinnings for this result have been unclear. Ghosh, Goldstein, and Narayanan [J. Acoust. Soc. Am. **129**, 4014–4022] shows that the articulatory information is maximized by the auditory transform, indicating that this transform is not solely in the acoustic domain. Moreover, a lowered F3 has been proposed as the production target for American English /r/ [e.g., Nieto-Castanon *et al*., J. Acoust. Soc. Am. 117, 3196–3212], but synthesis that retains an exemplary /r/ F3 while altering F1 and F2 results in other percepts, such as /w/ or a pharyngeal glide. F3 as an acoustic target is insufficient by itself and must incorporate articulatory dynamics implied by the other formants. Thus, "acoustic" invariants, to the extent they work at all, do so by incorporating articulation.

11:10

2aSC11. Perceptual recovery of phonetic features in blanked segments of disyllabic words. Pierre L. Divenyi (Speech and Hearing Res., VA Northern California Health Care System, Bldg. R4, 150 Muir Rd., Martinez, CA 94553, pdivenyi@ebire.org) and Adam C. Lammert (Signal Anal. and interpretation Lab., Dept. of Comput. Sci. Univ. of Southern California, Los Angeles, CA 90089)

Spondees, both true disyllabic words and concatenated monosyllabic word pairs, had their middle section replaced by silence that extended from the midpoint of the first to the midpoint of the second vowel; the silence

thus included the final consonant(s) of the first and the initial consonant(s) of the second syllable. These stimuli were presented to normal-hearing young listeners instructed to guess both monosyllabic half words. Input and response spondees were orthographically aligned and analyzed as confusion matrices. Articulatory gestures were estimated for each token via the Haskins Laboratories TADA articulatory synthesis. After time alignment, articulatory distances were calculated for each stimulus-response pair. In the blanked middle of the spondee, phoneme-based confusions of place-of-articulation were high (39% accuracy), while gestural distances underlying this feature (location and degree of tongue tip constriction, lip aperture, and velar constriction) were under 10%. These results suggest that acoustic traces of a significant portion of gesture trajectories underlying consonantal place-of-articulation that start before and/or terminate after the silence, are perceived by the listener. An articulatory phonology might thus be more robust to degradation. [Work supported by NSF, AFOSR, and the VA Medical Research.]

11:25–12:00 Panel Discussion

TUESDAY MORNING, 1 NOVEMBER 2011 ROYAL PALM 1/2, 8:00 A.M. TO 12:00 NOON

Session 2aSP

Signal Processing in Acoustics, Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics: Overcoming Environment/Medium Effects in Acoustic Tracking of Features or Parameters

Ananya Sen Gupta, Cochair
Dept. of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543-1053

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

Invited Paper

8:00

2aSP1. Modeling and tracking temporal fluctuations for underwater acoustic communications. James Preisig (Dept. of Appl. Ocean Phys. and Eng., WHOI, Woods Hole, MA 02543, jpreisig@whoi.edu)

High-rate, phase-coherent communications through the underwater acoustic channel requires implicitly or explicitly tracking the fluctuations of the channel impulse response. Current techniques estimate the time-varying channel impulse response using observations of signals that have propagated through the channel over some averaging interval and often rely on some method of representing the time variability in the channel. For example, the work by Stojanovic that enabled phase-coherent communications through the underwater channel represented the time-variation as a common phase rotation (Doppler shift) of all taps of the channel impulse response. The tracking of channel fluctuations involves a fundamental trade-off between the number of independent parameters used to represent both the channel impulse response coefficients (i.e., the dimensionality of the estimation problem) and their temporal variations, the rate of channel fluctuations that can be tracked, and the signal to noise ratio (SNR). This talk will present new and review established techniques for modeling time variability and reducing dimensionality in underwater acoustic channel. The performance of these approaches as a function of SNR and rate of channel fluctuation will be compared and their computational complexity will be discussed.

Contributed Papers

8:20

2aSP2. Effectiveness of sparse optimization techniques against rapid fluctuations in the shallow water acoustic channel. Ananya Sen Gupta and James Preisig (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543)

The shallow water acoustic channel exhibits rapid temporal fluctuations due to fluid and platform motion as well as reflection from moving surfaces. Tracking the time-varying channel delay spread effectively for subsequent equalization is an open signal processing challenge. The delay-Doppler spread function characterizes this time-variability over a selected range of Doppler frequencies but itself does not exhibit stationary behavior over longer time intervals. Typically the delay-Doppler spread function and sometimes the delay spread itself follow a sparse distribution where most of the energy is concentrated in a few significant components distributed sparsely over a larger support. A variety of sparse optimization techniques have recently been proposed in the compressive sensing literature to efficiently track sparsely distributed coefficients. However, the ill-conditioned nature of the delay-Doppler spread estimation problem, coupled with the necessity to track the complex-valued coefficients directly in real time, renders direct application of traditional sparse sensing techniques infeasible and intractable in the shallow water acoustic paradigm. The talk will provide a synopsis of well-known and recently proposed sparse optimization techniques, with focus on mixed norm algorithms, along with a comparative analysis of these techniques in the shallow water acoustic paradigm over simulated and experimental field data.

8:35

2aSP3. A wave-making towing tank for underwater communication in a multi-path environment using time reversal method. Gee-Pinn James Too (Natl. Cheng Kung Univ., Dept. of Systems and Naval Mechatronic Eng., No. 1 University Rd. Tainan 70101 China, z8008070@email.ncku.edu.tw)

A 4 m × 8 m × 176 m wave-making towing tank, which is original, used for resistance measurement for boats is a perfect test platform for underwater communication in a multi-path environment. The objective in the present study is to establish a process of underwater communication in a multi-path environment using the time reversal method. The source projector sends out a BPSK signal, while the signal is received and processed in order to restore the original signal at the source by time reversal method. The communication process is conducted via both numerical simulation and experiment for cases of underwater communication in a 4 m × 8 m × 176 m towing tank. It is found that TRM process reduces the error rate of the transmission significantly and improves the communication quality in multi-path environment. [Works are supported by National Science Council of Taiwan.]

Invited Papers

8:50

2aSP4. Geophysical parameter estimation. Max Deffenbaugh (ExxonMobil Res. and Eng. Co., 1545 Rte. 22 East, Annandale, NJ 08801, max.deffenbaugh@exxonmobil.com)

Seismic exploration for oil and gas is a parameter estimation problem. Geological properties and fluid content of subsurface reservoirs are sensed from the earth's surface. Seismic data are acquired by generating elastic waves at the surface and recording the reflections off subsurface targets using large receiver arrays. At present, it is not practical to determine earth parameters by iteratively refining an earth model until simulated seismic data match recorded field data. The computational demands of accurately simulating viscoelastic wave propagation in a 3-D heterogeneous earth are prohibitive. Instead, seismic data are processed to remove many complexities of wave propagation in the real environment, like multipath, attenuation, dispersion, shear waves, and interface waves. With these complexities removed, the processed data conform to simpler propagation models which can be solved in reasonable computer time. The cost of each simplification, however, is the loss of some information about the subsurface parameters of interest. This tradeoff between the tractability of the inversion and the accuracy of the result can be quantified by comparing computational time versus the Cramér-Rao bound on subsurface parameter estimates. Examples are discussed for some commonly used seismic signal processing algorithms.

9:10

2aSP5. Modeling dominant mode rejection beamformer notch depth using random matrix theory. John R. Buck (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747-2300, johnbuck@ieee.org) and Kathleen E. Wage (George Mason Univ., Fairfax, VA 22030)

Any practical tracking algorithm must mitigate the effect of both nonstationary environments and loud interfering sources. The dominant mode rejection (DMR) adaptive beamformer (ABF) is often used to address these challenges. The DMR beamformer periodically updates the sample covariance matrix (SCM) employed to compute the ABF weights. A tension arises between the large number of snapshots required for each SCM update to null loud interferers effectively and the small number of snapshots required for each SCM update to track nonstationarities. In practice, observed notch depths for DMR fall dramatically short of those predicted theoretically from ensemble statistics. This talk bridges the gap between theoretical and observed performance by presenting a simple linear asymptotic model for the DMR notch depth derived from random matrix theory results on the accuracy of the SCM eigenvectors. The model predicts the mean DMR notch depth as a function of the number of snapshots given the interferer-to-noise ratio (INR), the array size, and the interferer location relative to the look direction. Close agreement is demonstrated between the model and simulations over a wide range of INRs and array sizes. [Work supported by ONR 321US.]

9:30

2aSP6. Dominant mode rejection beamformer notch depth: Theory versus experiment. Kathleen E. Wage (ECE Dept., George Mason Univ., 4400 Univ. Dr. MSN 1G5, Fairfax, VA, kwage@gmu.edu), John R. Buck (Univ. of Massachusetts Dartmouth, N. Dartmouth, MA 02747), Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

The dominant mode rejection (DMR) adaptive beamformer attenuates loud interferers by directing beampattern nulls toward signals contained in the dominant subspace [Abraham/Owsley, Proc. Oceans, 1990]. The dominant subspace is defined by the eigenvectors associated with the largest eigenvalues of the sample covariance matrix. DMR performance is primarily determined by how closely the eigenvectors of the sample covariance matrix match the true interferer directions. Random matrix theory (RMT) describes how the accuracy of the sample eigenvectors varies with the interferer-to-noise ratio (INR), array size, and the number of snapshots used to estimate the sample covariance. A simplified analytical model based on RMT predicts the mean DMR notch depth as a function of INR, array size, interferer location and the number of snapshots. This talk compares the RMT predictions with experimental results obtained by cancelling array strum in data from a vertical array deployed in the Philippine Sea. The array strum interference predominantly falls within the subspace spanned by the first two eigenvectors of the covariance matrix. Notch depth statistics obtained using a large set of receptions show good agreement between theory and experiment. [Work supported by ONR.]

9:50

2aSP7. A moment-based technique to palliate pervasive clutter while preserving objects of interest. Roger C. Gauss and Joseph M. Fialkowski (Naval Res. Lab., Code 7144, Washington, DC 20375-5350, roger.gauss@nrl.navy.mil)

High false alarm rates are a persistent problem for ASW active sonars. This talk describes a recently developed sonar clutter characterization and control method, the "Poisson-Rayleigh Inspired Method" (PRIM). The closed-form PRIM is based on NRL's two-parameter Poisson-Rayleigh (P-R) model that, like the popular K-type model, provides a physical context for relating the characteristics of data distributions to scatterer attributes (density and relative strength). However, with its extra degree of freedom, the P-R model offers the potential to exploit more information through higher-order (4th and 6th) data moments, and thus do a better job of characterizing clutter. The technique is demonstrated on normalized, broadband sonar clutter data collected in two range-dependent shallow-water environments, one bottom dominated and one fish dominated. The data results suggest that in contrast to the popular K-distribution, the power of the discrete-scatterer component in the P-R distribution can provide feature information that is largely independent of the peak signal-to-noise ratio of an echo, implying a potential for improved rejection of clutter relative to target-like objects. [Work supported by ONR.]

10:10–10:30 Break

10:30

2aSP8. Underwater target tracking using the waveguide invariance. Lisa M. Zurk (NEAR-Lab., Portland State Univ., 1900 SW 4th, Portland, OR 97201)

Target tracking in underwater environments is complicated by false alarms introduced from scattering from the sea bottom and sea-air interface, as well as returns from biologics in the water column. It is difficult to devise a method to discriminate the false alarms from targets of interest because their nature is highly dependent on the environmental conditions, which are often poorly known. A topic of recent interest in active sonar is the application of the waveguide invariant to improve target discrimination and tracking. The invariant relationship gives a robust (i.e., not environmentally dependent) method of relating the time-evolving frequency content to attributes of the target. In this talk, the concept of target discrimination and tracking using the waveguide invariant is discussed in the context of other environmentally robust techniques. Results are presented for fixed and towed array systems.

10:50

2aSP9. Integrated approaches to tracking in cluttered environments. John S. Allen, Grant Pusey (Dept. of Mech. Eng., 2540 Dole St., Holmes 302, Honolulu, HI 96822), John Gebbie, and Martin Siderius (Dept. of ECE, NEAR Lab., Portland State Univ., Portland, OR 97207)

A variety of acoustic applications encompass the detection and tracking of signals in highly cluttered environments. Despite the interdisciplinary nature of this problem, often novel approaches and advances are not well known and hence not typically applied outside their respective sub-fields. In this study, we highlight array processing, time frequency analysis and non-stationary signal processing techniques with respect to applications in both physical and underwater acoustics. In particular, we examine the acoustic tracking of small vessels, divers and AUVs in shallow water, harbor areas. An examination of the underlying physical acoustics of the ambient noise sources is a significant factor in the development of improved and novel tracking methods. Time scale filters are investigated for ambient noise reduction from snapping shrimp. The signal processing advantages of a combined two array systems in an L shaped configuration are discussed. Theoretical predictions and simulations are compared with experiments results from a synchronized system of two 24 element arrays deployed at the Kilo Nalu Nearshore Observatory (Honolulu, HI) near the Honolulu Harbor. Novel acoustic tracking techniques based on depth are explored with the two array systems. [Work sponsored by DHS.]

11:10

2aSP10. Acoustic tracking using compressive sensing. Geoffrey F. Edelmann and Charles F. Gaumond (U. S. Naval Res. Lab., 4555 Overlook Ave., SW, Code 7140, Washington, DC 20375, geoffrey.edelmann@nrl.navy.mil)

This paper presents the application of compressive sensing to several problems of acoustic detection and localization on numerical and at-sea data. Compressive sensing results are shown for the detection of ship tones, bearing estimation from horizontal towed array, and target detection using a vertical array. Compressive sensing is established to be robust to low signal-to-noise ratio, to require few snap shots, and to need few sensors to achieve high probability of detection and low probability of false alarm. As a technique, compressive sensing appears insensitive to noise anisotropy, noise spectral coloration, and mild signal deviations from the sparseness paradigm. This technique potentially applies to a broad spectrum of acoustic applications. [Work supported by the Office of Naval Research.]

Contributed Papers

11:30

2aSP11. Tracking a defect in the multiple scattering regime. Planes Thomas, Larose Eric (ISTERRE, CNRS, & UJF, BP 53, 38041 Grenoble cedex 9, France), Rossetto Vincent (LPMMC, CNRS, & UJF, BP 166, 38042 Grenoble cedex 9, France.), and Margerin Ludovic (Universit de Toulouse, CNRS, Institut de Recherche en Astrophysique et Planétologie, 31400 Toulouse, France)

We describe a time-resolved monitoring technique for heterogeneous media, especially multiply scattering media. Our approach is based on the spatial variations of the cross-coherence of diffuse waves acquired at fixed positions but at different dates. The technique applies to all kind of waves, but a particular attention will be paid to ultrasound propagating in concrete. To locate and characterize a defect occurring between successive acquisitions, we use a maximum likelihood approach combined with a diffusive propagation model. We quantify the performance of this technique called LOCADIFF with numerical simulations. In several illustrative examples, we show that the change can be located with a precision of a few wavelengths and that its effective scattering cross-section can be retrieved. We investigate how the accuracy and precision of the method depends on the number of source–receiver pairs, on the time window used to compute the cross-correlation and on the errors in the propagation model. Applications can be found in nondestructive testing (civil engineering), seismology, radar, and sonar location.

11:45

2aSP12. Passive acoustic tracking of marine mammals and anthropogenic sound sources with autonomous three-dimensional small-aperture arrays. Martin Gassmann, Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0205)

Marine mammals produce a wide range of frequency-modulated sounds at low and high frequencies as well as directional broadband echolocation sounds in a refractive ocean environment. This creates several challenges for passive acoustic long-term tracking of the various marine mammal species. To overcome these, three-dimensional small-aperture hydrophone arrays coupled to seafloor multi-channel recording packages were deployed in a large aperture array in the Southern California Bight. Taking advantage of the experimental setup in the oceanic waveguide, time and frequency-domain tracking methods will be presented and tracks of marine mammals as well as anthropogenic sources will be shown. This provides a tool to study over long timescales behavioral responses of tracked marine mammals to tracked anthropogenic sources.

Session 2aUWa

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

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

Chair's Introduction—7:55

Invited Papers

8:00

2aUWa1. Sediment acoustic models and Biot's theory. Nicholas P. Chotiros and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, TX 78713-8029)

The development of geo-physical descriptions of the sediment and models to predict the acoustic properties are critical to the application of sediment acoustics. The former follow standard geo-physical methods of sediment classification, based on grain size, density, and other physical descriptors. The latter started as a fluid approximation, followed by a visco-elastic approximation with five frequency-independent parameters, consistent with sediment acoustic data up to the 1980s. Recent experimental data have revealed deficiencies in this approach, particularly in the case of sandy sediments, which cover a large fraction of the continental shelves. The measurements are more consistent with a poro-elastic model, consisting of Biot's theory with extensions to account for the particular physics of granular media. There are currently two approaches to the remedy: (a) a visco-elastic model with frequency dependent parameters that mimic the experimental data and (b) a poro-elastic model with the necessary attributes. It is shown that (a) would be a significant improvement over existing models, but (b) is the preferred solution. A recent discovery concerning the viscosity of nano-meter water films has resolved a problem with the dimensions of the grain contact gap. Future plans will center on further rationalization and reduction of input parameters to develop a practical poro-elastic model. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

8:20

2aUWa2. Bottom loss from geoacoustic inversions. N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada)

The concept of bottom loss has traditionally been used as a measure of acoustic reflectivity and transmission at the ocean bottom. The measure was generally derived experimentally from measurements of transmission loss versus range, using a simple model that assumed perfect reflection at the sea floor to process the transmission loss data of bottom reflected signal paths. More recently, the interaction of sound with the ocean bottom has been described in terms of a geoacoustic model that defines a physical structure of sound speed, density and attenuation in the material beneath the sea floor. The model parameters are inferred from acoustic field data or observables derived from the field using sophisticated inversion techniques. The estimated model can be compared with bottom loss measurements through calculations of the plane wave reflection coefficient. This paper illustrates comparisons of bottom loss measurements for low- and high-frequency bands (50 –20,000 Hz) and calculations from estimated geoacoustic profiles for deep and shallow water environments. Conditions are discussed that limit the performance of present day inversion techniques: rough interfaces on and below the sea floor, consolidated material that supports shear wave propagation and range variation of sub-bottom structure. [Work supported by ONR Ocean Acoustics Team.]

8:40

2aUWa3. A bottom-sediments province database and derived products, Naval Oceanographic Office. Peter Fleischer, William M. Becker, and Peter D. Baas (Naval Oceanogr. Office, Stennis Space Ctr., MS 39522, peter.fleischer@navy.mil)

The Naval Oceanographic Office maintains a bottom-sediment province database at three levels of resolution: (1) a low-resolution legacy data set derived from secondary sources for the low- and mid-latitude ocean basins; (2) medium-resolution, actively maintained ''Regional Sediments'' data sets covering most of the continental margins of Eurasia and North America; and (3) high-resolution, limited-extent data sets derived from acoustic imagery. Sediment provinces are categorized via an ''Enhanced'' sediment classification consisting of seven locational, ten compositional, and 97 textural components. The resultant large number of categories allows for retention of source-data nomenclature but can be unwieldy and redundant. To provide accessibility and consistency to a variety of users, the

Enhanced categories are reclassified into simplified or specialized sediment category sets. These include the High Frequency Environmental Acoustics categories for performance prediction, a mean grain-size set, a non-technical littoral set, as well as other bottom characteristics such as burial potential and bottom type. The Enhanced categories have also been assigned a limited number of geoacoustic and physical properties. Although the province approach and the reclassification schemes produce useful inputs for models and other predictions, users must recognize that they are subject to inherent limitations and ambiguities.

9:00

2aUWa4. Bottom loss measurements and their use in studying the physics of sound propagation in marine sediments. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA 16804)

Bottom reflection loss measurements have been conducted for four decades and have provided key insights into the physics of propagation in marine sediments. The advantage of bottom loss as an analysis quantity is that, in principle, it completely isolates the role of the seabed and permits separation and identification of key physical mechanisms. For example, it was early deep water bottom loss measurements that first made it clear that strong positive sediment sound speed gradients existed and the concomitant surprising realization that the dominant seabed interacting path was the refracted and not reflected path over a wide angular and frequency band. In this talk, various discoveries from bottom loss measurements are discussed including the role of sediment layering, attenuation gradients, and shear waves.

9:20

2aUWa5. Variability in normal-incidence acoustic response in shallow-water marine sediments. Roger W. Meredith (U. S. Naval Oceanogr. Office, Stennis Space Ctr., MS 39529), Richard W. Faas (Dept. of Marine Sci. Univ. of Southern MS, SSC, MS 39529), and Douglas N. Lambert (Naval Res. Lab., Stennis Space Ctr., MS)

Due to several processes, lateral deviations in geotechnical properties of sediments occur at many spatial scales and in all types of sediments. These deviations influence sediment bulk elasticity and density and determine the degree and type of acoustic response. Small changes in one or more properties can produce a wide variation in the acoustic response and empirical curve fitting most often serves as models for these relationships. Sedimentary data and acoustic variability at 30 and 50 kHz from three sites in the Mississippi Sound [Lambert *et al*., Mar. Geol. **182** (2002)] have been further analyzed and compared with the available Shepard's sediment classes. Initial observations revealed trends in sediment variability that affect the accuracy of sediment classifications based on acoustic response. This analysis deviates from traditional sediment classification schemes and utilized hierarchical clustering with user-selected combinations of geotechnical parameters to group individual sediment samples. The sediment properties are correlated with the acoustic coefficient of variation of the ensemble-averaged acoustic response and with the group-averaged acoustic response. The variability of sediment properties and acoustic response show varying degrees of correlation between parameters of major and minor importance.

9:40

2aUWa6. Milestones in naval through-the-sensor environmental acoustic measurements. Martin Barlett, Walter E. Brown, and Joe M. Clements (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713)

This paper describes recent developments in the Bottom Loss and Scattering Strength Measurement System (BLOSSMS). The objectives of the BLOSSMS project are to (1) create technology from which measurements of several types of active sonar scattering phenomena may be made using a single unifying measurement and analysis scheme, (2) integrate measurements from Fleet active sonar systems with seafloor bathymetry and modeled acoustic data into a single root database, (3) develop algorithms that are based on sound physics and oceanographic principles to identify salient information in the root database for analysis, and (4) produce acoustic scattering parameters that are useful in sonar system performance analysis and the optimization of sonar operations. In this paper, the methods used to measure bottom backscatter strength, bottom forward scatter (bottom loss), and volume scatter strength are discussed. Examples of the resulting scattering parameters are provided. Also discussed are the advantages, disadvantages, and challenges that are associated with this approach to measuring fundamental acoustic scattering phenomena.

10:00–10:15 Break

Contributed Papers

10:15

2aUWa7. Bottom loss modeling and sand grain size. Marcia J. Isakson and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, misakson@arlut.utexas.edu)

Bottom loss is a critical parameter in sonar performance assessment and propagation modeling. There exist many empirical models that predict bottom loss based both on measured parameters and on sand grain size. In this study, the empirical model predictions are compared with in situ measurements of bottom loss. The predictions are based on both the measured parameters and the sand grain size. It is found that at mid frequencies, the predictions based on sand grain size are much more consistent with the measurements than the model based on the measured parameters. This is due to unrealistic empirical sediment parameters such as density which mimic other physical processes not included in the model. However, when the empirical sand grain size model is extrapolated to predict bottom loss at higher frequencies, the results are inconsistent with measurements. [Work supported by ONR, Ocean Acoustics.]

10:30

2aUWa8. Comparative analysis of mid-frequency bottom loss derived from two distinct measurement paradigms. Martin Barlett, Walter E. Brown (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713), Carolyn Harris (Appl. Op. Res., Inc., Solana Beach, CA 92075-2077), and Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR 97201)

This paper describes a comparison of acoustic mid-frequency bottom loss data derived from two distinctly different measurement methods, systems, and platforms. One data set was derived from measurements made by a passive drifting line array. The second data set was derived from measurements made from a Navy destroyer equipped with an AN/SQS-53C active

sonar system that utilizes a hull-mounted volumetric cylindrical array. Data were collected off the west coast of the United States. Detailed comparisons and analysis of the derived co-located bottom loss data were made. The metric used for comparison was the difference of the estimated MGS curves at specific co-located sites. The set of MGS curves used as the metric for comparison between the two methods ranges from MGS-1, corresponding to a low-loss bottom, to MGS-9, corresponding to a high-loss bottom, in one tenth increments, i.e., MGS-1.1, MGS-1.2, to MGS-8.9, MGS-9.0. The results from the two methods were found to be in excellent agreement.

10:45

2aUWa9. Role of rattlers in bottom loss and sound wave propagation. James G. Berryman (Earth Sci. Div. Lawrence Berkeley Natl. Lab., One Cyclotron Rd., MS90-R1116, Berkeley, CA 94720)

It is known from both laboratory and numerical experiments that weakly or poorly consolidated granular media have some fraction of the grains that do not contribute to the overall mechanical strength of the granular system. Such loose grains are sometimes called rattlers, and their presence affects both acoustic wave speeds and attenuation. Sound wave speeds tend to be reduced while the wave attenuation tends to be increased. An analytical model of such systems will be presented and comparisons to available data will be shown.

11:00

2aUWa10. Analysis of through-the-sensor observed and modeled reverberation using sensor derived scattering parameters. Martin Barlett, Walter E. Brown (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713), and Ruth E. Keenan (SAIC, Austin, TX 78758)

Recent developments in the bottom loss and scattering strength measurement system (BLOSSMS) program support the use of on-scene measurement of acoustic scattering parameters important to the performance of mid-frequency sonar systems. These parameters could include bottom and volume scatter strength (backscatter), and bottom loss. In this paper, examples of BLOSSMS-derived bottom loss and backscatter estimates for data collected off the west coast of United States are used to compute reverberation levels for select locations and bearings. Included in the discussion are analysis of the oceanographic conditions and relevant bathymetric features.

11:15

2aUWa11. Implications of the presence of shell hash on the speed and attenuation of sound in water-saturated granular sediments. Theodore F. Argo IV and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78712-0292)

The propagation of sound through water-saturated granular sediments has been widely studied, yet there is no consensus on an expected wave speed and attenuation in these materials owing to variation in both physical properties and test methods used for their interrogation. One aspect confounding model predictions is the presence of inhomogeneities such as rocks, bubbles, and biological organisms in an otherwise homogeneous sediment. In a laboratory setting, it is possible to control both manufacture of artificial sediments with specific inclusions and the measurement method used to observe their properties. To study the effect of inclusions on the speed of sound and attenuation in an otherwise homogeneous sediment, shells were systematically added to a sand sediment. The volume fraction of shells relative to sand grains was varied and the speed of sound and attenuation was measured using a time of flight technique from 200 to 800 kHz. Results will be compared to both sediment acoustic models and scattering models.

11:30

2aUWa12. Direct measurements of sediment sound speed at mid- to high-frequency in a sand sediment. Jie Yang, Brian T. Hefner, and Dajun Tang (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Direct measurements of sediment sound speed were made near Panama City, Florida in April 2011. Considerable effort is being made to provide detailed environmental characterization for an upcoming mid-frequency sound propagation and reverberation experiment in 2013 and the measurements presented here are part of that effort. Two direct measurement systems are shown: one is called the Sediment Acoustic-speed Measurement System (SAMS) and the other is the attenuation array which was deployed in the Sediment Acoustics Experiment 2004. SAMS consists of ten fixed sources and one receiver. The receiver is driven into the seabed by a motor, which allows precise penetration depth up to 3 m with arbitrary step size. Measurements were made in the frequency range of 1 – 50 kHz. The attenuation array consists of two transmitters and two receivers mounted on a diver-deployable frame and can provide surficial sediment sound speed and attenuation to a depth of about 10 cm between 40 and 260 kHz. Sediment sound speeds obtained using the two systems can be compared in the overlapping frequency region for consistency. Initial analysis of sediment sound speed will be shown. [Work supported by ONR.]

11:45

2aUWa13. High frequency dispersion model for the water-saturated granular medium. Haesang Yang (Dept. of Ocean Eng., Seoul Natl. Univ., Seoul 151-742, Korea, coupon3@snu.ac.kr), Keunhwa Lee (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, kel003@ucsd.edu), and Woojae Seong (Seoul Natl. Univ., Seoul 151-742, Korea)

Dispersion relation for the p-wave sound speed and attenuation has been described by several models based on continuum or scattering theory. As an alternative approach, this study proposes a model describing the dispersion relation for the p-wave in case of grain scatterers existing in background porous medium. Dispersion relations are shown as a function of different grain size distribution and Rayleigh parameter ka. For quantitative analysis of the proposed model, experiments are performed using water-saturated glass beads. Two sets of experiments employing unimodal and bimodal grain size distributions are performed and are used for comparison with the current proposed model.

Session 2aUWb

Underwater Acoustics: Acoustic Prediction and Real-Time Sonar Signal Processing for Vehicle Autonomy Applications

Kevin D. LePage, Cochair
NATO Undersea Research Centre, Viale San Bartolomeo 400, La Spezia, 19126 Italy

Henrik Schmidt, Cochair
Dept. of Mechanical Engineering, Massachusetts Inst. of Technology, 77 Massachusetts Ave., Cambridge, MA 02139

Chair's Introduction—7:45

Invited Papers

7:50

2aUWb1. Pragmatic model-based adaptation for optimal acoustic communication and sensing on autonomous marine vehicles. Toby E. Schneider and Henrik Schmidt (Ctr. for Ocean Eng., Dept. of Mech. Eng., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, tes@mit.edu)

Autonomous underwater vehicles can be valuable acoustic sensing platforms due to their maneuverability, low cost, and sensor-driven adaptivity compared to ships. However, the logistics of integrating acoustics research into artificially intelligent systems can be daunting. In this work, the autonomy software provides an abstract representation of the acoustic environment (e.g., sea surface, water column, and sea floor parameters, source and receiver positions) which it updates continuously from local and remote sensor data. Upon request, this environment is translated into the native representation of an acoustic model which returns the requested calculation (e.g., transmission loss, travel times). The model is set up as a server, capable of handling requests from multiple autonomy subsystems at once (e.g., target tracking prediction, acoustic communications optimization). Thus, the acoustic model and the autonomy software are kept ignorant of each other's implementation specifics. Results will be presented from the shallow water GLINT10 experiment where a vehicle adaptively tracked in depth the minimum modeled transmission loss from a buoy source. Furthermore, a deep sea simulation study which combines target tracking and acoustic communications was conducted. For both studies, the autonomy software MOOS-IVP and the BELLHOP ray tracing codes were used.

8:15

2aUWb2. Deployment of wideband bio-inspired sonars for autonomous underwater operations. David M. Lane, Chris Capus, Keith Brown, and Yan Pailhas (Ocean Systems Lab., Heriot-Watt Univ. Edinburgh, Scotland, EH14 4AS United kingdom)

Recent developments in the understanding and application of wideband bio-inspired acoustic sensors are enabling new applications of autonomous underwater vehicles (AUVs) for security and oilfield use. Appropriate use of wideband signals has enabled improved discrimination between natural and man-made objects that have physically similar appearance, as well as the detection of buried and partially buried objects. Wideband sonar developments in the Ocean Systems Laboratory at Heriot-Watt University have focused on prototypes based on the bottlenose dolphin sonar, covering a frequency band from around 30 to 150 kHz and having a frequency dependent beam-width that is, considerably larger than conventional imaging sonars. In parallel, AUV technology has developed to allow much greater levels of autonomy in allowable vehicle behavior. New generations of vehicle have moved beyond switching of pre-programmed scripts (behaviors) and their parameters, to systems that interleave planning and execution at a fine grain, or can re-plan mission sections on the fly in response to unexpected events. Combining this improved sensor performance with increased autonomy has enabled a first generation of fielded system to act responsively in tasks such as tracking and inspection of proud or buried cables, and the detection and characterization of mines, beyond being mine like objects. Second generation systems that use their autonomy to optimally take multiple looks and control the spectral content of pings in an environmentally adaptable way are under consideration.

8:40

2aUWb3. Bearings-only target localization for unmanned underwater vehicles. Donald P. Eickstedt (iRobot Corp., 8 Crosby Dr., Bedford, MA 01730, eicksted@mit.edu)

This paper reports on target localization for autonomous underwater vehicles (AUVs) with acoustic bearing sensors where the bearing sensor provides detection and beamforming capabilities and reports target bearings with the associated measurement uncertainties. Estimating the state of a moving target using a bearing sensor on a moving platform is a difficult problem due to both the nonlinear nature of the measurement equations with respect to the unknown target parameters but also the observability conditions for the unknown parameters which require the observer to maneuver. This paper will report on a maximum likelihood (ML) method using Levenberg–Marquardt (LM) optimization for estimating the target state parameters. A known issue with ML solutions to this problem is that the algorithm may have difficulty reaching the global minimum unless the initial solution guess is close to the true solution. This paper will report on a genetic algorithm approach to making the initial solution guess. The performance of this approach, obtained from MATLAB

simulation, is presented. The implementation of the algorithms in the MOOS architecture is described using a very fast LAPACK implementation for the LM parameter estimation to support the goal of optimal AUV maneuvering to improve the target state estimate.

9:05

2aUWb4. Adaptive ocean sampling for acoustic ocean uncertainty reduction. Kevin D. Heaney and Richard L. Campbell (Ocean Acoust. and Instrumentation Systems, Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

For many oceanic environments, acoustic propagation is critically dependent upon the local environment through boundary interaction and volume variability and fluctuations. In shallow water, near the continental shelves, ocean dynamics and variability can lead to significantly different propagation regimes on timescales of hours and spatial scales of kilometers. Optimal adaptive sampling, with assimilation of oceanographic variables into a dynamic ocean model, can be used to reduce the model forecast uncertainty and improve the confidence in acoustic predictions. Oceanographic uncertainty and acoustic sensitivity, however, are not always well correlated. In this paper, an acoustic uncertainty cost function will be used to drive the adaptive sampling of an oceanographic AUV in coastal water. Applications to an experiment off the coast of Taiwan (quantifying, predicting, and exploiting uncertainty) will be performed, with model-data comparisons of acoustic propagation for moving sources and drifting receivers. During the test, acoustic adaptive sampling computations were performed in real time, oceanographic measurements were made, assimilated into a dynamic ocean model and acoustic transmissions were recorded. [Work Supported by ONR.]

9:30

2aUWb5. Ambient noise modeling for high fidelity acoustic simulation. Andrew J. Poulsen (Appl. Physical Sci. Corp., 49 Waltham St., Ste. 4, Lexington, MA 02421, apoulsen@aphysci.com) and Henrik Schmidt (MIT, Cambridge, MA 02139)

A high fidelity acoustic simulator has been developed to enable realistic simulation of array time series in order to test sensing/processing algorithms. This simulator, capable of generating calibrated hydrophone or vector sensor array data, is fully integrated into a three-dimensional hydrodynamic array model, generating sensor time series for dynamically evolving array shape, location, and orientation. Furthermore, the simulator handles a changing number and configuration of acoustic sources, targets, and receivers while utilizing the legacy ray tracing code BELLHOP to account for ocean multipath effects. Properly modeling ambient noise is particularly important when determining the effect of noise on sensor/processing performance, e.g., array gain can vary significantly based on the directionality of ambient noise. Embedded in the simulator is the capability to efficiently generate broadband noise for an arbitrary noise intensity distribution as a function of depth and elevation angle (azimuthally symmetric ambient noise) using theoretical expressions for the array covariance structure. This approach is ideal for modeling surface noise in situations where the array covariance expressions need to be repeatedly recomputed to account for changing array shape. Results illustrate the advantages of the proposed approach for generating high fidelity time series to model real-world complexities in an efficient, elegant manner.

Contributed Papers

9:55

2aUWb6. Results of the development of a long-range acoustic homing system for autonomous underwater vehicles. G. J. Heard, N. Pelavas, C. E. Lucas, and R. Fleming (Defence RD Canada Atlantic, PO Box 1012, Dartmouth, NS, Canada B2Y 3Z7, garry.heard@drdc-rddc.gc.ca)

Modified international submarine engineering (ISE) explorer AUVs with an endurance of over 400 km are being used to aid in the mapping of the under-ice Arctic seafloor. The explorers are equipped with a DRDC-developed acoustic homing system built into the AUV nose cone. The acoustic receiver consists of seven digital hydrophones arranged in a tri-axis cross-dipole array. A controller/data processor located within the AUV pressure hull handles the real-time acoustic arrival azimuth and elevation estimation, as well as the control and calculations for an on-demand short-range three dimensional (3-D) localization system, and the control of vehicle telemetry data flow. The processor and array consume less than 2 W. A small, easily transportable, DRDC-designed acoustic transducer provides a powerful acoustic homing signal. Using the homing system, the vehicles were able to locate an acoustic beacon at a randomly drifting Ice Camp from a range in excess of 50 km (100-km ranges possible). The design, development, and use of the homing system are described in this paper. In addition, on-going software improvements providing enhanced capabilities and system miniaturization are described.

10:10–10:25 Break

10:25

2aUWb7. Interfaces between acoustic prediction, embedded signal processing, and behaviors at NATO Undersea Research Centre. Kevin D. LePage, Francesco Baralli, Robert Been, Ryan Goldhahn, Michael J. Hamilton, Stephanie Kemna, Michele Micheli, Juri Sildam, and Arjan Vermeij (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, SP Italy)

The use of acoustic sensing systems for ASW in heterogeneous sensor networks utilizing marine robots has been a subject of research at the NATO Undersea Research Centre for the past several years. In this talk, we discuss the unique challenges of implementing ASW on autonomous, collaborative networks of AUVs, including the challenges of embedding the active sonar signal processing, implementing effective underwater messaging, and designing adaptive behaviors to optimize system performance. Theoretical studies, simulations, and results from the recent GLINT series of sea trials are shown and the way forward for autonomous sensor system studies at NURC is discussed.

10:40

2aUWb8. Application of the range dependent waveguide invariant distribution processor to unintentionally radiated broadband noise from surface ships in a range- and azimuthally dependent environment. Alexander W. Sell and R. Lee Culver (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., Univ. Park, PA 16802, aws164@psu.edu)

Waveguide invariant analysis is a useful tool for understanding spectral interference patterns from broadband sources in shallow water waveguides. These interference patterns (in the form of intensity striations) were observed in time-frequency plots of surface ships passing a horizontal line array located along the continental shelf off southeast Florida during a 2007 acoustical experiment. Previous work has shown that results from the Range Dependent Waveguide Invariant Distribution (RaDWID) method match well with simulations from parabolic equation acoustical models and require significantly less computation time; however, work to understand the processor's ability to recreate real data was incomplete. We will discuss how RaDWID processing can be applied to ship-radiated broadband noise to model spectral interference patterns. The implementation of the processor on these data including handling of environmental parameter uncertainty, broadband source power spectrum, and environmental features will also be discussed. [Work supported by ONR Undersea Signal Processing.]

10:55

2aUWb9. Environmentally sensitive autonomous underwater vehicle (AUV) behavior design. Kevin D. LePage (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, SP Italy)

The design of environmentally adaptive behaviors for AUVs conducting active ASW is discussed within the framework of the MOOS-IvP middleware. The control of a single or a group of autonomous, collaborative AUVs in the underwater environment requires the installation of on-board autonomy to enable the vehicles to optimize their trajectories and strategies as they prosecute underwater targets. As the underwater environment is harsh and large differences in performance can be anticipated as a function of lateral position, speed, and depth, a suite of environmentally sensitive behaviors has been developed to provide the required autonomy. Results from simulation and at-sea experiments will be shown.

11:10

2aUWb10. Real-time sonar signal processing on-board an autonomous underwater vehicle. Michael Hamilton and Michele Micheli (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, Italy, hamilton@nurc.nato.int)

An active, bistatic signal processing system has been implemented for use on-board autonomous underwater vehicles (AUVs) using towed arrays. The NATO Undersea Research Centre's (NURC) AUVs are programmed to maneuver in order to best track a target. To perform this action autonomously, the vehicle must be able to fully process and track targets via its towed array data in real time or faster. The processor implemented includes processing from the array hydrophone data, through beamforming, matched filtering, Doppler processing, and tracking. This system adapts many previously developed algorithms to function in real time. Research and algorithm adaptations in the areas of normalization, CFAR detection, and array navigation, have also been developed to deal with practical issues which have arisen in sea trials. The implementation, practical issues, and steps taken to address them will be presented. This research is supported by the NURC Consolidated Programme of Work, Cooperative ASW program.

11:25

2aUWb11. Supervised machine learning for real time classification of underwater targets from sampled scattered acoustic fields. Erin M. Fischell and Henrik Schmidt (Ctr. for Ocean Eng., Dept. of Mech. Eng., MIT, 77 Mass. Ave., Bldg. 5-223, Cambridge, MA 02139)

Autonomous underwater vehicles (AUVs) are often used for bottom surveys in the ocean, but the collected data is usually processed on shore. The ability to identify and classify targets while in flight would allow more effective use of mission time and reduce the need for additional surveys. It is demonstrated that geometry can be classified from hydrophone sampling of the scattering field off of a target in the presence of an acoustic source in real time simulation. Scattering field data for multiple target shapes is simulated using the OASES and SCATT software packages. This data is then used to produce training data sets for a support vector machine (SVM), a type of supervised machine learning that generates a classifying hyperplane. The trained SVM classifies new data, such as that collected by an AUV in a real or simulated scattered field, with minimal computation by comparing it to the hyperplane. The SVM also identifies divergence between target classes, allowing identification of regions where targets will look the most different to sampling, enabling optimized path planning for classification. This would allow a vehicle looking for a particular geometry to shift its search pattern to best identify relevant features.

11:40

2aUWb12. Time-varying spatial spectrum estimation using a maneuverable sonar array. Jonathan L. Odom and Jeffrey L. Krolik (Dept. of Elec. and Comput. Eng., Duke Univ., PO Box 90291, Durham, NC 27708, jonathan.odom@duke.edu)

This paper addresses the problem of spatial spectrum estimation in dynamic environments with a maneuverable sensor array. Estimation of the time-varying acoustic field directionality is of fundamental importance in passive sonar. In this paper, mobility of the array is treated as a feature allowing for left-right disambiguation as well as improved resolution toward endfire. Two new methods for on-line spatial spectrum estimation are presented: (1) recursive maximum likelihood estimation using the EM algorithm and (2) time-varying spatial spectrum estimation via derivative-based updating. A multi-source simulation is used to compare the proposed algorithms in terms of their ability to suppress ambiguous array backlobes. A broadband method is presented utilizing knowledge of the source temporal spectrum. Detection performance of weak high-bearing rate sources in interference-dominated environments is evaluated for a flat spectrum. [This work was supported by ONR under grant N000140810947.]

Meeting of the Standards Committee Plenary Group
to be held jointly with the meetings of the
ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:
ISO/TC 43, Acoustics,
ISO/TC 43/SC 1, Noise,
ISO/TC 108, Mechanical vibration, shock and condition monitoring,
ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures,
ISO/TC 108/SC 3, Use and calibration of vibration and shock measuring instruments,
ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machines,
ISO/TC 108/SC 6, Vibration and shock generating systems,
and
IEC/TC 29, Electroacoustics

P. D. Schomer, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43 Acoustics and ISO/TC 43/SC 1 Noise
Schomer and Associates, 2117 Robert Drive, Champaign, IL 61821

D. J. Evans, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108 Mechanical vibration, shock and condition monitoring, and ISO/TC 108/SC 3 Use and calibration of vibration and shock measuring devices
National Institute of Standards and Technology (NIST), 100 Bureau Drive, Stop 8220, Gaithersburg, MD 20899

W. C. Foiles, Co-Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
BP America, 501 Westlake Park Blvd., Houston, TX 77079

R. Taddeo, Co-Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
NAVSEA, 1333 Isaac Hull Ave., SE, Washington Navy Yard, Washington, DC 20376

D. D. Reynolds, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock
3939 Briar Crest Ct., Las Vegas, NV 89120

D. J. Vendittis, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
701 Northeast Harbour Ter., Boca Raton, FL 33431

R. Taddeo, Vice Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
NAVSEA, 1333 Isaac Hull Avenue, SE, Washington Navy Yard, Washington, DC 20376

C. Peterson, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 6 Vibration and shock generating systems
200 Dixie Ave., Kalamazoo, MI 49001

V. Nedzelnitsky, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
National Institute of Standards and Technology (NIST), Sound Building, Room A147, 100 Bureau Dr., Stop 8221, Gaithersburg, MD 20899-8221

The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting.

The meeting of the Standards Committee Plenary Group will precede the meetings of the Accredited Standards Committees S1, S2, S3, S3/SC 1 and S12, which are scheduled to take place in the following sequence:

Tuesday, November 1, 2011	10:30 a.m.–11:30 a.m.	ASC S2, Mechanical Vibration & Shock
Tuesday, November 1, 2011	1:00 p.m.–2:00 p.m.	ASC S1, Acoustics
Tuesday, November 1, 2011	2:15 p.m.–3:30 p.m.	ASC S12, Noise
Tuesday, November 1, 2011	3:45 p.m.–5:00 p.m.	ASC S3/SC 1, Animal Bioacoustics
Wednesday, November 2, 2011	8:30 a.m.–9:45 a.m.	ASC S3, Bioacoustics

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.
The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3 and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. Parallel Committee
ISO		
P.D. Schomer, Chair	**ISO/TC 43** Acoustics	ASC S1 and ASC S3
P.D. Schomer, Chair	**ISO/TC 43/SC1** Noise	ASC S12
D.J. Evans, Chair	**ISO/TC 108** Mechanical vibration, shock and condition monitoring	ASC S2
W.C. Foiles, Co-Chair R. Taddeo, Co-Chair	**ISO/TC 108/SC2** Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures	ASC S2
D.J. Evans, Chair	**ISO/TC 108/SC3** Use and calibration of vibration and shock measuring instruments	ASC S2
D.D. Reynolds, Chair	**ISO/TC 108/SC4** Human exposure to mechanical vibration and shock	ASC S3
D.J. Vendittis, Chair R. Taddeo, Vice Chair	**ISO/TC 108/SC5** Condition monitoring and diagnostics of machines	ASC S2
C. Peterson, Chair	**ISO/TC 108/SC6** Vibration and shock generating systems	ASC S2
IEC		
V. Nedzelnitsky, U.S. TA	**IEC/TC 29** Electroacoustics	ASC S1 and ASC S3

TUESDAY MORNING, 1 NOVEMBER 2011 — CRESCENT, 10:30 TO 11:30 A.M.

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

A. T Herfat, Chair ASC S2

Emerson Climate Technologies, Inc., 1675 West Campbell Rd., PO Box 669, Sidney, OH 45365-0669

C. F. Gaumond, Vice Chair ASC S2

Naval Research Laboratory, Code 7142, 4555 Overlook Ave. SW, Washington DC 20375-5320

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration 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 108, Mechanical vibration, shock and condition monitoring, and its five subcommittees, 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 S2: Standards, specification, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance and comfort.