Session 2aAA

Architectural Acoustics: General Topics in Architectural Acoustics

Neil A. Shaw, Chair
Menlo Scientific Acoustics, Inc., P.O. Box 1610, Topanga, California 90290-1610

Contributed Papers

8:45

2aAA1. The ancients have stolen our inventions—Recently discovered documents of Vern O. Knudsen. Neil A. Shaw (Menlo Sci. Acoust., Inc., P.O. Box 1610, Topanga, CA 90290-1610, menlo@ieee.org) and Charlotte B. Brown (UCLA Library, Los Angeles, CA 90095-1575, cbbrown@library.ucla.edu)

Vern O. Knudsen was a founder of the Acoustical Society of America (1928), a member of the Bell Telephone Laboratories staff (1918), a respected professor of physics at UCLA (1922–1958), a Chancellor of UCLA (1959–1960), and a consultant in acoustics. Recently, a trove of Dr. Knudsen's papers and books was gathered by his family and donated to the UCLA Dept. of Special Collections where they will be added to the existing 44 boxes of the Vern Oliver Knudsen Papers (UCLA Manuscript Collection No. 1153). These newly acquired documents and architectural drawings illustrate Dr. Knudsen's pioneering work in classroom acoustics, room acoustic design, and sound propagation–aspects of acoustics that are still investigated today, 80 years later. The presentation will include a review of selected items from the Knudsen Papers including: his work for clients such as the Hollywood Bowl (1926–1929) and MGM Studios; his work with architects in the design of classrooms, performance and production spaces; Dr. Knudsen's basic physical research; and his correspondence with luminaries from numerous fields. Many in the audience will rediscover that they do, indeed, follow in the footsteps of a giant.

9:00

2aAA2. Rock art acoustics. Lauren M. Ronsse (Univ. of Kansas, 1312 Louisiana St., Lawrence, KS 66044, lronsse@ku.edu)

The relationship between the location of rock art and the acoustical properties of its immediate environment has been a source of previous investigation. The markings the early peoples created on rock formations open a fascinating portal into the exploration of their lifestyles. Previous research has shown that often ancient rock art was placed on surfaces or in locations that echoed, whereas locations without such echoes were undecorated [S. J. Waller, 2002 Rock Art Acoustics in the Past, Present, and Future. 1999 International Rock Art Congress Proceedings **2**, 11–20]. Could the acoustical characteristics of the decorated outcroppings have been perceived by the early peoples of the American Plains? A detailed study of six rock art sites located in Ellsworth County, KS has been conducted to determine the acoustical properties of these sites. At each site, various impulsive sounds were created to energize the space. The impulse responses were recorded and analyzed using two acoustic recording and analysis computer programs. This study did find echoes occurring at the decorated sites. These echoes were quantifiably louder than any reflection of sound measured at the surrounding undecorated locations.

9:15

2aAA3. Acoustics at the shrine of St. Werburgh. David Lubman (Acoust. Consultant, 14301 Middletown Ln., Westminster, CA 92683-4514)

England's Chester Cathedral (Anglican) contains a shrine to its patron saint, St. Werburgh, a 7th century Saxon princess of Mercia who became a nun and abbess. The 8th or 9th century shrine is far older than the 16th century cathedral. The shrine was enlarged around 1340, apparently because of its popularity as a place of pilgrimage and reported miracles. It was smashed by 16th century Henrician reformers and restored in the 19th century. The present shrine contains six recesses. In pre-Reformation times, kneeling pilgrims placed their heads in recesses to deliver spoken petitions to the saint. Analyses of binaural recordings made in a recess reveal that vocalizations are dramatically reinforced (10.4 dBA, 15.5 dB overall) and distorted by resonances (the strongest is 25.1 dB @ 125 Hz). Documented acoustical features suggest the shrine comprises a forgotten or previously unknown form of religious theater-for-one in which the sound experience was important. Modern listeners can enjoy a pre-Reformation auditory virtual reality experience via binaural recordings made at the shrine (best with quality headphones). [The author gratefully acknowledges architect Peter Howell of the Dog Rose Trust, a registered English charity, who reported the unusual shrine acoustics; and the cooperation of Chester Cathedral authorities.]

9:30

2aAA4. The penguin and puffin coast: Application of a micro-perforated membrane stretch ceiling. Jeff B. Pride and Jason T. Weissenburger (Eng. Dynam. Intl., 8420 Delmar Blvd., Ste. 303, St. Louis, MO 63124, edi@edi-stl.com)

The St. Louis Zoo has been undergoing major renovations that make it one of the premier zoos in the world. The Penguin and Puffin Coast, a walk through exhibit, is as near to a perfect habitat reproduction as is technologically feasible. Reproduced are a rugged coastline with craggy rock outcroppings, frigid water and a barrel vault ceiling that has theatrical lighting that can be used to simulate a colorful sunrise, a sunset over the horizon or the reversed seasons. Sounds of crashing waves and a sea lion's bark can be heard in the distance. A micro-perforated membrane stretch ceiling was chosen for its ability to easily conform to the undulations required for the sky, its durability in the moist confines of the exhibit and, last but not least, its ability to provide much needed acoustical absorption, all with the same product. This talk will discuss experiences with the micro-perforated material, review the acoustical efficacy of the installation and show photographs of the construction from essential beginning to final product. A photo or two of the penguins and puffins will also be shown.

9:45

2aAA5. Influence of surface scattering characteristics on the sound quality of reverberation. Jacob Mueller, Mendel Kleiner, Ning Xiang, and Rendell Torres (Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY 12180)

The influence of the scattered sound from different sound diffusing surfaces on the subjective characteristics of the late reverberation of a room was studied using a simulation approach. The impulse response of various scale model scatterers was measured for a set of angles and used in combination with a simple room model for auralization. As shown by Kleiner one can quite easily hear the differences between the scattered sound from different scatterers [M. Kleiner *et al.*, Proc. 93rd Audio Eng. Soc. Convention, San Fransisco. Vol. **43**, "Auralization of QRD and Other Diffusing Surfaces using Scale Modelling" (1992)]. The results obtained here, for the late reverberation, are not as clear, indicating that the "individual" sound of scattering surfaces will influence primarily the sound quality of the early part of the reverberation. [Work supported by RPI.]

10:00–10:15 Break

10:15

2aAA6. Measured effects of diffusers and absorbers on the low-frequency modal structures in two reverberation chambers. David Nutter, Micah Shepherd, Timothy Leishman, and Benjamin Shafer (Acoust. Res. Group, Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT 84602, dave_nutter@hotmail.com)

The Brigham Young University Acoustics Research Group has recently constructed two rectangular reverberation chambers for its research efforts. Steps have been taken to qualify the chambers for a variety of applications. Stationary diffusers and low-frequency absorbers were designed and installed as part of this process. This paper presents measurements taken at various stages of the installation to assess the impact of the diffusers and absorbers on the modal structure of the low-frequency fields.

10:30

2aAA7. Use of surrogate samples to study variation of diffuse field absorption coefficients of fiberglass with altitude. Richard D. Godfrey (Owens Corning, Sci. & Technol., 2790 Columbus Rd., Granville, OH 43023)

ASTM C 423 identifies air temperature and relative humidity as significant parameters, but does not address air density effects. In previous papers, normal and diffuse field analysis showed significant changes in predicted absorption coefficients with altitude. These predictions were validated experimentally for normal incidence in a vacuum chamber, and by using surrogate samples, thus showing the feasibility to study altitude effects in a single laboratory. Mechel design charts are normalized by two parameters. One is not dependent on air density. The other (R) is the ratio of flow resistance and the impedance of air. At constant thickness, the effect of lowering air density can be studied by increasing the sample flow resistivity. Samples with flow resistivity ratios of 1.25 and 1.5 were studied in a diffuse field following ASTM C 423 methodology. These values correspond to altitudes of 6000 and 10 700 ft in altitude, respectively. These results followed the predicted trends. At both altitudes, samples with R's of 2 and 4 had higher average absorption coefficients. For $R = 8$, the reverse effect was observed. It was concluded that the effects of altitude are not limited to the impedance tube method, but are also present in the reverberation room method.

10:45

2aAA8. Prediction of sound transmission loss of honeycomb sandwich panel by higher order approach. Tongan Wang, Shankar Rajaram, and Steven R. Nutt (Mater. Sci. Dept., Gill Foundation Composite Ctr., Univ. of Southern California, 3651 Watt Way, VHE602, Los Angeles, CA 90089)

People have studied the sound transmission loss (STL) of sandwich panels since the 1970s. However, most of the existing prediction methods have been based on single-layer dynamical models, neglecting the symmetric (dilatational) movements of the skins. Consequently, the symmetric coincident frequency of the sandwich panel cannot be predicted using those approaches. To account for this dilatational motion of the sandwich structures, different methods were utilized. However, most of them were based on one dimensional sandwich beams theories. The authors have also applied the consistent higher order beam approach to calculate the sound transmission loss of a unidirectional sandwich panel. Although the one dimensional approximation is good in predicting STL, the effects of some factors, such as the anisotropy and orientation of the principle axis of the panel, cannot be estimated. In the current work, the authors extended that one dimensional beam model into two dimensions, which allows us to calculate the STL of sandwich plates with composite laminate face sheets and honeycomb core. Both flexural (antisymmetric) and dilatational (symmetric) motions of the sandwich panel were considered in the study. The predictions were finally compared with our experimental data.

11:00

2aAA9. Measurement of transmission loss trends for orthotropic sandwich panels at a subscale facility. Shankar Rajaram, Tongan Wang, and Steve Nutt (Dept. of Mater. Sci., Univ. of Southern California, 3651 Watt Way, VHE-602, Los Angeles, CA 90089)

Interior noise studies in airplanes have identified floors as one of the primary noise paths. The acoustic barrier properties of floor panels are typically quantified by sound transmission loss (STL) measurements. A subscale transmission loss suite consisting of a reverberant room and an anechoic room was constructed and qualified to study the relative transmission loss trends of orthotropic sandwich panels used in airplane floors. A host of material combinations accounting for a variety of mechanical properties were tested for their acoustic performance. The transmission loss measurements were based on a sound intensity technique (ASTM E 2249-02) and a sound pressure technique (SAE J 1400-90). The results from these two techniques were compared to results from a full-scale accredited facility using the two-reverberation room method (ASTM E 90-02). The STL trends for orthotropic sandwich panels from the sub-scale facility were comparable to trends from the full-scale facility between 315 and 5000 Hz. The measurements permitted a ranking of several standard floor panels with respect to acoustic performance. [Support for this research from Mervyn C. Gill Foundation is gratefully acknowledged.]

11:15

2aAA10. Noise control computer modeling for architectural academic education. Michael Salameh (712 E. St. Andrews, Midland, MI 48642, ms@tir.com)

Commercial software is widely used in professional acoustical design and consultation. However, there are few educational computer programs that are designed mainly for academic teaching of architectural acoustics. Some of these computer programs, such as ACOUSTIC2D and ACOUSTIC3D [J. Turner and N. Barnett, University of Michigan], do not include noise control modeling. A new educational program, NOISE CONTROL 2D, is presented. The approach used in the modeling of noise control concepts as well as the design criteria related to this program are explored. The visual user interface and the ability to study the impact of various parameters on room noise are demonstrated. These parameters include architectural factors such as the architectural layout and acoustical parameters such as transmission losses of the room boundaries. Many aspects of noise control in buildings are covered in the program, including noise levels, reverberant and direct sound, NC, NCB, STC, and composite STC.

11:30

2aAA11. Acoustic3D Teaching Program. Norman E. Barnett and James A. Turner (Taubman College of Architecture and Urban Planning, 2000 Bonisteel Blvd., Ann Arbor, MI 48109)

A computer program has been under development for several years to assist the teaching of architectural acoustics to architecture students. This program is an outgrowth of a 2D program used routinely by the students since 1999. Emphasis is on where, when and how much sound distributes within a geometrical envelope. The initial geometry of a space is drawn or modeled with commercial programs that students typically would use to represent a design. That geometry is exported in the dxf file format and imported into the acoustic program. There it is combined with at least one sound source and initial sound reflection characteristics representing the room surfaces. As such, an acoustical simulation is produced that can be manipulated to reveal the implicit acoustical behavior of the representation. A description is given about how the program is organized and how it functions. Two brief examples are provided. The first is for a simple rectangular schoolroom such as might be used to develop some introductory learning about architectural acoustics. The second is for a more complicated geometry more appropriate for students who are reasonably well grounded in acoustics. They are presumably ready to undertake some acoustical investigation at a design development level.

Session 2aAB

Animal Bioacoustics: Marine Mammal Acoustics: Session in Honor of Ron Schusterman I

Whitlow W. L. Au, Chair
Hawaii Institute of Marine Biology, P.O. Box 1106, Kailua, Hawaii 96734

Chair's Introduction—7:55

Invited Papers

8:00

2aAB1. Dr. Ronald Schusterman's contributions to national acoustic policy. Roger L. Gentry (NOAA Fisheries, 1315 East West Hwy., Silver Spring, MD, roger.gentry@noaa.gov)

When the effects of underwater noise on marine mammals became a national issue in the 1990's, the federal government needed people trained in those fields to manage programs. No trained managers with that background existed, so a few scientists took on management roles. Of today's five Washington, DC managers in this issue, three received advanced degrees under Ron Schusterman. One started ONR's comprehensive research program on this topic, and two run NMFS's regulatory policy program on acoustics. Dr. Schusterman intended his students to work in basic science, specifically cognition, learning, sensory perception, and social behavior of marine mammals. Ironically, and despite his aversion to bureaucracy, this background equipped his students to become government decision makers. Such are the twists of history. He also taught his students to rigorously apply the scientific method and to assume nothing. If his science castaways in Washington, DC are worthy of his training, the national noise issue will be the better for it.

8:20

2aAB2. Audiology to ecology: Auditory scene analysis goes underwater. Robert Gisiner (ONR, 800 N. Quincy St., Arlington, VA 22217)

For decades the study of marine mammal audiology was considered an arcane branch of comparative psychophysics with little to offer the marine mammal ecologist. But in the past decade the oval window of the marine mammal ear has effectively become the ecologist's window into a marine ecosystem that truly is a world of sound. We are learning to listen to the ocean as marine mammals do, in order to better understand an environment long hidden from our view. And we are learning about the importance of sound to marine mammals, including the effects of our own noisy entry into their world. The past decade of revolutionary change in the use of sound to study marine mammals, and the associated revolution in our appreciation of marine mammal uses of sound, will be reviewed. We now see the ocean world through a marine mammal's ears, thanks to Ron Schusterman and a few dedicated colleagues who have opened our eyes, and ears, to the importance of audiology in ecology.

8:40

2aAB3. Studying social cognition in marine mammals. Peter Tyack (Biol. Dept., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, ptyack@whoi.edu)

Ron Schusterman has played an important role in broadening the perspective of marine mammal cognition studies from narrow comparisons to human language to more general cognitive concepts. He has also contributed to our understanding of learning mechanisms for individual recognition in pinnipeds, linking naturalistic observations with controlled studies in captive settings. I discuss how odontocetes learn to develop signals for individual and group recognition. Bottlenose dolphins use vocal learning to develop individually distinctive whistles in the first 1–2 years of life, but they also maintain the ability to imitate whistles throughout their lifetime. As maturing males form a coalition, their whistle repertoires converge. Species with more stable groups than dolphins use vocal learning to develop repertoires that are group distinctive. Schusterman has recently developed theoretical approaches to thinking about how animals form categories of social knowledge such as coalition or group. Depending upon the social context, animals that modify their vocalizations based upon auditory input and social relationships may use similar vocal learning mechanisms to develop quite different vocal repertoires. I will discuss the interaction between communication, social knowledge, and cognition in marine mammals from the approaches suggested by Schusterman for the study of social knowledge.

9:00

2aAB4. How acoustic signals become meaningful to listeners: An experimental approach. Colleen Reichmuth Kastak, Kristy Lindemann, and Ronald Schusterman (UCSC Long Marine Lab., 100 Shaffer Rd., Santa Cruz, CA 95060)

Most models of animal acoustic communication describe how vocal cues produced by a signaler influence the behavior of a listener. The response made by a listener depends in large part on the perceived meaning of the signal. But, how do signals become meaningful to listeners? In some cases, such as imprinting, signal meaning can be attributed to structural cues that are perceived and acted upon through an innate releasing mechanism. In other instances, signals may be arbitrarily related to objects, individuals, or species. Equivalence theory provides a model describing how some arbitrary signals may acquire meaning. Here, we describe theory and experimental evidence in the form of cross-modal matching-to-sample tasks showing how acoustic signals can become referents

for visual stimuli. The subject of these behavioral experiments is a California sea lion with extensive experience in performing associative learning tasks. The aim of the experiments is to establish multiple auditory-visual discriminations and then test for the emergence of untrained relationships between disparate visual stimuli linked by a common auditory signal. Preliminary data show successful emergent matching across visual and auditory modalities. These findings suggest that acoustic signals become meaningful to listeners when learned associations lead to the formation of equivalence classes.

9:20

2aAB5. White whale echolocation pulses in the open sea at the surface and at depth. Sam Ridgway and Don Carder (U.S. Navy Marine Mammal Program, Space and Naval Warfare Syst. Ctr., San Diego, 53560 Hull St., San Diego, CA 92152-5001)

Previously we reported on the first ever hearing tests of trained cetaceans in the open ocean demonstrating that zones of audibility for sound were just as great throughout the depths to which white whales dive, down to at least 300 m. The tests also showed that the whale's response whistles changed with increasing depth, overall amplitude decreased and frequency emphasis shifted higher with increasing depth from 5 to 300 m. Subsequently a door was installed in the test apparatus and the whales were taught to whistle in response to the presence of a small cylindrical target 2 m away. When the door opened the whales would utter a train of pulses and then whistle if the target were present. There was no statistical difference in echolocation pulse frequencies or amplitudes between depths of 5, 100, 200, and 300 m. Surprisingly, all pulses recorded at the open ocean test site had peak frequencies between 4 and 40 kHz. These differed markedly from pulses recorded with the same cable and apparatus in San Diego Bay where the whale's pulses usually exhibited two peaks, one in the 30–80 kHz range and the other often around 100–120 kHz.

9:40

2aAB6. Hearing loss and echolocation signal change in dolphins. Patrick W. Moore, James Finneran (SPAWARSYSCEN San Diego, Code 2351, 53560 Hull St., San Diego, CA 92152-5001), and Dorian S. Houser (BIOMIMETICA5750, La Mesa, CA 91942)

Recent studies and ongoing research have shown that echolocating dolphins can change the structure of their emitted echolocation signals during active echo-investigation of targets. The presumption has been that the animal adjusts various parameters (source level, peak frequency, etc.) of the emitted signal to maximize the information return in the target echo as a function of task or environmental constraints and requirements. Other work has suggested that the frequency range over which this dynamic control is exerted may change due changes in the animals hearing ability. Specifically, dolphins that develop high frequency hearing loss, for example from age, noise exposure or ototoxic drugs, shift the center frequency of the emitted echolocation click to lower frequency ranges. Observations of several Navy Marine Mammal Program animals with known high frequency hearing loss have demonstrated these frequency shifts. In this paper we will elaborate and extend ongoing analysis of emitted echolocation signals of several dolphins that show hearing loss associated changes in emitted signal structure, discuss the implications of these measures and suggest approaches that may prove useful for evaluating basic hearing capabilities from collected echolocation signals.

10:00–10:10 Break

10:10

2aAB7. Relationship between auditory evoked potential (AEP) and behavioral audiograms in odontocete cetaceans. Dorian S. Houser (BIOMIMETICA, 7951 Shantung Dr., Santee, CA 92071), James J. Finneran, Donald A. Carder, Sam H. Ridgway, and Patrick W. Moore (SPAWARSYSCEN San Diego, San Diego, CA 92152)

Auditory evoked potentials (AEPs) offer an alternative to behavioral methods of determining auditory sensitivity in marine mammals. The technique can be performed without the need for animal training, substantially expediting the process, and has the potential for application to stranded and rehabilitating marine mammals, thus providing an opportunity to determine hearing sensitivity in animals not likely to be kept in captivity. As an emerging technology in the field of marine mammalogy, the equivalence of AEP and behavioral thresholds remains to be quantitatively assessed. Human and laboratory animal AEPs are typically -5 to $+20$ dB of behaviorally determined thresholds and vary by technique and frequency tested. To be an effective tool in the field of marine mammalogy, the expected variation in AEP thresholds relative to behavioral thresholds in marine mammal species needs to be determined. We compare the behavioral and AEP audiograms of several odontocetes covering a range of normal hearing to profound hearing loss and demonstrate the offsets between results obtained with the two methods. Thresholds determined by the two methods show generally good agreement and demonstrate the utility of AEPs as an emerging technology in the study of marine mammal audiometry.

10:30

2aAB8. Acoustic basis for fish prey selection by echolocating odontocetes. Whitlow W. L. Au, Kelly J. Benoit-Bird (Hawaii Inst. of Marine Biol., Univ. of Hawaii, P.O. 1106, Kailua, HI 96734), Ronald Kastelein, and Sander van de Heul (SEAMARCO, 3843 CC Harderwijk, The Netherlands)

Acoustic backscatter data were obtained from four fish species, sea bass (*Dicentrarchus labras*), pollack, (*Pollachius pollachius*), grey mullet (*Chelon labrosus*), and Atlantic cod (*Gadus morhua*), using broadband bottlenose dolphin and narrow-band harbor porpoise signals. The fishes were anesthetized and attached to a monofilament net that was in turn attached to a rotor so echoes could be collected along the lateral axis of each fish. The echo waveforms were complex with many highlights and varied with the orientation of the fish. The highlight structure was determined by calculating the envelope of the cross-correlation function between the incident signal and the echoes. The strongest echo occurred when the incident angle was perpendicular to the long axis of the swim bladder, however, the number of highlights was the fewest at this perpendicular orientation and increased as the fish orientation moved away from the perpendicular aspect. The echo structures were easily distinguishable between species and were generally consistent within species. The highlight structure of the echoes resulted in the spectrum being rippled, with local maxima and minima at different frequencies. However, differences in species were more obvious with the broadband dolphin signal than the narrow-band porpoise signal which had a much lower spatial resolution.

10:50

2aAB9. Hey Ron manatees do not echolocate either; the underwater hearing and acoustical behavior of West Indian manatees. Edmund Gerstein, Laura Gerstein (Leviathan Legacy Inc., 1318 SW 14th St., Boca Raton, FL 33486), Steve Forsythe (Naval Undersea Warfare Ctr., Newport, RI 02841), and Joseph Blue (Leviathan Legacy Inc., Boca Raton, FL 33486)

A comprehensive series of underwater psychoacoustic tests were conducted with captive manatees to measure their hearing abilities under varying acoustic conditions. Forced-choice paradigms with either a staircase, or method of constants, psychometric were used to define their audiogram, critical ratios, temporal integration, and directional hearing abilities. Pure tones, complex, broad and narrow-band noise were presented with a masker, at different intensities, to measure simultaneous masking effects at ambient levels recorded in manatee habitats. Masked thresholds across frequencies increased linearly with masker intensity. Critical ratios for pulsed signals were lower than nonpulsed, suggesting an inhibitory process affecting perception of nonpulsed tones. Comparisons with other mammals indicate manatees have acute filtering abilities for detecting pulsed sounds. While manatees do not exhibit a vocal repertoire to account for acute filtering, they are passive listeners, well adapted to selectively filter out continuous noise in favor of biologically significant sounds like their own 200-ms calls. Playbacks of band-limited calls suggest loudness summation across multiple critical bands may enable manatees to detect and locate their calls near or below ambient levels. This may explain why calibrated calls recorded in the wild exhibit no Lombard shifts. Low source levels and pulse rates negate their utility for active echolocation.

11:10

2aAB10. Underwater hearing thresholds in pinnipeds measured over a 6-year period. Brandon L. Southall (Long Marine Lab., Univ. of California, Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060 and NOAA Fisheries Acoust. Program), Ronald J. Schusterman, David Kastak, and Colleen Reichmuth Kastak (Univ. of California, Santa Cruz, CA)

While absolute hearing thresholds have been obtained for some marine mammals, few published data are available on how measurements of individual auditory sensitivity may change over relatively long periods of time. Studies that have investigated temporal changes in sensitivity have typically focused on animals in which differences in hearing are anticipated (age-related hearing loss). This study investigated the replicability of underwater hearing thresholds in prime-aged individuals of three pinniped species over a 6-year period. Aside from their age and experience with behavioral signal detection tasks, test subjects were of similar physical condition throughout this experiment. They were tested in the same enclosure at similar test frequencies (0.1–6.4 kHz) using identical methodology and criteria. Underwater hearing thresholds obtained throughout this testing period were not significantly different. These data indicate that underwater hearing sensitivity may remain relatively stable over long periods in nonsenescent marine mammals, including those regularly exposed to noise. Further, our results suggest that variability in testing equipment and experimental personnel may have little impact on behavioral hearing data, as long as similar testing methodologies and subject response bias are carefully maintained.

Contributed Papers

11:30

2aAB11. Testing the acoustic prey debilitation hypothesis: No stunning results. Kelly Benoit-Bird (College of Oceanic and Atmospheric Sci., Oregon State Univ., 104 Ocean Admin. Bldg., Corvallis, OR 97331), Whitlow Au (Hawaii Inst. of Marine Biol., Kailua, HI 96734), Ronald Kastelein, and Sander van de Huel (SeaMarco, 3843 CC Harderwijk, The Netherlands)

We examined the hypothesis that sounds produced by odontocetes can debilitate fish by testing the effects of three odontocete-like pulsed signals on three individuals of each of three fish species: sea bass, cod, and herring. We used a high-frequency click with a center frequency of 120 kHz exposing the fish to approximately 112 dB, a mid-frequency click with a center frequency of 70 kHz and 208 dB exposure level, and a low-frequency click with a center frequency of 40 kHz and 193 dB exposure level. Individual fish were placed in a 0.3-m-diam net enclosure immediately in front of a transducer. Each fish was allowed to remain in the experimental set up for at least 3 min prior to exposure to the clicks which were presented at a rate of 100 pulses/s grading to 700 pulses/s in 1.1, 2.2, and 3.3 s. Sea bass were also exposed to a constant pulse rate of 700 pulses/s for exposures of up to 30 s. No effect was observed in any of the fish for any signal type or pulse modulation rate. Based on our results, the hypothesis that acoustic signals of odontocetes alone can disorient or stun prey cannot be supported.

11:45

2aAB12. Auditory brainstem response hearing measurements in free-ranging bottlenose dolphins (*Tursiops truncatus*). Mandy L. H. Cook (USF College of Marine Sci., 140 Seventh Ave. S., St. Petersburg, FL 33701-5016), Randall S. Wells (Mote Marine Lab., Sarasota, FL 34236), and David A. Mann (USF College of Marine Sci., St. Petersburg, FL 33701-5016)

Bottlenose dolphins (*Tursiops truncatus*) rely on sound for communication, navigation, and foraging. Both natural and anthropogenic noise in the marine environment could mask the ability of wild dolphins to detect sounds, and chronic noise exposure could cause permanent hearing loss. The hearing abilities of a wild population of bottlenose dolphins in Sarasota Bay, FL are being investigated to determine whether they suffer hearing losses in comparison to animals living in quieter environments. This study is the first to measure the hearing sensitivity of a large population of wild dolphins that are exposed to significant levels of noise. Data on hearing sensitivities at frequencies used for acoustic communication (5–20 kHz) and echolocation (20–100 kHz) are reported. Hearing sensitivity was measured in the field using the non-invasive auditory brainstem response (ABR) procedure. ABR responses were evoked by the presentation of amplitude-modulated (AM) tones (carrier frequencies of 5, 10, 20, 30, 60, and 80 kHz) through a jawphone. The tones were modulated at 600 Hz, which elicited a robust envelope following response. A rapid ABR procedure was employed so that an entire audiogram could be obtained in approximately 30 min. This study also provides baseline data for longitudinal hearing studies in known individuals.

Session 2aAO

Acoustical Oceanography and Underwater Acoustics: Acoustic Sensing of Internal Waves I

James F. Lynch, Chair
Woods Hole Oceanographic Institute, 203 Bigelow Building, Woods Hole, Massachusetts 02543

Chair's Introduction—8:25

Invited Paper

8:30

2aAO1. Acoustic effects in presence of internal solitons in shallow water. Boris Katsnelson (Voronezh Univ., 1, Universitetskaya sq., Voronezh, 394006, Russia)

The given lecture (mini-tutorial) is dedicated to description of recent efforts (last 10–15 years) in experimental observation and theoretical modeling of the sound propagation in shallow water in the presence of traveling internal solitonlike waves (ISW). It contains a review of the main experimental observation of acoustical effects (Rubinstein and Rubinstein and Brill, Zhou *et al.*, Badiey *et al.*, etc.) The main theoretical approaches can be divided in dependence on orientation of ISW wave front: (1) Modes coupling due to ISW, traveling along acoustic track. Resonance effects in sound propagation, resonance coupling by packets and separate solitons (J-X Zhou, Preisig and Duda, etc.) are considered. (2) Horizontal refraction (HR) caused by ISW crossing acoustic track. Technique of vertical modes and horizontal rays, vertical modes and PE in horizontal plane and modeling of acoustic effects in typical conditions of Barents sea are shown. Analysis of specific peculiarities due to HR (synchronicity, periodicity, and depth dependence of intensity fluctuations) is given. Experimental observations of acoustic effects in the SWARM'95 (broadband sound propagation, shot and LFM signals) are presented. New specific features of signals, passing through ISW space-frequency horizontal structure, and fluctuations of modal spectrum are demonstrated, and experimental setup to register these is discussed. [Work was supported by RFBR and CRDF.]

9:30–9:45 Break

Contributed Papers

9:45

2aAO2. Focusing effects due to solitons: 3D Gaussian beam modeling. Paul Hursky, Michael B. Porter (Ctr. for Ocean Res., SAIC, San Diego, CA 92121), and Brian J. Sperry (SAIC, McLean, VA 22102)

As is well known, the SOFAR channel in deep water reduces spherical spreading, to cylindrical spreading, allowing sound to propagate to enormous distances. Similarly, soliton packets can produce channels in shallow water, forming acoustic corridors, their walls consecutive solitons. As the solitons pass over a propagation path they can generate dramatic focusing and defocusing effects. Acoustic modeling of such phenomena is challenging in that 3D (horizontal refraction) effects are clearly important (Nx2D approaches fail here). The geotime evolution is equally important—we must model a series of frozen oceans as the soliton packet passes by. Gaussian beam tracing models are ideally suited for such 4D modeling. We have developed a MATLAB Gaussian beam-tracing model to address these problems. It includes capabilities for a variety of useful beam options, from "geometric beams" to the most formal beam theory based on paraxial approximations. The latter is implemented using a novel "reduced delta-matrix formulation" that greatly simplifies the algorithms. The new model also allows for broadband calculations, 3D bathymetry, and 3D oceanography. We will discuss a variety of applications, with particular emphasis on the effects of solitons.

10:00

2aAO3. Evolution of internal soliton groups. Lev A. Ostrovsky (Zel Technologies and Univ. of Colorado, 325 Broadway, R./ET0, Boulder, CO 80305), Konstantin A. Gorshkov, and Irina A. Soustova (Russian Acad. Sci., Nizhny Novgorod 603095, Russia)

As known, acoustic wave propagation through a group of internal solitary waves (a solibore) in a coastal zone has a number of peculiarities; among them is a possible damping of sound upon passing a periodic group of solitons when the group period resonates with the interference distance of acoustic modes. However, solibores are typically not periodic, and the distance between solitons and possibly their order in the group can vary upon the onshore propagation. In this presentation, an evolution of a multisoliton group is considered in the framework of an evolution Gardner equation that takes both quadratic and cubic nonlinearity into account. For that, a perturbation method is used which allows the description of solitons as compounds of interacting fronts-kinks, and reduces the problem to a set of ordinary differential equations. The results are applied to strong solitons observed near the Oregon coast in 1995 where the same wave group was registered at two sites separated by 20 km. Although nonperiodic, the group retains its quasiperiodicity with a characteristic scale of order 1 km, which can affect sound propagation.

10:15

2aAO4. Uncertainty due to chaotic effects of internal waves in shallow water. Robert R. Luter, Jr. and Brian La Cour (Appl. Res. Labs., Univ. of Texas., Austin, TX 78713, rluter@arlut.utexas.edu)

Acoustic ray propagation is investigated in shallow water environments that include internal waves. Internal waves cause perturbations to the sound speed profile which results in ray chaos and causes uncertainty in the expected dynamics. In shallow water, the deep-water resonance structure exhibited in surfaces of section is largely missing due to the interaction of the rays with the ocean bottom. Instead, resonances caused by reflections create bands of chaos within large regions of regular dynamics. Lyapunov exponents are used to determine the extent of chaotic regions and to distinguish between regular and chaotic acoustic ray properties. Floquet theory is used to analyze range-periodic Hamiltonians that result from inclusion of the internal waves.

10:30

2aAO5. Matched-field replica vector stability in the winter ocean on the New Jersey (USA) shelf—RAGS03. Peter C. Mignerey and Marshall H. Orr (Acoust. Div. 7120, Naval Res. Lab., Washington, DC 20375-5350, mignerey@nrl.navy.mil)

A measure of the environmental influence on matched-field processing is the temporal autocorrelation of the coherent acoustic field passing through a vertical aperture. The matched-field autocorrelation times provide estimates of the time interval over which matched-field replica vectors will remain valid. In December 2003 the Naval Research Laboratory moored three vertical arrays at ranges of 10, 20 and 30 km distant from fixed 300- and 500-Hz CW acoustic sources. The purpose of the experiment was to measure the relationship of array gain to shelf break fluid processes (RAGS03). Data was recorded continuously for more than 20 days. Range and time dependent temporal autocorrelation times derived from this data will be presented and compared to similar measurements made at the South China Sea shelf break during the ONR AsiaEx01 acoustic propagation experiment. Preliminary RAGS03 results show that matched-field autocorrelation times are shortened during strong wind events and that sound speed fluctuations caused by semi-diurnal tidal processes forced quarter-diurnal fluctuations in the matched-field processor output. [Work supported by ONR.]

10:45

2aAO6. Estimating internal wave statistics from underwater acoustic transmission scintillation measurements on the New Jersey shelf with a 3-D stochastic model. Purnima Ratilal, Tianrun Chen, and Nicholas Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

The acoustic intensity expected after transmission through random inhomogeneities in an ocean waveguide is analytically expressed in terms of modal dispersion, attenuation, and energy redistribution in a 3-D multiple forward scattering formulation [Ratilal and Makris, J. Acoust. Soc. Am. **114**, 2428 (2003)]. This approach is used to model forward scattering through a random internal wavefield. Scattering from the density and compressibility inhomogeneities caused by the internal waves is approximated with the Rayleigh–Born series. The model is used to estimate the rms internal-wave height from low-to-mid-frequency underwater acoustic transmission scintillations measured during the Main Acoustic Clutter experiment of 2003 in the New Jersey Strataform area. Estimated internal wave height standard deviations matched those obtained from independent temperature and CTD measurements and suggest that the internal wave field was not temporally stationary.

11:00

2aAO7. Transmission loss and signal coherence statistics in the northeastern South China Sea shelf edge. Ching-Sang Chiu and Christopher Miller (Dept. Oceanogr., Naval Postgrad. School, Monterey, CA 93943, chiu@nps.edu)

Observations from the Northeastern South China Sea shelf edge in May 2001 showed spectacular changes in the sound-speed profile, transmission loss, and signal coherence at low frequency. These significant acoustical fluctuations were induced by the passage of large-amplitude, nonlinear internal waves that depressed the shallow isotherms to the ocean bottom along the transmission path. In this talk, the measured statistics of transmission loss, temporal coherence, and horizontal coherence of a 400-Hz signal transmitted upslope from a moored sound source to an L-shaped hydrophone array are presented and discussed. Specifically, the discussion is focused on both the inter- and intradaily variability of these observed statistics of the sound field and their dependence on the strength and timing of the nonlinear internal waves. [The research is sponsored by ONR.]

11:15

2aAO8. Shelf-break tidally induced environmental influences on acoustic propagation. Roger Oba, Steven Finette, Colin Shen, and Thomas Evans (Naval Res. Lab., Washington DC 20375)

Continuous wave propagation in the 100–500 Hz band in littoral regions depends upon both time-dependent oceanography and bathymetry. The environmental influences interact nonlinearly in the acoustical time variation, especially since the diurnal tide surface height changes creates time-dependent total water depth. A submesoscale hydrodynamic model developed by Shen and Evans is used with tidal forcing and a simple shelf-break bathymetry to produce surface height variation and internal wave activity due to internal tide in a stratified ocean environment. A three-dimensional parabolic equation acoustic model is used to acoustically probe this environment at various bearings relative to the shelf break and the resulting internal tidal dynamics. In particular, the acoustical results are examined for three-dimensional effects such as horizontal refraction. First, the influence of bathymetry alone is shown, and then compared to the full environment due to hydrodynamic action. The relative influences will then be compared by various measures such as modal decomposition, acoustic energy summed over depth, and signal gain degradation. [This research is sponsored by the ONR.]

Session 2aBB

Biomedical Ultrasound/Bioresponse to Vibration: Cavitation and Lithotripsy

R. Glynn Holt, Chair
Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215

Contributed Papers

8:30

2aBB1. Nonlinear oscillations of encapsulated gas bubbles in incompressible elastic media. E. A. Zabolotskaya, Yu. A. Ilinskii, G. D. Meegan, and M. F. Hamilton (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

An equation derived previously to describe nonlinear bubble oscillations in incompressible elastic media [Emelianov *et al.*, J. Acoust. Soc. Am. **115**, 581 (2004)] is modified to include an elastic shell with properties that are different from those of the host elastic medium. The new model equation is derived using Lagrangian mechanics. Nonlinearity is taken into account in the motion of the shell and surrounding medium, the compression of the gas, and the strain deformation in both elastic media. Two cases are considered for small but nonlinear bubble oscillations. The first applies to the case in which the equilibrium gas pressure is equal to the pressure at infinity, such that there is no equilibrium strain deformation in the medium. In this case, the Landau strain energy expansion can be used to obtain explicit expressions for the nonlinearity coefficients. For the second case, the equilibrium gas pressure in the bubble is different from the pressure at infinity, which results in strain deformation of both the shell and surrounding medium in equilibrium. Linear and nonlinear characteristics of the bubble oscillations are estimated for both cases. Limiting forms of the results are compared with those obtained by others. [Work supported by ARL:UT IR&D.]

8:45

2aBB2. Pressure and temperature fields from high-intensity focused ultrasound: Modeling the impact of bubbles and cavitation. Tianming Wu, Ronald. A Roy, and R. Glynn Holt (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, twu@bu.edu)

The propagation of high-intensity focused ultrasound (HIFU) in tissue-mimicking phantoms is modeled via a finite difference time-domain simulation. Above a threshold pressure, cavitation activity results and the HIFU focal zone becomes a bubbly medium. We assume an effective medium and account for the impact of bubbles by computing modified effective sound-speed and attenuation coefficients, where the latter includes cavitation-related dissipation mechanisms. Stability criteria establish the bubble equilibrium sizes, and comparison with experiments provides an estimate of the bubble number density. Nonlinear bubble responses are computed numerically and the resulting cycle-averaged void fraction is used to estimate the effective sound speed using a Woods approximation. Computed absorption cross sections related to viscous dissipation and the absorption of reradiated sound yield the effective attenuation coefficient. Using the updated sound-speed and attenuation coefficients, the pressure field is recomputed in an iterative process. Heat deposition is estimated using the averaged acoustic intensity as the heat source along with the evolving attenuation coefficient. The space-time-dependent temperature field and thermal dose is then calculated. Results indicate enhanced heating rates as well as a tadpole-shaped lesion that grows towards the HIFU transducer. [Work supported by the US Army.]

9:00

2aBB3. Observations of cavitation activity and lesion growth in optically clear tissue phantoms. Charles R. Thomas, Caleb H. Farny, Ronald A. Roy, and R. Glynn Holt (Dept. Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

Above a certain acoustic pressure threshold the heating rate of a tissue-mimicking phantom (as well as *in vivo* tissue) by high-intensity focused ultrasound (HIFU) is greatly enhanced. This enhanced heating regime has been shown to correlate well with an increase in cavitation activity; thus, it is believed that the enhanced heating is the result of bubbles formed at the focus of the HIFU source. In this talk we report the results of work carried out to observe the cavitation activity in the focus of a 1.1-MHz source, using optically clear acrylamide/BSA tissue phantoms. Three different methods were employed to make the measurements: simultaneous passive cavitation detection (PCD) and video imaging, simultaneous PCD and light emission measurements (using a photomultiplier tube), and video imaging with back light. Results complement previous work by other groups which showed that thermal lesion growth progresses towards the HIFU source; however in contrast to those studies, our results indicate that at some level cavitation is always present during the formation of thermal lesions in these particular tissue phantoms. [Work supported by the US Army and the Center for Subsurface Sensing and Imaging Systems via NSF ERC Award Number EEC-9986821.]

9:15

2aBB4. Measurement and correlation of acoustic cavitation with cellular bioeffects. Daniel M. Hallow (School of Chemical & Biomolecular Eng., Georgia Inst. of Technol., 311 Ferst Dr., Atlanta, GA 30332, daniel.hallow@chbe.gatech.edu), Todd E. McCutchen, Anuj D. Mahajan, Vladimir G. Zarnitsyn, and Mark R. Prausnitz (Georgia Inst. of Technol., Atlanta, GA 30332)

Noninvasive methods to measure and predict ultrasound effects on cells are needed to realize applications of ultrasound-mediated drug delivery to improve chemotherapy, gene therapy and targeted delivery. This study tested the hypothesis that (*i*) cellular bioeffects of ultrasound correlate with cavitation dose, (*ii*) broadband noise provides a measure of cavitation dose, and, thus, (*iii*) cellular bioeffects can be predicted by noninvasively measuring broadband noise. After exposing cell suspensions to ultrasound and measuring intracellular molecular uptake and loss of cell viability (bioeffects), a broad range of bioeffects were achieved by varying frequency, pressure, exposure time, cavitation nucleation site (Optison) concentration, and cell type. As a measure of cavitation activity, broadband noise measurements from acoustic spectra were collected during cell sonication and shown to be larger at elevated pressure and, after a high initial value, sharply decayed to a constant, background value at long exposure times. Combining these results, we found that broadband noise correlated well with molecular uptake and viability over the broad range of experimental conditions used (p-value <0.0001). This indicates that acoustic spectrum analysis provides a unifying parameter to correlate with bioeffects over a wide range of acoustic and experimental conditions. [Work supported by NIH, EKOS, DoEd GAANN Program.]

9:30

2aBB5. High-speed photography and acoustic emission synchronized observation of ultrasound induced acoustic cloud cavitation. Teiichiro Ikeda, Masataka Tosaki, Shin Yoshizawa, and Yoichiro Matsumoto (Mech. Eng., The Univ. of Tokyo, 7-3-1 Hongo Bunkyo-ku, Tokyo, 113-8656, Japan)

Though the violent collapse of the acoustic cavitation during ultrasound therapy may cause tissue traumas, it has a potential for therapeutic benefits if it is carefully controlled. The investigation of a two-frequency focused ultrasound method for the acoustic cloud cavitation control for lithotripsy is discussed. In the forcing cycle, cavitation is controlled by generation, growth, and shape stabilization during a high-frequency (1–4 MHz) phase, and a violent forced collapse that produces very high pressure during a low-frequency phase (400–550 kHz). Ultra high-speed photography for various conditions, gas concentration in the media and the acoustic field properties, are conducted. The photography (up to 200 MHz) framing rate is synchronized with the measurement of the acoustic emission from the cavitation bubbles by a concave PVDF hydrophone with a high directivity and a broadband response up to 10 MHz. The behavior of the cloud cavitation during the two-frequency cavitation control cycle is investigated with respect to the acoustic emission of the cavitation bubbles. The shape stabilization of the cloud cavitation and the high-pressure concentration on the solid surface by the forced collapse of the bubble cloud are discussed.

9:45

2aBB6. Controlling a high intensity focused ultrasound induced cavitation field via duty cycle. Caleb H. Farny, Charles R. Thomas, R. G. Holt, and Ronald A. Roy (Boston Univ., Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, cfarny@bu.edu)

Cavitation has been implicated in the lack of control over the shape of thermal lesions generated by high-intensity focused ultrasound (HIFU). A coincident effect the decline in the acoustic emissions from cavitation at the focus suggests that the HIFU energy is shielded from the focal region, possibly by prefocal bubble activity. Most clinical techniques employ continuous-wave (CW) ultrasound, which can exacerbate the problem depending on the acoustic intensities employed. This talk presents a series of experiments investigating techniques to control HIFU energy delivered to, and cavitation activity within, a tissue phantom. A passive cavitation detector (PCD) is employed as a sensor of cavitation activity. For 1.1-MHz CW ultrasound at focal pressures above 3 MPa, bubble shielding was inferred from a steady decline in the PCD signal over time. By lowering the duty cycle the PCD output remained constant over time. Finally, driving the HIFU source initially with a CW signal and then switching to a pulsed signal resulted in shielding, recovery, and a stable PCD signal, thus demonstrating our ability to control cavitation activity during HIFU exposure. [Work supported by the U.S. Army and the Center for Subsurface Sensing and Imaging Systems via NSF ERC Award No. EEC-9986821.]

10:00

2aBB7. Correlation of ultrasound-induced premature beats and cavitation *in vivo*. Claudio Rota, Carol H. Raeman, and Diane Dalecki (Dept. Biomed. Eng. and Rochester Ctr. for Biomed. Ultrasound, Univ. of Rochester, 319 Hopeman, Rochester, NY 14627, rota@bme.rochester.edu)

Exposure of the heart to pulsed ultrasound can produce arrhythmias such as premature ventricular contractions (PVCs). Recently, studies indicate that microbubble ultrasound contrast agents can further increase the sensitivity of the heart to ultrasound by lowering the pressure threshold for PVCs. It was hypothesized that the physical mechanisms for ultrasound-induced PVCs involve acoustic cavitation. To test this hypothesis, a passive cavitation detector (PCD) was used to directly measure cavitation *in vivo* and to correlate the detected output with the occurrence of a premature beat. Experiments were performed with adult anesthetized mice. Boluses of either a contrast agent (Optison) or saline were delivered via tail vein injections. Pulsed ultrasound exposures were performed at 200 kHz with pulse durations of 1 ms and peak negative pressures ranging between 0.1 and 0.25 MPa. A 5-MHz focused transducer was used as a passive listening device of acoustic signals. For the acoustic conditions above, premature beats were found in all mice injected with Optison and the effect correlated with PCD signal amplitude. Neither premature beats nor cavitation activity were observed among animals injected with saline and exposed to ultrasound. These results are consistent with cavitation as a mechanism for this bioeffect.

10:15–10:30 Break

10:30

2aBB8. A contrast source inversion method for imaging acoustic contrasts. Koen W. A. van Dongen and William M. D. Wright (Dept. of Elec. and Electron. Eng., Univ. College Cork, Cork, Ireland)

The propagation and scattering of acoustic wavefields is described by an integral equation of the second kind. In the forward problem, the kernel contains known Green's tensors and contrast functions and is applied on the unknown total wavefields. Since the contrast functions, given by changes in density and compressibility, and the incident wavefields are known, the total wavefields can be obtained by solving the integral equation iteratively via a conjugate gradient (CG) method. In the inverse problem, the complete incident wavefields and the total wavefields at a limited number of positions are known, while the contrast function is unknown. To solve the inverse problem, the contrast function can be obtained from the integral equation by using the same CG scheme. However, for each update of the contrast function the forward problem must be solved. To avoid this problem, contrast sources are introduced, defined by the product of contrast functions and corresponding total wavefields. Hence, the integral equation will be solved by reconstructing contrast sources, from which the true amplitude contrast function is obtained via a single step minimization procedure. Results obtained with standard imaging methods like back propagation will be compared with images obtained via the contrast source formulation.

10:45

2aBB9. A model for image formation in vibro-acoustography. Glauber T. Silva (Dept. de Tecnologia da Informacao, Universidade Federal de Alagoas, BR104N, km 14, Maceio, AL, Brasil, 57072-970) and Mostafa Fatemi (Mayo Clinic College of Medicine, Rochester, MN 55905)

Vibro-acoustography is a technique that images the vibro-acoustic response of an object to the harmonic ultrasound radiation force. The system point-spread function is associated to the radiation force exerted on a point-target by a dual-frequency ultrasound beam. So far the radiation force was calculated by assuming a dual-frequency beam composed by two plane waves. This reduces the radiation force to a one-dimensional quantity neglecting transverse components. Here, we model the radiation force as a three-dimensional vector. In this model, the scattering of a dual-frequency beam with any spatial distribution by a point-target is solved. The incident and scattered fields are used to calculate the radiation force on a sphere whose radius approaches to zero. The force is proportional to the gradient of the product of the incident wave amplitudes. Evaluation of the radiation force produced by a two-element co-focused transducer shows that the transverse component of the force is about -20 dB smaller than the corresponding axial component. Effects of the transverse force on image formation are still under investigation. In conclusion, the presented model describes vibro-acoustography systems with dual-frequency beams of any spatial distribution. [Work partially supported by Grant No. DCR2003.013-FAPEAL/CNPq.]

11:00

2aBB10. Microbubble contrast agents in vibro-acoutography. Prasika Manilal, Ahmed Al-Jumaily (Diagnostics and Control Res. Ctr., Auckland Univ. of Technol., Auckland, New Zealand), and Mostafa Fatemi (Mayo Clinic College of Medicine, Rochester, MN 55905)

The use of contrast agents is becoming a routine practice in diagnostic ultrasound imaging. Microbubble contrast agents are proving to be a safe and practical way of enhancing conventional ultrasound images. This paper discusses the concept of using microbubble contrast agents in the vibro-acoustography (VA) technique. A brief background of microbubbles and their clinical applications is given as well as a summary of the different attempts at modeling microbubble contrast agents in ultrasound fields. A general Keller–Herring equation is modified to mathematically model the interaction of encapsulated microbubbles in blood with VA ultrasound. The frequency response and scattered pressure of microbubbles with initial radius of 1 and 3 m are presented. This method is accurate for pressure amplitudes up to 500 kPa. Understanding this interaction is important for the correct interpretation of clinical investigations when using VA.

11:15

2aBB11. Vibro-acoustography for targeting kidney stones during lithotripsy. Neil R. Owen, Michael R. Bailey, Adam Maxwell, Brian MacConaghy (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, cimu@apl.washington.edu), Tatiana D. Khokhlova, and Lawrence A. Crum (Univ. of Washington, Seattle, WA 98105)

Vibro-acoustography can be used to measure material properties and detect calcifications within the body. Two transducers (diameter 10 cm, curvature 20 cm, frequency 1.1 MHz) are placed with overlapping foci in degassed water and driven at different frequencies to produce a dynamic radiation force in the range of 5–50 kHz. A LABVIEW program instructs the transducers to sweep through this frequency range at 500-Hz increments while a synthetic cylindrical kidney stone is held in the focus in one of three ways: with a rubber band, within an acrylamide gel, or within a finger cot. A low-frequency hydrophone, 10 cm from the focus and 90 deg from the direction of propagation, detects the radiated acoustic emission from the stone. The average amplitude of five signals is recorded to measure the frequency response of the stone. Unbroken stones exhibited higher amplitude response at frequencies near 10, 25, and 35 kHz. Stones are moved to simulate patient breathing and different in-focus and out-of-focus acoustic emissions indicate that vibro-acoustography may potentially target kidney stones during lithotripsy. Comminution is improved *in vitro* by gating SWs with targeting. [Work supported by NIH Grants DK43881 and DK55674, NSBRI Grant SMS00203, and CRDF.]

11:30

2aBB12. Finite difference time domain simulation of nonlinear ultrasonic pulse propagation. Keisuke Fukuhara and Nagayoshi Morita (Dept. of Elec., Electron., and Comput. Eng., Chiba Inst. of Tech., 2-17-1 Tsudanuma, Narashino, 275-0016, Japan, g0274502@cc.it-chiba.ac.jp)

The extracorporeal shock wave lithotripsy has come into wide use rapidly owing to its advantage of non-invasiveness. However, as for the shock wave propagation in the human body in relation to practical lithotripsy, reports of numerical simulation based on models conforming to a practical situation are very few. In this paper, excited pulse wave forms are measured by using a water tank model based on a practical lithotriptor and numerical simulation is made on the basis of these measured data. A new FDTD algorithm is proposed and used for this simulation, the problem being connected with extracorporeal shock wave lithotripsy. In this simulation method, conventional plane wave approximation is not used but original equations including convection terms are directly employed to derive new FDTD algorithms. This method is applied to an experimental setup and its numerical model that resemble an actual treatment situation to compare sound pressure distributions obtained numerically with those obtained experimentally. It is shown that the present method gives clearly better results than the earlier method does, in the viewpoint of numerical reappearance of strongly nonlinear wave forms.

11:45

2aBB13. Potential mechanism for the effect of shock wave rate in shock wave lithotripsy. Yuri A. Pishchalnikov, Richard J. VonDerHaar, James A. McAteer, Irina V. Pishchalnikova (Dept. of Anatomy and Cell Biol., Indiana Univ., School of Medicine, 635 Barnhill Dr., Indianapolis, IN 46202-5120, yura@anatomy.iupui.edu), Michael R. Bailey (Univ. of Washington, Seattle, WA 98105-6698), James C. Williams, Jr., and Andrew P. Evan (Indiana Univ., School of Medicine, Indianapolis, IN 46202-5120)

Artificial stones break significantly better when shock waves (SWs) are delivered at 0.5 Hz than at 2 Hz, and patients treated at slower rates have improved stone-free rates. One possible explanation may be cavitation bubbles that might persist between SWs at high rate and distort subsequent SWs sufficiently to reduce their effectiveness at stone comminution. High-speed photography gives evidence that bubble numbers are greater at higher rates. B-mode ultrasound echo in the free field typically disappears between pulses administered at 0.5 Hz but persists at 2 Hz. Fiberoptic hydrophone measurements at 2 Hz showed, in the free field of an electrohydraulic lithotripter, that SW negative tail was truncated, and proximal to a stone, that SW waveform varied and was distorted such that often the positive pressure amplitude was reduced. Changes in waveform proximal to stone declined as the stone disintegrated and fell away. Thus, data support the persistence of cavitation bubbles at high SW rate, and consequent distortion of waveform. Additionally, debris from a stone appears to accentuate this effect. These measurements may help not only define an effective SW rate and shape, but may further improve our understanding of the comminution process. [Work Supported by NIH-DK43881, DK55674, and ONRIFO-N00014-04-1-4010.]

Session 2aEA

Engineering Acoustics and Committee on Standards: MEMS Microphones: Fabrication, Calibration, and Application to High-Density Arrays

Allan J. Zuckerwar, Cochair
NASA Langley Research Center, Hampton, Virginia 23681

Qamar A. Shams, Cochair
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Chair's Introduction—8:30

Invited Papers

8:35

2aEA1. A commercialized MEMS microphone for high-volume consumer electronics. Peter V. Loeppert and Sung B. Lee (Knowles Acoust., 1151 Maplewood Dr., Itasca, IL 60143, pete.loeppert@knowles.com)

More than a billion microphones will be used this year in consumer electronic items such as cell phones, PDAs, MP3 players, and cameras. Because standard electret condenser microphones (ECMs) are temperature sensitive, these microphones are either hand placed or inserted with specialized equipment into the applications. MEMS microphones are tolerant of the high temperatures used in a lead-free solder process and hence are ideal for surface mounting with standard pick and place equipment. The challenge has been to develop a stable, low-cost MEMS microphone. This paper will present the design of the Knowles $SiSonic^{TM}$ microphone, which has been successfully commercialized over the past 2 years. Cost is a key driver in this market and total silicon area is a concern; however, the $SiSonic^{TM}$ microphone utilizes separate CMOS and MEMS dies because this has yielded a lower cost than proposed integrated solutions. The MEMS die size is driven by scaling issues and performance requirements. To achieve consistent performance, the $SiSonic^{TM}$ microphone uses a patented free-plate diaphragm. The microphone packaging has been developed to facilitate batch fabrication and a high-speed automated testing system has been developed to deliver a low cost component.

9:00

2aEA2. MEMS-based acoustic arrays: Promise and challenges. Mark Sheplak, Toshikazu Nishida, and Louis Cattafesta (Interdisciplinary Microsystems Group, Univ. of Florida, 231 MAE-A Bldg., P.O. Box 116250, Gainesville, FL 32611-6250, sheplak@ufl.edu)

A review of microelectromechanical system (MEMS)-based directional acoustic array technology is presented. The prospects for reducing cost, improving speed, and increasing mobility over conventional array technologies is critically reviewed. The advantages and limitations of existing devices are discussed. Finally, unresolved technical issues are summarized for future sensor development. A specific example of a system is presented that uses 16 hybrid-packaged silicon-micromachined piezoresistive microphones mounted to a printed-circuit board and a high-speed signal processing system to generate the array response over 2400 scan locations in under 20 s. The hybrid microphone packages show an average sensitivity of 0.8 mV/Pa with matched magnitude (0.6 dB) and phase (1 deg) responses between devices. The measured array response matches the theoretical response over the frequency range of 3 to 8 kHz with a localization error of 0.3 in. The array has a minimum detectable signal of 43.5 dB SPL for a 1-Hz bin at 6 kHz and a maximum pressure input of at least 160 dB SPL. These results represent a proof-of-concept demonstration of a high-speed, low-cost directional acoustic array system.

9:25

2aEA3. Design and fabrication of 128-channel MEMS-based acoustic array. Qamar A. Shams, William M. Humphreys, Bradley S. Sealey, Jimmy K. Adams, Toby Comeaux (NASA Langley Res. Ctr., M.S. 238, Hampton, VA 23681), John C. Ingham (Old Dominion Univ., Norfolk VA 23529), and Walter C. Babel (SAIC, Hampton, VA 23681)

Surface-mount microphones based on MEMS (micro-electromechanical system) technologies have recently become viable as component-level engineering solutions for acoustic measurements. In addition, advances in microelectronics, flexible circuitry, and array processing have motivated the design of a high-speed, low-cost acoustic array system for aeroacoustic measurements. A variety of microphones are available in the market today. Each type of microphone has its benefits and drawbacks. For example, standard condenser microphones have excellent sensitivity, stability, and high frequency response, but tend to be unwieldy, expensive, and require relatively high operational voltages. Electret microphones are small and fairly inexpensive but their performance deteriorates if exposed to moderately elevated temperatures. MEMS microphones combine the best features of the electret and condenser microphones while occupying a volume of less than 20 cubic millimeters. This paper details the custom-made 128-channel MEMS-based acoustic array for wind tunnel applications as well as the electrical, mechanical, and acoustic properties of the MEMS microphones utilized here.

9:50–10:00 Break

10:00

2aEA4. Development of MEMS microphone array technology for aeroacoustic testing. Qamar A. Shams, Sharon S. Graves, Scott M. Bartram, Bradley S. Sealey, and Toby Comeaux (NASA Langley Res. Ctr., Hampton, VA 23681)

A new approach to aeroacoustic microphone array design and implementation is described and demonstrated. Using commercially available, low-cost MEMS microphones exhibiting a suitable low-frequency response, a series of 128-channel arrays were constructed on flexible Kapton circuit boards which were bonded to rigid aluminum backplates. Cover panels with precision cutouts for the microphones were bonded on top of the Kapton circuit boards to create a smooth surface providing flush-mounting for all microphones. Connections for the microphones were created by extending strips of Kapton containing power and signal busses to the rear of the backplates. All channels were powered from a common 3 V power source, and all signals were conditioned using custom-manufactured filtering and line-driving hardware. The conditioned signals were digitized and processed in near real-time using both commercially available and customized data acquisition and analysis hardware. This new type of array construction addresses two challenges which currently limit the widespread use of large channel-count arrays for aeroacoustic applications, namely by providing a lower cost-per-channel solution and by providing a simpler method for mounting microphones in wind tunnels. The MEMS arrays have been extensively tested in anechoic and hard-walled facilities, and their performance has been found comparable to that of condenser microphone arrays.

10:25

2aEA5. Pressure calibration of MEM microphones. George S. K. Wong (Inst. for Natl. Measurement Standards, Natl. Res. Council, Ottawa, ON, Canada, K1A 0R6)

Micro-electro-mechanical microphones (MEMS) are micro devices approximately $4\times6\times1.5$ mm in size, with relatively low sensitivities (typically -40 dB *re* 1V/Pa) and operate to ultrasonic frequencies. However, the calibration of these microphones has been a challenge to most Metrology Institutes. One obvious route is to perform a free-field calibration. This presentation will discuss a preliminary coupler design for pressure comparison calibration of microphones to frequencies as high as 80 kHz.

Contributed Papers

10:50

2aEA6. Micromachined microphones with integrated optical nanoscale displacement sensors. Neal A. Hall, Wook Lee, Mohammad K. Jeelani, F. Levent Degertekin (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, gte802s@mail.gatech.edu), and Murat Okandan (Sandia Natl. Labs., Albuquerque, NM 87123)

Micromachined microphones with diffraction-based optical displacement detection are presented. A compliant membrane is made part of a phase-sensitive diffraction grating, and the deflection resulting from external acoustic pressure alters the intensities of the diffracted orders which are monitored with integrated photodiodes. The scheme provides the displacement sensitivity of a Michelson interferometer and can be integrated without beam splitters or critical alignment problems into volumes on the order of 1 mm^3. Preliminary characterization with ultrasonic sensors shows a displacement resolution of 1×10^{-4} Å/$Hz^{1/2}$ near 100 kHz with 60 μW of laser power incident on the photodetector. Current research is aimed at achieving similar displacement resolution in the audio frequency range while demonstrating the potential for high-fidelity miniature microphone arrays for hearing and measurement applications. The approach is implemented and characterized using microphone membranes with integrated diffraction grating bottom electrodes fabricated on silicon using Sandia National Laboratories' dedicated processing platform. Preliminary results on nonoptimized implementations show a flat frequency response to 15 kHz with internal noise levels below 40 dBA. [The authors would like to thank NIH and DARPA for supporting this research.]

11:05

2aEA7. Application MEMS microphones in photoacoustic instrumentation for bio-chemical detection. Michael Pedersen (CNRI, 1895 Preston White Dr., Ste. 100, Reston, VA 20191)

In this paper we discuss the design and application of MEMS-based microphones in photoacoustic (PAS) instrumentation, and it is demonstrated that by tailoring the properties of the microphone, tremendous improvements can be achieved in performance of particular importance to photoacoustic detection, such as sensitivity and noise level. Microphones typically used for PAS applications have bandwidths that exceed the requirements by an order of magnitude or more. New microphone designs, with bandwidths specifically targeted towards PAS applications, are shown to have open-circuit sensitivities in excess of 400 mV/Pa and noise levels around 0 dB SPL for a microphone diaphragm as small as 1×1 mm. This gain in performance may lead to a $10\times$–$100\times$ reduction of the detection limit in a state-of-the-art photoacoustic cell. Since the diaphragm in the MEMS microphone is very small, the mass of inertia is also greatly reduced, which leads to a reduction in vibration sensitivity of >10 dB over miniature hearing aid microphones and >20–30 dB over measurement microphones. The reduced vibration sensitivity is critical to push photoacoustic instruments from the laboratory into the field, where high performance rugged instruments are needed for bio-chemical detection.

11:20

2aEA8. Sources of excess noise in silicon piezoresistive microphones. Robert Dieme, Mark Sheplak, and Toshikazu Nishida (Interdisciplinary Microsystems Group, Univ. of Florida, Gainesville, FL 32611-6200)

The reduction of acoustic microphone size using microelectromechanical systems (MEMS) technology enables increased spatial and temporal resolution. Whether the small size can be effectively utilized depends on the signal-to-noise ratio and minimum detectable signal (MDS) that are a function of the structural geometry, material properties, and transduction method. The optimal MDS depends on both electronic and thermomechanical noise sources. Fundamental noise sources may be divided into frequency independent thermal noise and frequency dependent excess noise dominating at low frequencies. There have been some questions regarding the dominance of electrical or mechanical sources of the excess noise in piezoresistive microphones [A. Zuckerwar *et al.*, J. Acoust. Soc. Am. **113**, p. 3179–3187 (2003)]. Noise power spectra have been measured for various piezoresistive microphones. We present results on the bias

dependence of the excess noise that indicate that the primary source of excess noise is electrical. The relative contributions of mechanical and electrical noise sources will be discussed.

11:35

2aEA9. Free-field calibration of the pressure sensitivity of air-condenser, electret, MEMS, and piezoresistive microphones at frequencies up to 80 kHz. Gregory C. Herring, Allan J. Zuckerwar (NASA Langley Res. Ctr., Hampton, VA 23681), and Brian R. Elbing (Univ. of Michigan, Ann Arbor, MI 48109)

A free-field substitution method for calibrating measurement microphones at frequencies up to 80 kHz is demonstrated with both grazing and normal-incidence geometries. The method is suitable for newer technology microphones that cannot be calibrated with the electrostatic actuator (EA), currently the industry standard for high frequency calibrations. A substitution-based method, as opposed to a simultaneous method, avoids problems associated with the nonuniformity of the sound field and uses a $\frac{1}{4}$-inch pressure sensitive microphone as a known reference. A commercially available reference sound source (centrifugal fan) is used as a broadband acoustic source. Although the broadband excitation produces smaller instantaneous signal-to-noise ratios than tonal excitation, it minimizes reflection-related interferences that often plague free-field measurements. Calibrations were performed on $\frac{1}{4}$-in. air-condenser, electret, piezoresistive, and MEMS microphones in an anechoic chamber. Five repetitions of a single microphone, over three months, give a typical reproducibility of ± 0.2 dB. Because the air-condenser microphone can be calibrated with the EA, it was possible to estimate the accuracy of this free-field method by comparing the pressure sensitivity, as derived from the free-field measurement, with that of the EA calibration. A typical comparison gives a rms difference of ± 0.4 dB, over the range 2–80 kHz.

TUESDAY MORNING, 16 NOVEMBER 2004 SAN DIEGO ROOM, 10:00 A.M. TO 12:00 NOON

Session 2aED

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

Paul A. Wheeler, Chair
Utah State University, 1595 N 1600 E, Logan, Utah 84322

Chair's Introduction—10:00

Acoustics demonstrations will be distributed around the room. All demonstrations will be available for high school students' hands-on experimentation. Participation by other conference attenders is welcome as long as their activity does not interfere with student learning.

TUESDAY MORNING, 16 NOVEMBER 2004 ROYAL PALM SALONS 3 & 4, 8:30 A.M. TO 12:00 NOON

Session 2aMU

Musical Acoustics: Pipe Organs

Thomas D. Rossing, Chair
Physics Department, Northern Illinois Univeristy, DeKalb, Illinois 60115

Invited Papers

8:30

2aMU1. On the acoustical design of the ears of flue organ pipes. Yumiko Sakamoto, Shigeru Yoshikawa (Dept. of Acoust. Design, Grad. School of Kyushu Univ., Fukuoka, Japan, yumiko@rms.kyushu-id.ac.jp), and Judit M. Angster (Fraunhofer-Inst. Bauphysik, Stuttgart, Germany)

Application of the ears to the flue organ pipe is one of the important voicing techniques. Ears are the projections on both sides of the pipe mouth. Organ builders say the ears make not only the sound lower and darker, but also the buildup of tone smoother and quicker. The aim of this research is to confirm their recognitions and make the causes clear. For that purpose, we made acoustical and flow measurements (measurement of the velocity profiles at the mouth) with model pipes. As a result, we could confirm the recognition of the organ builders. In addition, our experiments indicate a slight increase in the blowing pressure in the foot and an increase in the inharmonicity of the pipe eigenmodes. The ear reduces the maximum jet velocity but keeps the characteristic profiles. In some cases, the profiles move as a whole more inside of the pipe. Recent acoustic measurements (eigenmodes of the pipe resonator and of the mouth tone, attack transient and stationary spectrum) on real organ pipes with ears of different heights will be also reported at the meeting.

9:00

2aMU2. CFD simulation of deflection of a jet emerging from organ pipe flue. Seiji Adachi (ATR Human Info. Sci. Labs, Keihanna Sci. City, Kyoto 619-0288 Japan, sadachi@atr.jp)

The phenomenon of an air jet deflected by sound is the crucial element in the sounding principle of air-jet driven instruments. Currently, the jet deflection model proposed by Fletcher is most widely accepted. This model successfully predicts basic properties of the instrument such as overblowing behavior with a change in blowing pressure. However, a gap still exists between the actual behavior and that predicted by the model. This gap is probably due to the model's conceptual approximations, such as an inviscid fluid and a linear response of the deflection amplitude to the magnitude of the sound field. This research aims to develop a jet deflection model directly from the Navier–Stokes equations, which govern all flow-related phenomena. Therefore, computational fluid dynamical (CFD) simulation of a jet emerging from a flue slit and deflected by the external sound field was carried out. The results were analyzed to obtain the velocity profile of the jet, the phase delay, and the magnitude of the jet deflection amplitude. The overblowing behavior estimated from the analysis is discussed by comparing it with that actually observed.

9:30

2aMU3. The influence of pipe scaling parameters on the sound of flue organ pipes. Judit M. Angster, Tilo Wik, Christian Taesch, Yumiko Sakamoto, and Andras Miklos (Fraunhofer Inst. Bauphysik, Nobelstr. 12., D-70569 Stuttgart, Germany, rata@ibp.fhg.de)

When basic phenomena of the physics of flue organ pipes is studied, experiments on models are acceptable. But these models often differ considerably from real organ pipes. For this reason the fine details of pipe sounds should be investigated on real pipes. The sound quality of an organ pipe is mainly influenced by the attack transients. This onset is first dominated by the edge tone, while later the pipe resonator will play a more important role. To understand the physics of a flue organ pipe it is necessary to measure the acoustic properties of the pipe resonator to analyze the edge tone, the attack transient, and the stationary sound of the pipe. Several special pipes with the same pitch have been investigated: pipes with different diameters; a pipe of which the cut up and a pipe of which the length is adjustable. By the evaluation all physical effects contributing to the production of sound were taken into account. The results together with the results of subjective listening tests will be used for developing a scaling method for dimensioning labial organ pipes and a software for designing organ pipe dimensions of the most important ranks. [Work supported by the European Commission.]

10:00

2aMU4. Attack transient for combinations of flue pipes. A. W. Nolle (Dept. of Phys., Univ. of Texas, Austin, TX 78712, nolle@mail.utexas.edu.)

The tonal attack due to two or more flue pipes sounded simultaneously is compared to that for a single pipe. Attack duration is measured by the times required to progress from 1% to 50% and 90% of final amplitude. Duration can be reduced by the presence of the pipe(s) of higher pitch, because the number of periods in the sound buildup for pipes of a given design does not change greatly with the pitch. Experimental results are from sound records taken in close proximity to the pipes of a Casavant organ having wind chests of Pittman type. The positive division provides stops intended for use with a closed-end 8-foot rank. The great division provides ranks intended for use with an 8-foot (open) principal. A special situation is found in principal pipes near 200 Hz. Sounding one of these with another stop at unison produces an initial sound suggesting a celeste combination. This is found to be the result of a prolonged burst of second harmonic, locked to the growing fundamental, not of frequency difference. This possibility appears in an analytical mode-locking model by Fletcher [N. H. Fletcher, J. Acoust. Soc. Am. **64**, 1566–1569 (1978)].

10:30

2aMU5. Experiments on redirection of organ pipe sound by coupling. Mendel Kleiner (Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY 12180, kleiner@rpi.edu), Matthias Scholz (Chalmers Univ. of Technol., Gothenburg, Sweden), and Munetaka Yokota (Gothenburg Univ., Gothenburg, Sweden)

It is well known among organ builders that the sound of an organ pipe may be influenced by the addition of closely tuned pipes in the vicinity of the pipe. Rather than being a result of absorption of sound it is hypothesized that the effect is due to the sound being redirected due to coupling between pipes, similar to the coupling between elements in a Yagi antenna array. Measurement results will be shown and discussed. [Work supported by RPI and Chalmers University of Technology.]

11:00

2aMU6. Acoustics of organ reed pipes. Eric Cox and Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115)

In most lingual organ pipes, the reed vibrates against a fixed shallot, and modulates the flow of air passing the shallot into the resonator. We have measured vibrations of plucked and blown reeds of lingual pipes with and without the resonators. We discuss the effects of the acoustic field on the reed vibration and especially the interaction between the reed and the tuned resonator.

11:30

2aMU7. A comparison of different expression devices in pipe organs. Jonas Braasch (CIRMMT, Faculty of Music, McGill Univ., Montreal, QC H3A IE3, Canada, jb@music.mgill.ca) and Thomas D. Rossing (Northern Illinois Univ., DeKalb, IL)

After the introduction of the orchestra crescendo at the end of the 18th century by the ''Mannheim school,'' the ability to play the organ expressively like an orchestra was one of the organ builders' greatest concerns. Soon, both the wind swell and door swell came into fashion, and later another swell system, the crescendo wheel, was introduced. While both door swell and crescendo wheel can be successfully applied to all types of organ stops, the wind swell only works well with free reeds for tuning reasons. In this investigation, all three swell systems were measured on various instruments and compared to each other. In addition, two free-reed pipes were measured at the Northern Illinois University using a laser vibrometer. The crescendo wheel was found to be most effective, and for frequencies around 2 kHz the increase in sound-pressure level could be up to 50 dB between the softest and the loudest adjustment. The maximum dynamic range for the wind and the door swells is approximately 10 dB in the same frequency range. While the dynamic range is lower for the wind swell and the door swell, their advantage is the continuous variability.

TUESDAY MORNING, 16 NOVEMBER 2004 ROYAL PALM SALON 1, 9:00 TO 11:35 A.M.

Session 2aNS

Noise: Propulsion/Airframe Aeroacoustics I

Joe W. Posey, Chair
NASA Langley Research Center, Hampton, Virginia 23681

Chair's Introduction—9:00

Invited Papers

9:05

2aNS1. NASA's propulsion airframe aeroacoustics research. Russell H. Thomas (NASA, MS 166, NASA Langley Res. Ctr., Hampton, VA 23681-2199)

The integration of propulsion and airframe is a fundamental consideration in the design of an aircraft system. Many considerations influence the integration, such as structural, aerodynamic, and maintenance factors. In the future, a focus on the aerodynamic and acoustic interaction effects of installation, propulsion airframe aeroacoustics will become more important as noise reduction targets become more difficult to achieve. In addition to continued fundamental component reduction efforts, a system level approach that includes propulsion airframe aeroacoustics will be required in order to achieve the 20-dB noise reductions envisioned by the aggressive NASA goals. This emphasis on the aeroacoustics of propulsion airframe integration is a new part of NASA's ongoing acoustics research. The presentation will review current efforts and highlight technical challenges and approaches.

9:35

2aNS2. Diagnostics and reduction of propulsion airframe aeroacoustic interactions. Vinod G. Mengle, Robert W. Stoker, and Ronen Elkoby (The Boeing Co., P.O. Box 3707, MC 67-ML, Seattle, WA 98124)

In the past, significant reduction in aircraft noise has been achieved by studying engine noise and airframe noise in isolation as two separate components. However, when an engine is installed on an aircraft the flow and acoustic interactions between them produce a total noise signature which is often different than the sum of the component noise signatures. While take-off conditions are typically dominated by engine noise, approach conditions are characterized by both engine and airframe noise. Nevertheless, propulsion airframe aeroacoustic (PAA) interactions are present under all conditions and gain in importance when engine or airframe noise is further reduced. This paper focuses on the diagnostics and reduction of the flow-acoustic interaction effects between an engine exhaust and the airframe, especially the high-lift system on wings. Flow and acoustic results are presented for an isolated scale-model nozzle in a free jet, and also when it is installed under a wing at take-off and approach conditions. The difference between these two noise signatures, in the far field and at the source, captures the PAA effect. In particular, the PAA effect due to changes in nozzles, pylons, and flaperons is presented and the overall noise reducing mechanisms are postulated.

10:05

2aNS3. Propulsion airframe aeroacoustics practices at Honeywell. Donald S. Weir (Honeywell, P.O. Box 52181, MS 503-333, Phoenix, AZ 85072-2181)

Honeywell has been developing and applying acoustic models of propulsion airframe aeroacoustic phenomena for over 20 years. The initial application of a wing-shielding model was developed for the NASA General Aviation Synthesis Program in 1982. Since that time, more sophisticated models of wing shielding and reflection have been developed with internal and NASA funding. Recent work has involved models of wing shielding for aft mounted engines and wing reflection for wing mounted engines. These methods are described in the presentation. Comparisons with the Raynoise Code by LMS and measured aircraft fly over noise data are made to show the effectiveness of the model. The attenuation of the inlet noise by the wing of an aft mounted engine and the amplification of the noise by wing mounted engines are evaluated.

10:25–10:40 Break

10:40

2aNS4. Distributed exhaust for jet noise reduction. David B. Schein (Northrop Grumman Corp., 9S26/W6, One Hornet Way, El Segundo, CA 90245-2804, david.schein@ngc.com)

Acoustic detection of low flying, jet-powered military air vehicles and takeoff noise levels from commercial jets are often driven by jet mixing noise radiated from the exhaust. Cueing provided by the noise signature results in increased opportunity for a ground-based visual observer against such a target. Significant reductions in detectability and takeoff noise levels can be achieved through reductions in jet noise. Research has been conducted over the past several years to develop innovative, quiet, distributed exhaust nozzle (DEN) concepts, which attempt to achieve revolutionary reductions in jet mixing noise while minimizing propulsion penalties. The DEN's benefit relies on discharging the exhaust flow through many miniature nozzles rather than one or two large nozzles. Noise suppression from the DEN concept results from a favorable shift in frequency content compared to conventional jets. Significant increases in atmospheric attenuation and decreases in the ear's sensitivity at these higher frequencies result in much reduced detection ranges and perceived noise levels. This technology is applicable to both subsonic and supersonic aircraft, and particularly to low flying fixed-wing unmanned air vehicles, special operations forces transports, and future commercial transports with especially stringent noise reduction requirements. [Work supported by NASA LaRC.]

11:00

2aNS5. Lateral attenuation and airport noise modeling. Kevin P. Shepherd (NASA Langley Res. Ctr., M.S. 463, Hampton, VA 23681, k.p.shepherd@nasa.gov)

Prediction of airport noise is generally accomplished using semiempirical methods such as those contained within the F.A.A.'s Integrated Noise Model or the U.S.A.F.'s Noisemap. One component of these models addresses the prediction of noise for observers located at lateral, or sideline, positions for which the elevation angle to the source is small. Under these conditions the effect of the presence of the ground surface on sound propagation is important, as is the source directivity. Recent efforts in the United States and Europe have been aimed at separating these two phenomena with the result that acoustical directivity characteristics have been quantified for a range of commercial aircraft. Differences between aircraft with wing- and tail-mounted engines are significant. These recent studies will be reviewed and attempts made to identify the physical mechanisms responsible for the observed source directivity characteristics.

Contributed Paper

11:20

2aNS6. Further developments in aircraft flyover noise synthesis and propagation. Brenda M. Sullivan and Stephen A. Rizzi (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA 23681)

Subjective assessments of the noise from aircraft flight operations require time histories of acoustic pressure at listener positions. Synthesized sound has an advantage over recordings by allowing the examination of proposed aircraft, flight procedures, and other conditions or configurations for which recordings are unavailable. A two-stage process for synthesizing flyover noise at listener positions on the ground was previously developed, enabling the creation of an immersive test environment. The first stage entails synthesizing time histories at the flying source. Rizzi and Sullivan [J. Acoust. Soc. Am. **113**, 2245 (2003); **114**, 23 (2003)] presented an approach for synthesizing sound from broadband sources (e.g., jet noise) based on predicted $\frac{1}{3}$-octave band source spectra, with the inclusion of temporal fluctuations based on empirical data. Reported here are further developments in the synthesis of tone-dominated source (e.g., fan noise). The second stage entails propagation of the synthesized sound from the flying source to the listener. Improvements in the atmospheric attenuation part of the second stage are also presented.

Session 2aPA

Physical Acoustics: Topics in Atmospheric Sound Propagation

Roger M. Waxler, Chair
National Center for Physical Acoustics, University of Mississippi, 1 Coliseum Drive, University, Mississippi 38677

Contributed Papers

7:45

2aPA1. The acoustical equivalence principle for ray tracing. Edward R. Floyd (10 Jamaica Village Rd., Coronado, CA 92118-3208)

While the acoustical wave functions for various acoustical problems can be mapped into each other by a coordinate transformation (the acoustical equivalence principle), the corresponding ray tracings cannot be so mapped. Applying the acoustical equivalence principle [A. E. Faraggi and M. Matone, Int. J. Mod. Phys. A **15**, 1869–2017 (2000) for the analogous quantum equivalence principle] to the underlying acoustical Hamilton–Jacobi equation for ray tracing renders a modified acoustical Hamilton–Jacobi equation which does obey the quantum equivalence principle. Solutions of the modified acoustical Hamilton-Jacobi equation, which are either acoustical Hamilton's principal function or the acoustical reduced action (acoustical Hamilton's characteristic function), are the generators of motion for rigorous ray tracing.

8:00

2aPA2. Application of the Green's function parabolic equation (GFPE) method to realistic outdoor sound propagation scenarios. Jennifer L. Cooper (ENSCO, Inc., 5400 Port Royal Rd., Springfield, VA 22151, cooper.jennifer@ensco.com)

The Green's function parabolic equation (GFPE) is a powerful method for outdoor sound propagation prediction. Given appropriate inputs, GFPE predictions can include the effects of terrain, ground cover, sound speed profile, and turbulence on atmospheric propagation. In many situations, however, access to such input information can be limited. The input data resolution and accuracy can have a significant impact on model output accuracy. In turn, variability of terrain and sound speed profile and statistical aspects of the turbulence impact the model parameter requirements. For example, in the presence of turbulence, lower limits on computational grid height are determined by turbulence correlation scales as well as acoustic wavelength. Input data resolution requirements and some techniques that may be employed to account for insufficient input data will be discussed. Model results for realistic situations will be presented.

8:15

2aPA3. A parabolic equation solution for advected acousto-gravity waves. Jeremy Bruch, Michael D. Collins, Dalcio K. Dacol, Joseph F. Lingevich (Naval Res. Lab., Washington, DC 20375), and William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180)

It is relatively difficult to account for advection in parabolic equation solutions. An effective solution was recently derived for the acoustic case [J. Acoust. Soc. Am. **111**, 729–734 (2002)]. This work is currently being extended to include buoyancy effects. In the initial study, emphasis is placed on wave numbers near the Lamb wave, where buoyancy and compressibility both have a significant effect. An approximate parabolic solution is derived and tested for range-indendent cases using a spectral solution. The limiting cases of small ambient flow and/or no gravity are compared to the extended result for consistency. [Work sponsored by the ONR.]

8:30

2aPA4. The theory of the generation of atmospheric microbaroms. Roger Waxler and Kenneth E. Gilbert (NCPA, 1 Coliseum Dr., P.O. Box 1848, University, MS 38677, rwax@olemiss.edu)

It is well known that the standing wave components of large ocean wave systems, such as those produced by storms, radiate infrasound (microbaroms) into both the ocean and the atmosphere. The radiated sound is found in a narrow band (from 0.2 to 0.3 Hz wide) centered near 0.2 Hz. It is also well known that the radiation mechanism is nonlinear since ocean-wave wavelengths are too short to directly radiate. A systematic theoretical study of the radiation mechanism has been undertaken. The model considered is the two-fluid system consisting of atmosphere (a very rare fluid) over ocean (a very dense fluid). The interface between the fluids is allowed to undulate. The equations of fluid mechanics are then solved to second order in Mach number. It is found that the physical mechanism for (as well as the mathematical form of) the radiation into the atmosphere is entirely different from that of the radiation into the ocean. In particular, the compressibility of the atmosphere plays a crucial role.

8:45

2aPA5. Fluctuating pressure distributions around spherical foam windscreens. Jeremy Webster, Richard Raspet, and Kevin Dillion (Natl. Ctr. for Physical Acoust, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, jwebster@olemiss.edu)

The low-frequency wind noise reduction produced by spherical foam windscreens outdoors can be estimated by measuring the steady-state flow pressure distribution around the windscreen and then area averaging to produce a pressure response at the center of the sphere [Zheng (2003)]. We have constructed a foam windscreen with a distribution of four probe microphones near the surface to investigate the validity of this hypothesis at low frequencies and to measure correlations between the pressure fluctuations at different positions on the surface and at the center. The aim of this research is to develop methods of wind noise reduction which combine the effectiveness of passive foam windscreens with active multiple microphone processing techniques. In this paper we describe the construction of the test windscreen and initial measurements made outdoors.

9:00

2aPA6. A simple normal-mode model for nighttime traffic noise. Kenneth E. Gilbert, Roger Waxler, and Carrick L. Talmadge (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, Coliseum Dr., University, MS 38677, kgilbert@olemiss.edu)

At night, traffic noise propagates to long distance via normal modes trapped near the ground in the sound duct created by surface cooling. Consequently, at distances of 500 m or more, traffic noise at any given location is the sum of contributions from many individual vehicles. To predict the band-averaged and time-averaged levels in such situations, one can use an incoherent sum of normal modes. Using an incoherent modal sum, we derive a simple analytic expression for the noise due to traffic on an infinitely long highway. For long highways (10–20 km), it is shown that the controlling factor in the noise propagation is not distance alone,

but, instead, the product of the mode attenuation coefficients and the distance from the highway. The predictions for an infinitely long highway are compared with those for a highway of finite length and error bounds are given.

9:15

2aPA7. The near ground structure of the nocturnal sound field: The existence of a quiet height in the frequency domain. Roger Waxler, Carrick L. Talmadge, Kenneth E. Gilbert, and Shantharam Dravida (NCPA, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu)

The sound field generated by a point source in the downward refracting atmosphere typical of the nocturnal boundary layer is considered. Theoretical arguments are given that, at a fixed frequency, at ranges greater than a few hundred meters from the source the magnitude of the sound field necessarily has a deep minimum a few meters above the ground. The precise height of this minimum depends on the sound speed profile as well as the frequency, decreasing with increasing frequency. It is shown that a recently developed modal description of the nocturnal sound field yields a simple explanation for the existence of this quiet height and its emergence with increasing range. Experimental results verifying the existence and frequency dependence of the quiet height are presented.

9:30

2aPA8. Forest effects on acoustic pulse propagation. Donald G. Albert, Frank E. Perron, Jr., and Stephen N. Decato (USA ERDC Cold Regions Reasearch and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755)

A series of short-range outdoor measurements was conducted to investigate forest effects on acoustic pulse propagation. The measurements investigated seven different forest stands with a variety of different tree species including deciduous, evergreen, and mixed. A 0.45-caliber blank pistol shot was used as the source of the acoustic pulses, and the signatures were recorded 30 to 60 m away using a digital seismograph. An identical measurement was conducted in an open field for comparison. The recorded waveforms generally show the elongation characteristic of pulse propagation over a highly porous ground surface, with high-frequency reverberation arrivals superimposed on the basic waveform shape. These measurements can provide parameters useful for theoretical predictions of acoustic propagation within forests, and also illustrate some of the natural variability encountered in these environments. [Work funded by U.S. Army.]

9:45

2aPA9. Contributions of a thin attenuating layer in forest sound propagation. Michelle Swearingen and Michael White (Engineer Res. and Development Ctr., Construction Eng. Res. Lab., 2902 Farber Dr., Champaign, IL 61822)

The GFPE has been modified to incorporate a forest by replacing the local wavenumber with the bulk wavenumber of Twersky's multiple scattering theory. For modeling the effect of the forest canopy, this corresponds to an attenuating layer at some height above the ground surface. Because of propagation paths above and beneath the canopy, the attenuation per unit distance is generally less than that of the canopy. In this presentation, the attenuation contribution of this layer will be examined by comparing propagation predictions with and without the attenuation layer, with refracting and non-refracting sound speed profiles, and with rigid and realistic ground surfaces. It is interesting that some propagating modes in the forest are suppressed by the attenuation, removing a source of interference and leading to an apparent decrease in attenuation. An analysis of the sensitivity of the GFPE to attenuation layers of different strengths will be performed for receivers near the ground, taking both frequency and distance into account.

10:00

2aPA10. Nonlinearity in outdoor propagation of periodic signals: Measurement results. J. Micah Downing, Michael M. James, Christopher M. Hobbs (Wyle Labs., 2001 Jefferson Davis Hwy. Ste. 701, Arlington, VA 22202), Kent L. Gee, Victor W. Sparrow (The Pennslyvania State Univ., University Park, PA 16802), and Sally A. McInerny (The Univ. of Alabama, Tuscaloosa, AL 35487)

As a continuation of a joint program among Wyle Laboratories, Penn State, and The University of Alabama to develop better noise models for military aircraft, field measurements of high-amplitude periodic and broadband waveforms were conducted at Blossom Point Range, MD. The source of the acoustic signals was the Army Research Laboratory's Mobile Acoustic Source (MOAS) pneumatic speaker, which was designed to produce high-amplitude, low-frequency sounds. The results of the measurements demonstrate nonlinear effects in the higher harmonics when compared to linear models and provide a relative comparison of nonlinear effects with other propagation effects such as atmospheric and ground. The results also provide a simple data set for evaluation of proposed propagations models without introducing some of the complexities produced by real jet noise sources. One such comparison is provided in a companion paper (Kent L. Gee *et al.* [J. Acoust. Soc. Am 116, 2517 (2004)]). [Work is supported by the Strategic Environmental Research and Develop Program.]

10:15–10:30 Break

10:30

2aPA11. Nonlinearity in outdoor propagation of periodic signals: Prediction model development. Kent L. Gee, Victor W. Sparrow (Grad. Program in Acoust., The Penn State Univ., 217 Appl. Sci. Bldg., University Park, PA 16802), Michael M. James, and J. Micah Downing (Wyle Labs., Arlington, VA 22202)

Propagation measurements made of high-amplitude periodic signals generated by the Army Research Laboratory's Mobile Acoustic Source (MOAS) demonstrate greater energy at high harmonics relative to linear predictions, suggesting the possible influence of nonlinear effects (see Downing *et al.* [J. Acoust. Soc. Am. 116, 2517 (2004)]). An arbitrary-waveform version of an Anderson-type algorithm has been developed in order to compare numerical predictions with the measured MOAS spectra. In general, results demonstrate good agreement between predicted and measured spectra out to 375 m. However, comparisons at greater distances (approximately 1 km) and also for measurements made later in the afternoon exhibit less agreement. For these cases, the Anderson calculations generally overpredict the amount of high-frequency energy present in the measurements. Probable causes for these discrepancies include the effects of propagation at grazing incidence over a finite-impedance ground as well as increased atmospheric turbulence. [Work supported by the Strategic Environmental Research and Development Program.]

10:45

2aPA12. Comparison of computer codes for propagation of high-frequency energy from blast waves. Alexandra Loubeau and Victor W. Sparrow (Grad. Program in Acoust., The Penn State Univ., 202 Appl. Sci. Bldg., University Park, PA 16802, aloubeau@psu.edu)

Current environmental regulation requires that the Department of Defense assess the potential impact of noise from military training on endangered wildlife. One concern is the effect of noise from Army weapons on the hearing of bats. High frequencies are generated from nonlinear propagation of finite-amplitude shock waves created by explosions. These frequencies may be harmful to bats because their auditory systems are sensitive to high-frequency information that they use for flight navigation, communication, and hunting. Determining the spatial extent of high-frequency propagation requires an understanding of the short rise times associated with blast waves. A comparison of computer codes is performed for the most sensitive frequency range of bat hearing, 10–100 kHz. Earlier published results by the authors [A. Loubeau and V. W. Spar-

2a TUE. AM

row, Proc. NOISE-CON 2004, 193–201 (2004)] are compared to numerical solutions of the generalized Burgers equation with molecular relaxation included. The approaches considered are a time-domain method by Cleveland and a hybrid time–frequency-domain algorithm by Anderson. The results are also compared to recent experimental explosion data obtained by the US Army. [Work supported by US Army Engineer Research and Development Center CERL.]

11:00

2aPA13. Height-of-burst influence on spectra emitted by small explosions. Michael J. White (U.S. Army ERDC/CERL, P.O. Box 9005, Champaign, IL 61826)

Explosions in air nearby to the ground produce high-pressure waves that obey nonlinear superposition. The ground-reflected wave, when superimposed on the positive pressure phase of the direct wave, advances to merge in a mach stem. ANSI Standard S2.20 offers a correction factor applied to the real charge weight, to account for the increase in peak overpressure. Over a hard surface, the factor ranges from 2 near the ground to a maximum value of 5.6 at a scaled height (ratio of height to cube root of charge mass) equal to 3.9 m/[(kg)$^{(\frac{1}{3})}$]. We performed a set of experiments to measure the height-of-burst effect from explosions of composition C-4 over grass-covered ground at horizontal distances between 5 and 174 m. We offer an empirical analysis of the resulting spectra, and extend the height-of-burst analysis to far-field spectral sound power.

11:15

2aPA14. Propagation of spark generated N waves through turbulent media. Philippe Blanc-Benon and Sébastien Ollivier (Ctr. Acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, 69134 Ecully Cedex, France)

It is generally accepted that turbulence plays a role in the propagation and the distortion of sonic booms generated by supersonic flights. This paper describes laboratory experiments in which acoustic shocks from electric sparks have been used to model the propagation of sonic booms [Ollivier *et al.*, 10th AIAA/CEAS Conference, AIAA 2004-2921]. In order to study separately the influence of random fluctuations of temperature and velocity, the N-waves are propagated through a turbulent plane jet or over a heated grid. The recording of 1000 snapshots allows statistical analysis of the variation of the parameters used to describe N-waves, including the maximum and the minimum peak pressures, the rise time, the arrival times and the duration of the wave. In particular, the data show that the increase of the average rise time and the decrease of the average peak pressure with the increase of the rms value of the fluctuations or the increase of the propagation distance could be linked to the probability of occurence of random caustics as observed in numerical simulations [Blanc-Benon *et al.*, J. Acoust. Soc. Am. **111**, 487–498 (2002)]. [This work is partly supported by the European Community SOBER project, Contract No. G4RD-CT-2000-00398 and by the French Ministère de la Recherche et des Nouvelles Technologies.]

11:30

2aPA15. Exploitability of fluctuations to enhance signal processor performance. Kenneth E. Gilbert and Ronald A. Wagstaff (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, rwagstaf@olemiss.edu)

Acoustic literature abounds with experimental and theoretical studies of fluctuations. Fluctuations are often the cause of a signal processors failure to satisfy its objectives, e.g., not achieving enough signal-to-noise ratio (SNR) gain to detect a signal of interest. Fortunately, the spectral time-histories of fluctuating parameters, such as amplitude, are encoded with signal present or signal absent information. Overcoming and exploiting the fluctuations to achieve various forms of gain requires understanding the changes in the fluctuations that are caused by the signal being present and then choosing the appropriate fluctuation-based signal processing algorithm to exploit those differences. First, detrimental influences of fluctuations on the general performance of signal processors will be discussed. Next, differences in the fluctuations for signal present and for signal absent will be addressed, with their corresponding implications for degrading signal processor performance. It will be illustrated that when knowledge of the differences in the fluctuations is applied to signal processing algorithms, higher levels of signal processor performance can be achieved. Improvements include increases in SNR, spatial and spectral resolution, temporal coherence, and unalerted auto-detection of signals of interest. [Work was sponsored by the U.S. Army under Contract No. DASG60-00-C-0061.]

11:45

2aPA16. Three generic signal processing techniques in fluctuation-based processing for increasing gain. Ronald A. Wagstaff and Kenneth E. Gilbert (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677, rwagstaf@olemiss.edu)

Fluctation-based processors (FBPs) utilize an approach to signal processing that recognizes there is signal present or signal absent information imbedded in the fluctuations of the spectral time-histories. FBP algorithms are designed to focus on the changes in the fluctuations that are caused by a signal being present. There are three generic classes of FBP. Each class is defined by the fluctuating parameter and the method that the signal processing algorithm uses to exploit it. Typical gains include increases in signal-to-noise ratio, spectral and spatial resolution, and auto-identification of signals. The signal processing versatility of two of the classes provide additional gains by increasing temporal coherence. This versatility also led to development of processors that self-adapt to become more robust and provide higher gains. This includes adaptive processors that utilize the fluctuations to self-tune their governing equations. Thus they are able to preserve desired signals, while eliminating noise, transients, and undesirable signals, e.g., clutter. The operational manner of these three classes of FBP will be described. Key equations and algorithms will be explained. Results from single sensors and arrays for undersea and atmosphere acoustic data will be presented. [Work sponsored by the U.S. Army under Contract No. DASG60-00-C-0061.]

Session 2aSA

Structural Acoustics and Vibration: Analysis Methods: Statistical, Numerical and Analytical

Courtney B. Burroughs, Chair
Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804-0030

Contributed Papers

8:30

2aSA1. A multiscale computational approach for structural and acoustic medium-frequency vibrations. Herve Riou and Pierre Ladeveze (LMT Cachan, 61 Ave. du President Wilson, 94235 Cachan Cedex, France, riou@lmt.ens-cachan.fr)

Today, all major numerical modeling techniques for the analysis of medium-frequency vibrations are based on finite element or boundary element approaches. In order to account for small-wavelength phenomena, these techniques require huge numbers of degrees of freedom. High-frequency approaches, such as statistical energy analysis or any of its improved variations, do not appear to be suitable for medium-frequency vibrations: the resulting vibrational behavior is too smooth and, in general, the coupling loss factor cannot be calculated in a predictive way. The variational theory of complex rays (VTCR) is a predictive computational tool for dealing with medium-frequency vibration problems. This strategy is based on a multiscale treatment of the shape functions (propagative and evanescent waves) chosen to represent the solution. All the waves are taken into account and only the amplitudes of the waves (the slowly varying scale of the solution) are discretized. The boundary and interface conditions between substructures are weakly enforced using an *ad hoc* variational formulation. The performance of the VTCR for an acoustic problem will be assessed.

8:45

2aSA2. The (energy) loss factors in a dynamic system composed of a master dynamic system and an adjunct dynamic system that are coupled. G. Maidanik (Carderock Div., Naval Surface Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817)

The (energy) loss factor (η) of a dynamic system is defined such that together with the stored energy (E) it yields the power Π dissipated in that dynamic system; $(\omega\eta)E=\Pi$, where ω is the frequency and η, E, and Π are functions of ω. An externally driven master dynamic system is defined by the loss factor (η_0) and the modal density (ν_0). The relevant quantities in the absence of coupling are the external input power (Π_e^0), the stored energy (E_0^0), and the power (Π_0^0) dissipated. The conservation of energy demands $\Pi_e^0=\Pi_0^0$. In the presence of coupling Π_e^0 E_0^0 and Π_0^0 become the quantities Π_e, E_0, and Π_0, respectively. The externally driven master dynamic system is coupled to an adjunct dynamic system defined by the loss factor (η_s) and the modal density (ν_s). The relevant additional quantities are the net power (Π_s) transferred from the master dynamic system to the adjunct dynamic system and the stored energy (E_s) and the power (Π_s) dissipated in the adjunct dynamic system. The conservation of energy demands $\Pi_e=\Pi_0+\Pi_s$. A number of (energy) loss factors may be defined to describe the energetics of these coupled dynamic systems. A few asymptotic relationships among these loss factors are cited. [Work supported by ONR.]

9:00

2aSA3. Variance of energy and energy density in statistical energy analysis of complex systems. Vincent Cotoni (ESI R&D, 12555 High Bluff Dr., San Diego, CA 92130) and Robin Langley (Univ. of Cambridge, Cambridge, CB2 1PZ UK)

Standard statistical energy analysis (SEA) predicts the mean energy level in each subsystem of a complex system at high frequencies. The resulting energy is to be seen as averaged over an ensemble of similar systems (like nominally identical products coming off an assemble line). This prediction of the mean has been recently extended to the ensemble variance of the energy. The variance gives indications of how far from the ensemble mean can be the response of one particular member of the ensemble. Using few additional assumptions, the variance can be used to predict the confidence intervals around the SEA mean prediction, and can be extended to the variance of the response at a point (i.e., the energy density). Some numerical validations for several types of vibro-acoustic systems are presented.

9:15

2aSA4. A 3-D hierarchic finite-element tool with infinite elements for structural acoustic scatter modeling. Mario Zampolli, David S. Burnett, Alessandra Tesei (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19138 La Spezia, Italy), John B. BlottmanIII (Naval Undersea Warfare Ctr., Newport, RI 02841), and Timothy A. Westermann (2219 Parkland Cove, Round Rock, TX 78681)

Fully three dimensional prolate spheroidal acoustic infinite elements have been added to a steady state finite-element structural acoustics tool (FESTA), which is based on hierarchic polynomial shape functions. The infinite elements are used as an alternative to the Bayliss–Turkel approximate radiation condition to ensure outward propagation of the scattered and/or radiated acoustic fields surrounding one or more objects. The main advantages of the infinite elements compared to the Bayliss–Turkel condition are the reduction of the distance required between the object(s) and the boundary of the finite element computational domain, and meshes which can be used over a broader frequency band. Furthermore, the infinite element formulation adopted, which is based on the prolate spheroidal version of the Atkinson–Wilcox expansion, is ideal for modeling the scattering and/or radiation from elongated structures of particular interest to the underwater acoustics community. The performance of the tool is assessed by comparison to an axisymmetric thin-shell FE/virtual source code, and to a hierarchic axisymmetric FE code with infinite elements. Furthermore, the tradeoffs between the infinite elements and the prolate spheroidal Bayliss–Turkel conditions are assessed.

9:30

2aSA5. Solving for the vibro-acoustic interaction between a fluid and a structure with discontinuities by the boundary element method using local/global homogenization. Pavel Danilov and Donald Bliss (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708, pvd2@duke.edu)

The boundary element method (BEM) was used to determine the radiation properties of finite structures with periodically attached impedances. Due to the discontinuities and fine-scale structural wave motion, BEM would generally require small size elements. Local/global homogenization (LGH) was used in order to eliminate the discontinuities and short waves from the problem. LGH provides the global smooth problem, free of discontinuities, but containing all the information needed to describe the radiation. The global problem results in modification of BEM kernels in order to implement coupling with the local part. Instead of the exact solution, BEM used with LGH requires only the smooth global part, allowing fewer elements. Model problems show good accuracy of the method along with a significant reduction of computational costs.

9:45

2aSA6. Reducing the dispersion error in acoustic wave modeling with modified integration rules. Bin Yue and Murthy N. Guddati (Dept. of Civil Eng., North Carolina State Univ., Raleigh, NC 27695, byue@ncsu.edu)

Numerical analysis of transient acoustic wave propagation is often performed using finite element or finite difference methods along with central difference time stepping. Such analysis results in wave velocities that are different from exact ones. For conventional algorithms, this dispersion error is second order with respect to discretization parameters. In this paper, modified integration rules are combined with a modified central difference scheme to reduce this error to fourth order. Essentially, the locations of integration points used for evaluating the system matrices are selected such that the wave velocity error is minimized. It was found that the optimal integration points are square roots of 2/3 for the stiffness matrix, which eliminates the anisotropy in the second-order error. The optimal integration points for mass matrix, which depend on the wave velocity, the mesh size, and the time-step size, are also identified to completely eliminate the second-order error. The resulting dispersion error is thus reduced to fourth order. Numerical experiments illustrate that the proposed method has superior performance over existing methods, not only for structured square meshes, but also for unstructured meshes.

10:00–10:15 Break

10:15

2aSA7. Sound radiation from rectangular plates with elastic boundary restraints. Wen Li (Dept. of Mech. Eng., Mississippi State Univ., Mississippi State, MS 39762, li@me.msstate.edu)

Sound radiation from rectangular plates is of considerable interest to both researchers and engineers. Different methods and techniques have been developed for determining the sound radiated from plates with various complicating effects. However, the previous investigations are mostly focused on the simply supported plates, although it is widely known that the radiation efficiency of a plate will be strongly dependent upon its boundary condition. In this study, a general analytical method is presented for the calculation of the sound power radiation from a rectangular plate that is elastically restrained along its edges. The vibration of the elastically restrained plates is expressed in terms of a generalized Fourier series expansion. Thus, many of the existing solutions for the sound radiation from the simply supported plates can be directly applied to the plates with general elastic boundary restraints. The radiation characteristics are compared to the plates under different boundary conditions.

10:30

2aSA8. Acoustic radiation by a submerged source in shallow water—Influence of the boundaries. Emmanuel Charmes, Pierre-Philippe Beaujean (Dept. of Ocean Eng., Florida Atlantic Univ., Seatech, 101 North Beach Rd., Dania Beach, FL 33004), and Joseph M. Cuschieri (Perry Technologies, Riviera Beach, FL 33404)

The acoustic radiation and scattering characteristics of an acoustic source in an unbounded deep water environment are different from those in a shallow water environment, where one has to take into account the influence of the boundaries on the source radiation impedance. In this paper the influence of the waveguide (surface and bottom of the ocean in a shallow water environment) on the radiation from an elastic noise source is investigated by means of a superposition method. The radiated sound is initially determined from the scattering of an incident wave from the elastic target. Using the superposition method [A. Sarkissian, J. Acoust. Soc. Am. (1994)], the scattered field, including multiple scattering effects, is computed and represented by a series of point sources positioned inside the scatterer. A number of calibration points located at the surface of the scatterer are selected to determine the source strength of each point source using a least square approximation. Having obtained the source strength of each point source, the combined propagation field in the waveguide from all point sources and the influence back on the source are determined using normal mode or ray tracing solutions common in underwater sound propagation. [Work supported by ONR.]

10:45

2aSA9. Axisymmetric acoustic radiation from submerged prolate spheroidal shells. Jeffrey E. Boisvert (NAVSEA Newport, Newport, RI 02841) and Sabih I. Hayek (Dept. of Eng. Sci. and Mech., Penn State Univ., University Park, PA 16802)

The equations of motion for nonaxisymmetric vibration of prolate spheroidal shells of constant thickness were derived using Hamiltons principle [S. I. Hayek and J. E. Boisvert, J. Acoust. Soc. Am. **114**, 2799–2811 (2003)]. The shell theory used in this derivation includes shear deformations and rotatory inertias. The shell displacements and rotations were expanded in infinite series of comparison functions. These include associated Legendre functions in terms of the prolate spheroidal angular coordinate and circular functions in the azimuthal angle coordinate. For axisymmetric vibration of a submerged shell, the external (heavy) fluid loading impedance was computed using expansions of prolate spheroidal wavefunctions. The shell was excited by axisymmetric normal surface forces, including a point load at the shell apex and ring load at other locations. Far-field radiated pressure spectra are presented for several shell thickness-to-half-length ratios ranging from 0.005 to 0.1, and for various shape parameters, a, ranging from an elongated spheroidal shell ($a=1.01$) to a spherical shell ($a\sim 100$). The far-field directivity of acoustic radiation is presented at selected frequencies. [Work supported by the NUWC ILIR Program and the ONR/ASEE Summer Faculty Research Program.]

11:00

2aSA10. Local/global decomposition for fluid-loaded periodic structures. Pavel Danilov and Donald Bliss (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708, pvd2@duke.edu)

The problem of interaction between a fluid and a structure with periodic impedance attachments is considered. The full solution is divided into two coupled parts: the global part, providing overall structural motion and acoustic radiation, and the local part, containing small structural oscillations and evanescent pressure modes. Fluid loading in the local part results in additional structural inertia, which is conveniently accounted for in the global problem. A separate stand-alone equation is constructed for the global solution. The local part can be recovered afterwards. Specific conditions are imposed on the local and global parts, e.g., the global solution with finite wavenumber support is used to recover the radiating part of the

spectrum and the local solution is made to vanish at the impedances. Thus, the global solution describes radiating properties and structural response. Different conditions on the local/global decomposition are considered in order to enhance the accuracy of the method. Application to structures of finite extent is discussed. Model problems showed good accuracy of radiated pressure as well as structural response.

11:15

2aSA11. Analysis of wave dispersion in cylindrical shells with anisotropic layers by Rayleigh–Ritz expansions. Elizabeth A. Magliula (Dept. of Aero. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215), J. Gregory McDaniel, and Charles N. Corrado (Appl. Physical Sci. Corp., New London, CT 06320)

Recent advances in composite materials offer the potential to dramatically affect the dispersions of helical waves that propagate in layered cylindrical shells. Effective choices and configurations of these materials require rapid dispersion predictions that take into account general anisotropy and variations of elastic variables through the thickness of each layer. This presentation describes a solution to this problem that is based on a previously developed theory [J. G. McDaniel and J. H. Ginsberg, J. Appl. Mech. **60**, 463–469 (1993)] for the vibrations of cylindrical shells with isotropic layers. The approach uses propagating wave representations in the axial and circumferential coordinates. A series expansion with polynomial basis functions represents dependences on the radial coordinate and equations for the series coefficients are derived using the Rayleigh–Ritz method. These equations yield a dispersion relation that is solved for the complex-valued axial wavenumber at each frequency and circumferential harmonic. Examples illustrate the interface of this approach with optimization algorithms that identify layer designs possessing specified dispersion properties. [Work supported by ONR.]

11:30

2aSA12. Characterizing close fitting enclosures for fluid-loaded cylindrical sources. Joseph M. Cuschieri (Lockheed Martin MS2, Perry Technologies, 100 East 17th St., Riviera Beach, FL 33404, joe.cuschieri2lmco.com)

The influence of a close fitting enclosure on the radiated noise from a submerged (in water) acoustic source has been shown to be beyond the scope of simple close fitting enclosure expressions found in some textbooks, especially when dealing with relatively soft intermediate layers [Cuschieri, J . Acoust. Soc. Am. **115**, 2537 (2004)]. A more detailed analysis is thus warranted. Using a multilayer shell approach, with the innermost layer representing the source with uniform radial velocity, the middle layer representing the intermediate isolation layer and the outside layer representing the exterior cladding, with water fluid-loading on the outside of the exterior cladding, the insertion loss (IL) of the multiple layers is investigated for different intermediate layer materials, different exterior cladding layer characteristics and source size. IL results are presented as a function of frequency and layer/cladding characteristics. From the formulation the behavior of the intermediate layers is also obtained which help explain the IL results. [Work sponsored by ONR.]

TUESDAY MORNING, 16 NOVEMBER 2004 CALIFORNIA ROOM, 9:00 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication: Clinical and Developmental Perspectives on Speech (Poster Session)

Peggy B. Nelson, Chair
Communication Disorders, University of Minnesota, 164 Pillsbury Drive, SE, Minneapolis, Minnesota 55455

Contributed Papers

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

2aSC1. An intonational analysis of disfluency patterns in chronically stuttered speech. Timothy R. Arbisi-Kelm (UCLA, Los Angeles, CA)

While previous stuttering research has successfully revealed areas vulnerable to disfluency at the word level in stuttering, identifying the specific factors responsible for this instability has proved difficult; moreover, inconsistent results are complicated by a failure to control for the effects of phrasal prosody, which govern such word-level factors as lexical stress. The present experiment tested the hypothesis that disfluencies in stuttering are directly proportional to the prominence-level of a given production. Three stuttering subjects participated in an oral sentence-reading task testing a variety of sentence types while manipulating intonational factors such as pitch accent type and location. It was anticipated that pitch-accented syllables, representing a higher degree of stress in an intonation phrase than stressed but non-pitch-accented syllables, would be most prone to triggering disfluency, since they bear a greater level of prominence in the utterance. The results of the study confirmed the major hypothesis: in all of the comparisons between pitch-accented and non-pitch-accented positions of stress, the former attracted the highest rate of disfluent speech productions. This supports the principal hypothesis that intonationally prominent domains, not simply lexically stressed syllables, are a better indicator of unstable positions in stuttered speech.

2aSC2. Effects of a prosodic control approach for patients with motor speech disorders. Noriko Kobayashi, Hajime Hirose, Minako Koike, Yuki Hara (Sch. Allied Health Sci., Kitasato Univ., 1-15-1 Kitasato, Sagamihara, Japan 228-8555), Hiroki Mori, and Hideki Kasuya (Utsunomiya Univ., 7-1-2 Yoto, Utsunomiya, Japan 321-8585)

For patients with motor speech disorders, the Lee Silverman method (Ramig, 1997) has been known as an effective voice therapy method. In our clinical experiences, however, some Japanese patients could not easily produce very loud voice required in the method, presumably due to the cultural background. Therefore, in this study, a prosodic control approach called "the intonation emphasis therapy" was used as well as the Lee Silverman method for three types of patients. The disorder types were

amyotrophic lateral sclerosis (ALS), Parkinson's disease (PK), and olivo-ponto-cerebellar atrophy (OPCA). Acoustic analyses revealed wider $F0$ ranges after the intonation emphasis therapy than the Lee Silverman method for three types of the patients. In the perceptual judgments by speech therapists regarding articulation, voice quality, intonation, and abnormal impression of speech, better ratings were obtained for the intonation emphasis therapy than for the pretherapy speech and the Lee Silverman method in the patients with ALS and PK. However, the listeners perceived inappropriate intonation and increased abnormality for the OPCA patient after the intonation emphasis therapy. It was suggested that our therapy method was effective for patients with motor speech disorders unless the disorders were associated with poor muscular coordination such as OPCA.

2aSC3. Vowels and final consonant production by adults with Down's syndrome. Mary Fung, Rizzah Decopain, and Nancy McGarr (St. John's Univ., Jamaica, NY 11439, mcgarrn@stjohns.edu)

Speech production studies have suggested a typical voice quality as well as vowel and consonant errors in speakers with Down's syndrome. In this study, the production of bVt or bVd words produced by ten adults with Down's syndrome and age and gender matched controls was examined for influence of final consonant voicing status. Acoustic measures of vowel duration and consonant voicing were made. Listener judgments were also obtained. Contrasts between vowels pairs were maintained by most subjects, e.g., duration for tense vowels was greater than for lax, although duration for the test subjects productions was greater overall than for the controls. Similarly, consonant voicing contrasts were maintained, e.g., vowel duration preceding voiced consonants was longer than for voiceless, but again overall duration was longer for the Down's syndrome adults than the controls. The results of the acoustic analyses suggest that while the test subjects demonstrate knowledge for producing final voicing contrasts, aspects of fine coordination are not achieved. Intelligibility of final consonants was high and listener judgments generally agreed with acoustic measures. [Work supported by McNair Scholars' Program to St. Johns University.]

2aSC4. Articulatory characteristics of speakers with apraxia of speech during sentence production at different speaking rates. Michiko Hashi (Health Sci., Univ. of Hokkaido, Sapporo Hokkaido, Japan), Katharine Odell, H (Meriter Hospital, Madison WI USA), Ryoko Hayashi (Kobe Univ., Kobe, Japan), and Takeshi Nakayama, R (Health Sci., Univ. of Hokkaido, Sapporo Hokkaido, Japan)

The present study developed quantitative descriptions of articulatory movements in three speakers with apraxia of speech during sentences produced at different speaking rates. Point-parameterized articulatory data were obtained using the x-ray microbeam (XRMB) technique, and were compared with similar materials from 24 normal young-adult speakers of American English drawn from an existing XRMB database. Speed histories of markers attached to the lips and jaw during production of the test sentence "The other one is too big." spoken at slow, habitual and fast rates were analyzed to determine mean peak speed, the number of speed peaks, and the timing of particular speed peaks for each sentence replicate by each talker. An attempt was made to derive speed histories for the lip marker expressed relative to concurrent movements of the jaw, to evaluate the relative contribution of jaw speed to lip speed in both talker groups. A description and discussion of results from the analysis will emphasize speaking rate effects on articulator speed and inter-articulator timing.

2aSC5. Motherese and Chinese: Evidence of acoustic changes in speech directed at infants and foreigners. Monja Knoll and Maria Uther (Dept. of Psych., Univ. of Portsmouth, King Henry Bldg., King Henry 1 St., Portsmouth, PO1 2DY, UK)

Infant-directed speech (IDS) is characterized by hyperarticulation, increased pitch, and high emotional affect, which is in turn thought to reflect a linguistic and emotional role for IDS. If the linguistic role is an independent contributor of the changes in IDS, then similar hyperarticulation should also occur in foreigner-directed speech (FDS) but without positive affective changes. To test this, nine mothers were recorded talking to their infants, and British- and foreign-adult confederates. Mothers were provided with toys to elicit the target words shark, sheep, and shoe, containing the corner vowels /a/, /i/, and /u/. Speech samples were analyzed to determine mean pitch and formant 1 and 2 ($F1/F2$) values of target vowels. Low-pass filtered speech samples of the mother's interactions in each condition were rated on positive and negative vocal affect by 24 independent raters. Results showed that mothers hyperarticulated their vowels (indexed by $F1/F2$ values) in both IDS and FDS relative to ADS. Furthermore, mean pitch was highest in IDS compared to FDS and ADS. Positive affect was highest in IDS and lowest in FDS compared to ADS. These data suggest a linguistic function for prosody modifications in FDS and IDS that is independent of affective changes.

2aSC6. Strategies used to increase speech clarity by normal-hearing children. Dana L. Ide-Helvie, Elizabeth A. McCleary, Sarah C. Sullivan, Andrew J. Lotto (Boys Town Natl. Res. Hospital, 425 North 30th St., Omaha, NE 68131, lottoa@boystown.org), and Maureen B. Higgins (Boys Town Natl. Res. Hospital, Omaha, NE 68131)

It is common among clinicians to ask children to produce their best speech during intervention. However, it is unclear whether children know how to make their speech clearer. The strategies used by children with and without hearing loss have implications for maximizing intelligibility and for understanding the development of communication competency. As a first step toward this understanding, children (7 to 12 years of age) with normal hearing were asked to read ten simple sentences. They were told that they were testing a new computer program designed to recognize speech. There was, in fact, no recognition program and each child received the same output feedback. After providing normal speech to allow the program to get used to their voices, they subsequently produced their best and then very best, very clearest speech in order to see how accurate the recognition program could be. Acoustic analyses (intensity, fundamental and first two formant frequencies for all vowels, as well as sentence, vowel, and VOT durations) were performed on recorded waveforms from each repetition in order to determine what the children were varying to comply with best speech instructions. The results demonstrate large individual differences in strategies and persistent gender differences. [Work supported by NIH.]

2aSC7. Speech perception in noise among language-impaired individuals. Ratree Wayland and Linda Lombardino (Program in Linguist., Univ. of Florida, Dept. of Commun. Sci. and Disord., Univ. of Florida, Gainesville, FL 32611-5454)

This study compared and contrasted the ability to process speech signals between language-impaired individuals and their age-matched normal controls. Participants were administered a speech categorization task and a sentence processing task. A synthetic /ba–da/ continuum varying in second formant onset frequencies was used in the speech categorization task and 64 short (4–5-word-long) English sentences were used in the sentence processing task. The stimuli were presented with and without background noise to participants for categorization and identification. The /ba–da/ categorical boundaries and the number of target words correctly identified by two groups of participants under the two presentation conditions were compared. A connection between phonological processing and language impairment was discussed.

2aSC8. An echological theory of language learning. Francisco Lacerda and Ulla Sundberg (Dept. of Linguist., Stockholm Univ., Stockholm, Sweden)

An ecological theory of language acquisition (ETLA), attempting to account for the linguistic development during about the first 18 months of age, is proposed in this contribution. The theory does not assume that the infant is endowed with specialized linguistic devices or strategies at the onset of life. ETLA considers instead the multi-sensory aspects of the adult-infant interaction and the typical ecological setting of that interaction. Rather than focusing on the speech signal *per se*, ETLA considers the infant's multi-sensory exposure to the phonetic, prosodic, syntactic and semantic characteristics along with visual, tactile, olfative and other sensory information as a key to the spontaneous emergence of linguistic structure early in life—a structure implicit in the adults use of the language, and that is partly simplified by the infant's limited production, perception and representation capacities. A functional model illustrating how such general-purpose mechanisms interacting with the typical sensory input available to the infant may lay the ground for linguistic structure will be presented for discussion. [Research supported by grants from the Swedish Research Council and the Bank of Sweden Tercentenary Foundation.]

2aSC9. Children's reliance on spectral information for understanding speech in noise. Peggy Nelson and Yingjiu Nie (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, nelso477@umn.edu)

Children have significantly greater difficulty understanding in noisy situations than do adults. This has serious implications for the design and structure of classrooms and for communicating effectively with young children. The reasons for children's susceptibility to noise are not well understood. One factor may be children's reliance on frequency cues for speech understanding. This project investigated children's perception of speech in noise by systematically evaluating the perceptual importance of speech frequency information. Typically developing children (ages 7 to 13 years) and adults were tested using full spectrum speech as well as 4-, 8-, and 12-band simulations of cochlear implant processing (after Shannon *et al.*, 1995). Stimuli were color and number words selected from the coordinate response materials (Brungart, 2001), and were presented in quiet and in pink noise at +5-dB signal-to-noise ratio. Preliminary results support the hypothesis that children rely more on the frequency content of speech than do adults for understanding speech in noise, and that children need fuller frequency representations of speech than do adults to segregate speech from background noise. [Work supported by the Graduate School of the University of Minnesota.]

2aSC10. Phonemic contrasts in a mutiple talker task with hearing impaired listeners. Mark A. Ericson (Air Force Res. Lab., Wright–Patterson AFB, OH) and Pamela J. Mishler (Dept. of Veteran Affairs, Dayton, OH)

Word identification ability of mild to moderately hearing impaired listeners was measured in both spatial and nonspatial multiple talker tasks. A digital recording of the modified rhyme test was made for three male talkers. The beginning of the test word in each phrase was synchronized using Cool Edit and equalized to the same average rms power. The phrases were processed for binaural presentation using nonindividualized HRTFs with Tucker Davis Technology System 3 equipment and presented over Sennheisser 600 headphones. The phrases were presented in random order at a self-paced manner. Five mild and five moderately hearing impaired subjects, ranging in age from 24 to 54 years, participated in this study. Results were analyzed by six attributes: place, voicing, nasality, frication, duration, and stops. Phoneme identification performance was ranked based on most salient (most information transferred) to least salient (least information transferred). Spatial separation had no effect on the relative importance of the speech attributes. Duration was the most salient cue for hearing impaired listeners, followed by voicing and frication. Stops and nasality were weak cues and place was by far the least important cue.

2aSC11. Perceptual overshoot in listeners with cochlear implants. Radhika Aravamudhan and Andrew J. Lotto (Boys Town Natl. Res. Hospital, Omaha, NE 68131)

Perceptual overshoot (PO), a phenomenon in which the boundary for perceived vowel categories shifts as a result of preceding formant transitions, has been demonstrated previously with synthetic vowels in CV contexts in normal-hearing individuals. In one of the previous studies by the author, the same phenomenon was tested with sinewave analogs that mimicked the synthetic vowels. The results demonstrated that PO could be elicited for sinewave analogs after training on categorization of sinewave steady states. These findings suggest that PO may be partly mediated by general processes in the auditory system. In the current study, subjects with cochlear implants were presented with synthetic vowel and sinewave continua with and without transitions. It was predicted that PO would be greatly diminished or absent for this population because of the degraded nature of the spectral input. Implications of the results for theories of speech perception will be presented.

2aSC12. Fundamental frequency and the intelligibility of competing sentences with cochlear implant processing. Ginger S. Stickney (Dept. of Otolaryngol., Univ. of California, Irvine, 364 Med Surg II, Irvine, CA 92697), Janice Chang (Univ. of California, Berkeley, CA 94720-1762), Peter F. Assmann (Univ. of Texas at Dallas, Richardson, TX 85083-0688), and Fan-Gang Zeng (Univ. of California, Irvine, CA 92697)

Speech perception in the presence of another competing voice is one of the most challenging tasks for cochlear implant users. Several studies have shown (1) that the fundamental frequency ($F0$) is a useful cue for segregating competing speech sounds and (2) the $F0$ is better represented by the temporal fine structure than the temporal envelope. However, current cochlear implant speech processing algorithms emphasize the temporal envelope information and discard the temporal fine structure. In this study, speech recognition was measured as a function of the $F0$ separation of the target and competing sentence in normal-hearing and cochlear implant listeners. For the normal-hearing listeners, the combined sentences were processed through either a traditional cochlear implant simulation or a new algorithm which additionally extracts a slowed-down version of the temporal fine structure. The results showed no benefit of increasing $F0$ separation for the cochlear implant or traditional simulation groups. However, as found with unprocessed sentences presented to normal-hearing listeners, the new algorithm resulted in gradual improvements with increasing $F0$ separations. These results demonstrate that inadequate coding of the temporal fine structure in current speech processing algorithms may complicate the segregation of competing speech sounds. [Work supported by NIDCD F32DC05900 and R01DC02267.]

2aSC13. Individual differences in auditory discrimination of spectral shape and speech-identification performance among elderly listeners. Mini N. Shrivastav[a], Larry E. Humes, and Diane Kewley-Port (Dept. of Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, mnarendr@csd.ufl.edu)

Speech-understanding difficulties observed in elderly hearing-impaired listeners are clearly associated with recognition and discrimination of consonants, particularly within consonants that share the same manner of articulation. Spectral shape is an important acoustic cue that serves to distinguish such consonants. The present study examined whether indi-

vidual differences in speech understanding among the elderly could be explained by individual differences in spectral-shape discrimination ability. This study included a group of 20 elderly hearing-impaired listeners, with a group of young normal-hearing adults also included for comparison purposes. All subjects were tested on speech-identification tasks, with natural and computer-synthesized speech stimuli, and on a series of spectral-shape discrimination tasks. The young normal-hearing adults performed better than the elderly listeners on many of the identification tasks and on all but two discrimination tasks. Regression analyses on data from the elderly listeners revealed moderate predictive relationships between some of the spectral-shape discrimination thresholds and speech-identification performance. The results indicated that when all stimuli were at least minimally audible, some of the individual differences in the identification of natural and synthetic speech tokens by elderly hearing-impaired listeners could be attributed to the differences in their spectral-shape discrimination abilities for similar sounds. [a] Currently at Dept. of Commun. Sci. and Disord., 336 Dauer Hall, Univ. of Florida, Gainesville, FL 32611.

2aSC14. Interactions between the frequency allocation, stimulation mode, and pitch perception on speech recognition by cochlear implant listeners. John GalvinIII and Qian-Jie Fu (House Ear Inst., 2100 W. 3rd. St., Los Angeles, CA 90057)

In cochlear implants, speech recognition depends strongly on the acoustic frequency-to-electrode assignment. In a previous experiment, the pitch of an electrode was shown to vary with stimulation mode, especially for widely spaced electrode pairs. In the current study, Nucleus-22 users' speech recognition was measured as functions of the stimulation mode and frequency allocation. Two stimulation modes were tested: BP+1 and ''mixed-mode'' [the active electrode was fixed and the return electrode varied, i.e., (2,22), (2,21), etc.]. A range of frequency allocations was tested to measure the effect of spectral mismatch between the acoustic input and the electrode configurations. Preliminary results showed a differential effect of frequency allocation, depending on the stimulation mode used in the speech processor. For the BP+1 configuration, multi-talker vowel recognition decreased significantly as the frequency allocation was shifted from Table 9 (150–10 823 Hz) to Table 3 (85–6184 Hz). However, for the mixed-mode processor, performance improved as the allocation was shifted from Table 3 to Table 9. Because Table 3 may have produced less spectral mismatch (given the generally lower pitch percepts for widely spaced electrode pairs), mixed mode processors may be used to extend the pitch range and thereby map more low-frequency spectral content to apical electrodes.

2aSC15. Cortical networks underlying audio-visual speech perception in normal-hearing and hearing impaired individuals. Julie J. Yoo (Speech and Hearing Biosci. and Technol. Program, Harvard-MIT Div. of Health Sci. & Technol., Cambridge, MA 02139), Frank H. Guenther (Boston Univ., Boston, MA 02215), and Joseph S. Perkell (MIT, Cambridge, MA 02139)

Functional magnetic resonance imaging (fMRI) was used to investigate the brain activity underlying audio-visual speech perception in normal-hearing and congenitally deaf individuals. Data were collected while subjects experienced three different types of speech stimuli: audio stimuli without visual input, video of a speaking face without audio input, and video of a speaking face with audio input. A control condition consisted of viewing a blank screen. The stimuli were vowels or CVCV syllables, presented in different blocks. Active brain regions for normal-hearing subjects during the visual-only condition included visual cortex, angular gyrus, fusiform gyrus, and auditory cortex, as well as premotor areas in the frontal cortex. The pattern of activation found for deaf subjects while viewing visual-only stimuli was similar to that of normal hearing subjects, but showed distinctly more activity in the right hemisphere (for both vowels and CVCVs), and far less activity in premotor and parietal cortex. Interestingly, the pattern of activity for deaf subjects in the visual-only case was found to be similar to normal-hearing subject activation in the audio-visual case. Finally, effective connectivity analyses were done to investigate connectivity between brain regions in the different conditions for both groups of subjects. [Research supported by NIDCD.]

2aSC16. Can context diminish the effects of rapid speech recognition in older listeners? Izumi Furukawa, Nancy Vaughan, and Daniel Storzbach (Natl. Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97207)

This is a part of a larger study investigating the effects of cognitive slowing on speech perception among older adults. In addition to neurocognitive tests (working memory, attention, and speed of processing), time-compressed speech was used to evaluate the effects of age on rapid speech recognition. Two different speech materials (with and without contextual cues) were prerecorded and time-compressed to four different rates (40%, 50%, 60%, 65%) in order to increase the demand on processing speed. All sentences were presented monaurally through MedRx Otowizard via ER3 insert phones to older adults (50 to 75 year olds) with normal hearing and with mild to moderated hearing loss. For the listeners with hearing loss, the master hearing aid function with NAL-NP target was used. At the slower time-compression rate (40%), there was no significant difference between the two groups. However, at faster rates, the hearing loss group performed significantly worse than the normal hearing group regardless of contextual cues. These results suggest that neither of the older groups was able to benefit from the use of context in these rapid speech materials. Implications of these speech recognition test results will be discussed with regard to the neurocognitive findings.

Session 2aSP

Signal Processing in Acoustics and Underwater Acoustics: Signal Processing Arrays with Many Elements in Novel Configurations or Novel Environments Part I

Jens M. Meyer, Cochair
20 River Terrace, New York, New York 10282

David I. Havelock, Cochair
National Research Council, IMS/ASP, Montreal Road, Ottawa, Ontario, K1A 0R6, Canada

Invited Papers

8:30

2aSP1. Processing microphone arrays for spatial selectivity in three dimensions. J. L. Flanagan (Ctr. for Adv. Information Processing, Rutgers Univ., 96 Frelinghuysen Rd., Piscataway, NJ 08854, jlf@caip.rutgers.edu)

Reverberation degrades the performance of conventional beamformers. Matched-filter processing combats multi-path distortion and provides improved quality by utilizing signal information arriving from different paths. Implicit is spatial selectivity in three dimensions—which enables an array to "reach over" intervening sources of interference. The cost is arithmetic, storage, and *a priori* characterization of the enclosure. Source-to-receiver transmissions are described by impulse responses, which are employed in causal time-reversed form to accomplish matched filtering. Work conducted earlier for the National Science Foundation is summarized and demonstrated, along with a description of algorithms for automatic tracking of moving talkers. Array configuration and placement remain challenges in geometric optimization.

9:00

2aSP2. A scalable spherical microphone array for spatial sound capture. Gary W. Elko (Avaya Labs, 233 Mt. Airy Rd., Basking Ridge, NJ 07920, gwe@ieee.org) and Jens Meyer (mh acoustics, New York, NY 10282)

With recent popular consumer acceptance of surround-sound audio, there is the need to develop new technologies that will allow for more accurate sound-field recording commensurate with or exceeding current multichannel audio playback systems. Real-world playback systems can have widely varying geometries and number of channels due to space and aesthetic requirements. Thus, there is a need to have a general recording scheme that allows for computationally simple modifications of the signals to optimize the playback for any geometry and number of playback speakers. One would also like to have a recording system that is scalable for future advances in spatial audio playback technology. A spherical microphone array has been proposed as a solution to these requirements. A spherical array has the desired property of enabling a beam pattern that is steering direction invariant with a relatively simple beamformer structure. The spherical array topology enables scalability and an elegant and mathematically straightforward beamformer design for efficient sampling and processing of the acoustic field. This talk will describe how the spherical array beamformer decomposes the sound field into compact multichannel representations that allow computationally efficient transformations to uniquely tailor the playback signals to any specific playback geometry.

9:30

2aSP3. Two-dimensional wave field decomposition using circular microphone arrays and its application to acoustic source localization. Heinz Teutsch and Walter Kellermann (Multimedia Commun. and Signal Process., Univ. of Erlangen-Nuremberg, Cauerstr. 7, 91058 Erlangen, Germany, teutsch@lnt.de)

Two-dimensional wave fields can be used as reasonable models for propagating acoustic sound fields in closed rooms where ceiling and floor reflections are sufficiently attenuated. A natural way of analyzing a 2D wave field is to decompose it into an orthogonal set of eigensolutions to the acoustic wave equation in cylindrical coordinates, i.e., the cylindrical harmonics. These harmonics can be shown to be the coefficients of a Fourier series expansion applied along a circular aperture that is located within the 2D wave field under observation. Circular microphone arrays and, in particular, microphones mounted in a rigid cylindrical baffle that perform the cylindrical harmonics decomposition will be presented and its advantages and limitations will be discussed. As a specific example for utilizing the wave field decomposition approach, its application to acoustic source localization will be considered. By exploiting structural similarities between the response of linear microphone arrays and the decomposed response of circular microphone arrays, most well-known subspace tracking algorithms can be applied to this problem with only minor modifications. It will be shown that acoustic source localization based on wave field decomposition has the potential to track multiple simultaneously active sources in the array's full 360° field of view.

10:00–10:30 Break

10:30

2aSP4. Sectorized beam-base adaptation for long line arrays. Henry Cox (LM Orincon Defense, 4350 North Fairfax Dr., Ste. 470, Arlington, VA 22203)

Long line arrays of many elements present special problems for adaptive beamforming. The long travel time for an endfire signal to traverse the array requires the use of narrow frequency bins to avoid a significant phase change within a frequency bin. The narrow beamwidth near broadside makes the array sensitive to small bearing charges that occur within the coherent integration time so that adaptation must take place rapidly. These two effects combine to cause a snapshot-starved situation. A beam-based adaptation approach is suggested in which the first step is to form shaded conventional far-field beams. This is followed by a sectorized adaptive combination of beams in a sector to provide adaptive near-field and far-field beams. For a sector near broadside where the beams are very narrow and sensitive to motion but the travel time across the array is short, less temporal averaging and more frequency averaging is used. Near endfire, where the beams are wide but the travel is long, narrow frequency beams are used with longer temporal averaging. The approach is applied to the case of strong far-field interference masking a weak near-field signal. The approach is computationally efficient as well as providing improved snapshot support for adaptation.

11:00

2aSP5. Time-reversal arrays in underwater acoustics. Philippe Roux, Tuncay Akal, W. S. Hodgkiss, W. A. Kuperman, Hee Chun Song (Marine Physical Lab., Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA, 92093), and Mark Stevenson (NATO Undersea Res. Ctr., La Spezia, Italy)

Since 1996, several time-reversal experiments have been performed in shallow water acoustics involving source and receive arrays made of many elements. This equipment allowed us to reach different goals among which were long-range time-reversal focusing, passive and active multiple input multiple output communications, bottom refocusing from reverberation, acoustic barrier detection, comparison between reciprocal and nonreciprocal time-reversal and shallow water tomography. This presentation will review this work with emphasis on array signal processing.

Contributed Papers

11:30

2aSP6. Experimental demonstration of time reversed reverberation focusing in an oceanic waveguide. Karim Sabra, Philippe Roux, Hee-Chun Song, William Hodgkiss, William Kuperman, Tuncay Akal (MPL-SIO, UCSD. 9500 Gilman Dr., La Jolla, CA 92093-0238), and Mark Stevenson (NATO Undersea Res. Ctr., La Spezia, Italy)

The robust focusing and pulse compression provided by time reversal techniques may be exploited for active sonar. Backscattering from the rough water–bottom interface can be used as a surrogate probe source [Lingevitch *et al.*, J. Acoust. Soc. Am. **111**, 2609–2614]. This method uses a selected time-gated portion of the reverberation signals (or backscattered signals) to provide a transfer function between a time-reversing array (TRA) and a corresponding range interval on the bottom. The potential of a TRA for studying the application of shallow water bottom scattering is investigated experimentally using both sea-data collected in July 2004 in the Mediterranean sea off the Italian coast, and ultrasonic water tank data.

11:45

2aSP7. Processing of reverberation data from 400–3500 Hz using line arrays with left/right discrimination. John Preston (Appl. Res. Lab., The Penn State Univ., State College, PA, 16804, preston@ciao.arl.psu.edu)

The author recently participated in two experiments using directional towed arrays. One was the 2004 boundary characterization experiment near the Malta Plateau. That experiment was led by the NATO Undersea Research Centre (NURC). The area is rich in clutter objects like wrecks and mudvolcanoes and has some sub-bottom features that may be important. Sources were monostatic coherent pulses and SUS. The receiver was the NURC cardioid array. The other experiment was ONR's 2003 Geoclutter effort to study shallow-water bottom reverberation and clutter in the STRATAFORM off New Jersey. That experiment was led by M.I.T. Sources were bistatic coherent pulses. The receiver was the five-octave research array (FORA). The STRATAFORM is known to have benign surface morphology but strong clutter is observed. Some highlights of the reverberant returns are discussed that include the correlation of returns with manmade targets and probable fish schools. The purpose of this work is to assess the directional characterization of the observed clutter and reverberation. The cardioid arrays should yield good directional estimates of reverberation sources above ~1600–1800 Hz. Examples from the data analysis are presented using a cardioid beamforming algorithm developed by NURC. [Work supported by ONR Code 32, Grant N00014-03-1-0113.]

Session 2aUW

Underwater Acoustics: Reverberation, Scattering and Boundary Interaction

Kevin D. LePage, Chair
Naval Research Laboratory, Code 7144, 4555 Overlook Avenue, SW, Washington, DC 20375

Contributed Papers

8:30

2aUW1. Effect of multipath propagation on the statistics of reverberation in shallow-water environments. Kevin D. LePage (Naval Res. Lab., Code 7144, 4555 Overlook Ave. SW, Washington, DC 20375)

The envelope of direct-path scattered energy has been shown theoretically to deviate from Rayleigh statistics for small ensembles of scatterers with non-Gaussian amplitude distributions [Abraham and Lyons, IEEE J. Ocean. Eng. (2002)]. Here, we investigate the effects of multipath propagation, scatterer correlation length scale, scatterer amplitude distribution, and sonar source and receiver characteristics on the distributional properties of reverberation envelopes in shallow-water environments. For this study we employ a broadband version of the coherent time-domain R-SNAP reverberation model. Model predictions indicate that the non-Rayleighness of shallow-water reverberation caused by distributions of homogeneously distributed diffuse scatterers is time dependent, and sensitive to the number of resolved multipath in the waveguide, the source bandwidth, and the source/receiver beamwidths of the sonar system under consideration. [Work supported by ONR.]

8:45

2aUW2. Unified acoustic model for simulating propagation and scattering effects in oceanic waveguides. Ildar M. Tamendarov and Natalia A. Sidorovskaia (Phys. Dept., Univ. of Louisiana at Lafayette, UL BOX 44210, Lafayette, LA 70504)

The paper addresses the recent developments of Shallow Water Acoustic Modal Propagation (SWAMP) model (http://www.ucs.louisiana.edu/~nxs7560/swamp.html) to account for the scattering events along the acoustic signal propagation path. The theoretical and numerical aspects of SWAMP extension to model the acoustic pulse scattering by a spherical elastic shell in an inhomogeneous oceanic waveguide within the T-matrix approach are discussed. The algorithm utilizes the incident modal functions obtained by SWAMP for an empty waveguide in the spherical representation and the free-field T-matrix. The theoretical foundation for the approach is based on the work by Hackman and Sammelmann [J. Acoust. Soc. Am. **80**, 1447–1458 (1986)]. The T-matrix calculations are implemented in MATLAB software. The interpretation of the scattering event as the modal transformation through the T-matrix is attempted. The numerical studies based on the new model are presented. [Work supported by the Louisiana Board of Regents Support Fund, Contract No. LEQSF(2001-04)-RD-A-38.]

9:00

2aUW3. Unified model for 3-D scattering and forward propagation in a stratified ocean waveguide with random seabed inhomogeneities. Purnima Ratilal and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

The field forward propagated through a waveguide containing random volume inhomogeneities in the seabed or random seafloor roughness is modeled using a modal formulation that analytically expresses the effects of dispersion, attenuation and redistribution of modal energy in the forward direction [Ratilal and Makris, J. Acoust. Soc. Am. **114**, 2428 (2003)]. The scatter function density of the random medium is modeled using the Rayleigh–Born approximation. The model is used to determine whether scattering from seabed inhomogeneities can have a noticeable effect on the forward propagated field in the New Jersey Strataform area. Relevant statistical properties of the seabed are obtained from geophysical surveys [Goff, Marine Geol. (2004)] and are further constrained by fitting modeled bottom backscatter with data obtained during the Main Acoustic Clutter Experiment of 2003. The results indicate that scattering from seabed volume inhomogenieties has a negligible effect on the forward propagated field in the 300-Hz to 2-kHz frequency range, but rough surface scattering may be non-negligible.

9:15

2aUW4. Scintillation of short duration records. Steven Lutz, R. Lee Culver, and David Bradley (Appl. Res. Lab, Penn State, P.O. Box 30, State College, PA 16804)

We have investigated propagation of sound in the ocean using the theory of wave propagation in random media. The theory classifies specific source/receiver geometries as either unsaturated, partially saturated, or saturated. In the unsaturated regime the variance of the intensity fluctuations will generally be quite small such that the scintillation index is less than unity. The partially saturated regime can give rise to scintillation indices larger than unity. In the fully saturated regime the scintillation index asymptotically approaches unity. We have conducted an experiment in which the two-dimensional sound-speed field was measured directly using a towed CTD chain, and relatively short acoustic records were captured using a number of source/receiver pairs. The length scales and index of refraction variation place the experiment in the saturated regime. However, measured scintillation indices are much less than 1. We provide an explanation for this difference and utilize local current measurements to bring the theory into agreement with the data. Our results are applicable to operational sonar systems, where short duration records are ordinarily utilized to transmit information or detect targets. [Work sponsored by ONR Code 321US.]

9:30

2aUW5. Analysis of digital side-scan sonar survey data in support of KauaiEx. Jerald W. Caruthers (Univ. of Southern MS, Stennis Space Ctr., MS 39529) and The KauaiEx Group

A side-scan sonar (SSS) survey was conducted off the northwest coast of Kauai, HaI, in support of the high-frequency, channel-characterization experiment (KauaiEx). The SSS used in this survey was a modified system that operates alternately at 150 and 300 kHz and produces digital data as well as standard tiff images of the seafloor. This paper discusses analyses of the digital backscattering data. Previous reports discussed the system characteristics and analyses of standard tiff images produced by the SSS as originally configured. The digital data are high-resolution in space and in dynamic range and offer improved interpretations of the character of the bottom in the KauaiEx range. The data in the range yield interpretations of sand ripples parallel to the depth contours with wavelengths of about 1 m and heights of a few centimeters. These ripples appear to cover the entire KauaiEx range. Data taken at a shallow-water site nearby provide additional information on bottom characteristics that include lava flows, mud, and sand. [Work supported by the Ocean Acoustics Program of ONR.]

9:45

2aUW6. Broadband match-field processing applied in a highly reverberant environment. Natasha A. Chang and David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, 1231 Beal Rd., 2019 Lay Auto Lab., Ann Arbor, MI 48109)

Application of matched-field processing (MFP) techniques to locate and identify broadband acoustic sources is of interest for studying low event rate cavitation and other hydroacoustic noise sources in water tunnels and other reverberant environments. Here, the main challenge lies in fully exploiting the potentially hundreds of kHz of bandwidth of the cavitation signal to refine the source location estimate. This presentation reports on the preliminary application of MFP in a reverberant enclosure that geometrically mimics the test section of a laboratory water tunnel. For this effort a sound projector and one to four hydrophones are used to make narrow-band and broadband sound field measurements within the enclosure. The requisite field model was initially based on a sum of modes assuming pressure release boundaries, but was empirically modified to accommodate finite impedance boundaries. The model modification allowed the enclosure's impulse response to be accurately modeled from 3 to 11 kHz and the incoherent Bartlett processor to locate the source when it emitted a series of continuous single tone signals from 3 to 12 kHz. Extension of this effort to signal pulses, a ray-based field model, and water-tunnel testing and sea trials will be discussed as well. [Work sponsored by ONR.]

10:00

2aUW7. Gaussian beam summation formulation for rough surface scattering in complex configurations. Goren Gordon, Ehud Heyman (School of Elec. Eng., Tel Aviv Univ., Tel Aviv 69978, Israel, heyman@eng.tau.ac.il), and Reuven Mazar (Ben-Gurion Univ. of the Negev, Beer-Sheva 84105, Israel)

A Gaussian beam summation (GBS) representation for wave propagation problems in the presence of rough surfaces is introduced. In this formulation, the coherent and incoherent scattered fields are decomposed into a discrete phase space summation of Gaussian beams (GBs) that emanate from a discrete set of points and directions on the rough surface. The formulation therefore involves stochastic GB-to-GB scattering matrices for the coherent and incoherent fields, and deterministic GB propagators. It benefits from the simplicity and accuracy of the latter, and can be used in applications involving propagation in complex environments and in inverse imaging. The coherent and noncoherent GB2GB scattering matrices are calculated from the statistical moments of the rough surface scattering amplitude, which is given either analytically or empirically. As an example, closed form expressions are calculated within the small perturbation regime. The GB2GB matrices are then used to analyze the propagation along rough-surface waveguides and to estimate the bi-static reverberations. The results clearly demonstrate and explain the phase-space footprints of the stochastic multiple scattering events and of the deterministic propagation between the surfaces.

10:15–10:30 Break

10:30

2aUW8. Acoustic scattering by axisymmetric finite-length bodies. D. Benjamin Reeder (Naval Postgrad. School, Monterey, CA 93943) and Timothy K. Stanton (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

A general scattering formulation is presented for predicting the far-field scattered pressure from irregular, axisymmetric, finite-length bodies for three boundary conditions—soft, rigid, and fluid. The formulation is an extension of a two-dimensional conformal mapping approach [D. T. DiPerna and T. K. Stanton, J. Acoust. Soc. Am. **96**, 3064–3079 (1994)] to scattering by three-dimensional finite-length bodies. This extended formulation, which is inherently numerically efficient to evaluate, involves conformally mapping the surface of an irregular, finite-length body to a new, orthogonal coordinate system in which the separation of variables method may be used to solve the Helmholtz equation and satisfy the boundary conditions. Extensive comparisons with previously published results using other formulations are presented. This formulation is shown to be very accurate in the prediction of scattering from smooth, symmetric bodies for a wide range of frequencies (Rayleigh through geometric scattering region), scattering angles (monostatic and bistatic), aspect ratios, and for each of the three boundary conditions listed above. Reasonable agreement has also been demonstrated for irregular, realistic shapes with soft boundary conditions. [Work supported by ONR.]

10:45

2aUW9. Efficient computation of acoustical scattering from N spheres via the fast multipole method accelerated flexible generalized minimal residual method. Nail A. Gumerov and Ramani Duraiswami (Inst. for Adv. Comput. Studies, Univ. of Maryland, A.V. Williams Bldg., College Park, MD 20742, gumerov@umiacs.umd.edu)

Many problems require computation of acoustic fields in systems consisting of a large number of scatterers, which can be modeled as spheres (or enclosed by them). These spheres can have different sizes, can be arbitrarily distributed in three dimensional space, and can have different surface impedance. Solution of this problem via direct T-matrix approach [Gumerov and Duraiswami, J. Acoust. Soc. Am., **112**, 2688–2701 (2002)] is practical only for relatively low number of scatterers, N, since its computational complexity grows as $O(N^3)$. We developed and implemented an efficient computational technique, based on an iterative solver employing a flexible generalized minimal residual method with a right preconditioner. Matrix-vector multiplications involving a large system matrix and the preconditioner are sped up with the aid of the multilevel fast multipole method. We tested the accuracy, convergence and complexity of the method on example problems with $N \sim 10^4$ (millions of unknowns). These tests showed that the method is accurate for a range of frequencies, and experimentally scales as $O(N^{1.25})$. The method has substantial advantages in speed and convergence compared to the reflection method reported earlier [Gumerov and Duraiswami, J. Acoust. Soc. Am., **113**, 2334 (2002)]. [Work supported NSF Awards 0086075 and 0219681, which are gratefully acknowledged.]

11:00

2aUW10. Acoustic intensity measurements involving forward scatter from prolate spheroids. Brian R. Rapids and Gerald C. Lauchle (Grad. Prog. in Acoust. and Appl. Res. Lab., Pennsylvannia State Univ., State College, PA 16804)

Underwater acoustic measurements made by scalar pressure sensor are only able to provide an estimate of the magnitude of the total intensity associated with an equivalent plane wave. This equivalence is based upon an assumption that the relative phase between pressure and particle velocity is identically zero. True intensity sensors measure simultaneously the acoustic pressure and components of particle velocity (or related quantity such as acceleration, displacement, or pressure gradient) at a single coordinate in space. The measurement of both pressure and velocity provides the magnitude of acoustic intensity as well as the relative phase between acoustic pressure and velocity. Numerical computations of a steady-state acoustic field perturbed by the presence of a rigid spheroid indicate that the plane wave equivalence may not be a perfect assumption in certain bistatic geometries and that additional information regarding the total acoustic field can be observed only with true intensity sensors. A varnished prolate spheroid constructed from red oak was employed as part of an underwater scattering experiment. A dual axis p-a probe was employed at frequencies near 10 kHz to test the theoretical hypotheses. Theoretical and experimental results will be discussed. [Work supported by ONR, Code 321MS under Grant No. N00014-01-1-0108.]

11:15

2aUW11. Experimental and theoretical target strength of fluid filled spheres. David M. Deveau (PSC 1012, Box 701 FPO, AA 34058)

Investigation into the scattering nature of surfaces or other physical objects often requires the use of measurement systems which cannot always be well controlled. This lack of control can be compensated for by calibrating the resulting measurements against a known target. While these targets can be any object, the goal is to use a target that has a stable scattering response and is independent of angle. The ideal shape is that of a sphere but even this can be improved with the addition of internal fluids that focus and temperature stabilize the scattering response. A scatter response model of a sphere has been developed and used to design four thin walled spheres, each with a different diameter and filled with a focusing fill fluid (Fluorolube). One pair of spheres was measured in an ocean environment while the second pair was tested in a controlled test tank from 5 to 50 kHz but using shorter continuous wave pulses. While the ocean measured spheres closely matched the model, the test tank measurements showed a marked difference from the model. Changes to the model will be explored to determine if theoretical minimums for pulse length are insufficient for targets of this density or focusing capability.

11:30

2aUW12. Interpretation of the ground wave arrival as a head wave sequence. Jee Woong Choi and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698)

The ground wave arrival (sometimes referred to as a precursor arrival) is interpreted in this work as a train of the head waves, defined here as the head wave sequence. For a Pekeris waveguide, the spectrum of the ground wave arrival is found to be that corresponding to the spectrum of a discrete (time-separated) head wave arrival, evaluated at the modal cutoff frequencies, and the spectrum of a discrete head wave arrival is known to be proportional to the source spectrum divided by frequency. These properties are illustrated for a Pekeris waveguide via Fourier synthesis of a narrow-band complex parabolic equation (PE) field, using the RAM PE code. Although the ground wave arrival in a more realistic waveguide is complicated by several effects, e.g., layering, gradients in the sediment sound speed, some of the above ideas carry over. The introduction of a sound speed gradient in the sediment also produces a key difference in the spectrum of a ground wave arrival (or head wave sequence); this effect is demonstrated through PE simulation, and is also related to data collected in the Yellow Sea. [Research supported by ONR Ocean Acoustics.]

11:45

2aUW13. Frame bulk moduli of air- and water-saturated granular marine sediment models. Masao Kimura (Dept. of Geo-Environ. Technol., Tokai Univ., mkimura@scc.u-tokai.ac.jp)

The frame bulk modulus is important in analyzing the acoustic wave propagation in granular marine sediments. The frame bulk modulus is related to the longitudinal and shear wave velocities. We have reported that the derived values of the frame bulk moduli of air- and water-saturated glass beads are different. In this study, the longitudinal and shear wave velocities in air- and water-saturated beach sands, and in same media in vacuum, are measured. Then the frame bulk moduli are derived from the values of these velocities. The results show that the frame bulk moduli in water-saturated beach sands are about ten times greater than that in air-saturated beach sands, and the frame bulk moduli have dependence on the grain size. These phenomena are investigated using the idea of the gap stiffness between the grains.

TUESDAY AFTERNOON, 16 NOVEMBER 2004 PACIFIC SALON 2, 1:30 TO 4:45 P.M.

Session 2pAA

Architectural Acoustics and Musical Acoustics: Integration of Synthesis Techniques and "Acoustical" Music

K. Anthony Hoover, Chair
Cavanaugh Tocci Associates, Inc., 327F Boston Post Road, Sudbury, Massachusetts 01776

Chair's Introduction—1:30

Invited Papers

1:35

2pAA1. Musical instrument sound levels. K. Anthony Hoover, Andrew Carballeira (Cavanaugh Tocci Assoc. Inc., 327 F Boston Post Rd., Sudbury, MA 01776, thoover@cavtocci.com), Pam Harght, and Sam Ortallono (Berklee College of Music, Boston, MA 02215)

Sound-pressure levels produced by different types of musical instrument can vary considerably. Convincing auralization and appropriate sound-isolation design can be greatly affected by this variation. This paper will present A- and C-weighted statistical sound levels produced by a variety of musical instruments and musical genres, measured at Berklee College of Music. A simple comparison between measured sound isolations using a broadband noise source and a reproduced-music source will also be discussed.

1:55

2pAA2. Conquering space and time—for art's sake. Alex U. Case (Dept. of Music, Univ. of Massachusetts Lowell, 35 Wilder St., Lowell, MA 01854, alex_case@uml.edu)

While sound within a performance space must adhere to the laws of physics governing its propagation, the music synthesized for loudspeaker playback may take some liberties. Recording artists—with the help of the signal processing tools available in the recording studio—often fabricate multiple, physically unrealizable, audio spaces for simultaneous presentation in a single music performance. Additionally, these musicians are free to slow, accelerate, repeat, and reverse time for musical benefit. This paper isolates and demonstrates some of the unusual space- and time-focused synthesis techniques in popular recorded music, evaluating their technical basis and artistic merit.

2:15

2pAA3. Loudspeaker array as a musical composition genre. David Moulton (Moulton Labs., 39 Ames Rd., Groton, MA 01450)

A fixed array of loudspeakers can be used as a proprietary format for electronic music composition, leading to a fixed genre of work. The author will describe his experiences using a 6-channel array of full-range loudspeakers, describing the compositional principles he has employed, including issues of source localization, ambient and reverberant fields, available bandwidth, and sound-pressure levels. Spatial relationships, musical issues and problems, and related concerns will all be discussed.

2:35

2pAA4. Spectrum-based analysis and synthesis of percussion sounds. James W. Beauchamp (School of Music and Dept. of Elec. and Computer Eng., Univ. of Illinois at Urbana–Champaign, Urbana, IL 61801)

From the standpoint of the listener, convenient ways to catagorize percussion sounds are in terms of mode frequency positions, mode frequency spacing, and decay rates. Some sounds, like those of bells, chimes, and other struck bars, have modes which are widely spaced. Other sounds, such as those of timpani, tam-tam, and cymbals, have many more closely spaced modes. Measurements of average modal spacing, time-varying spectral shift, spectral incoherence, attack time, and decay rate are presented for a variety of percussion instrument sounds. Examples of percussion synthesis using this information will be presented.

2:55–3:10 Break

3:10

2pAA5. The influence of synthesized music on the expansion of trombone performance techniques. Thomas Plsek (Berklee College of Music, 1140 Boyslton St., Boston, MA 02215)

Much of the development of synthesized music has been involved with the modeling of traditional acoustic sources such as strings, brass, woodwinds, and percussion. It has, of course, also been concerned with the creation of new, yet unheard, sounds and the modifications and manipulations of existing sources through signal processing. The author has been involved with these processes for many years as a trombone explorer, and has had to interact and integrate wth them. Much was discovered and learned about how trombone performance can be expanded to acoustically simulate these developments in electronic music. This paper demonstrates some of the techniques such as the real-time control of sound source location and the manipulation of timbre. In many instances the acoustical solution proved to be as, or even more, interesting as the synthetic process.

3:30

2pAA6. Measurements of room effects on digital simulations of a concert hall sound field. Patrick R. McAtee and Lily M. Wang (Architectural Eng. Program, Univ. of Nebraska - Lincoln, Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0681, pmcatee@unlnotes.unl.edu)

Digital signal processing is becoming more and more commonplace in the field of architectural acoustics. One way it has been utilized is to simulate the acoustics of a space within another one through convolution of that space's digitized acoustic characteristics with a source signal. In the consumer market, Yamaha pioneered and leads this area, producing a full range of amplifiers with several multichannel auralizations built in. It is not possible to ensure that all users will tune their listening space to allow the auralization to perform its best. This raises questions about how well the auralization can perform in various acoustic environments, and what indicators there might be as to the quality of that performance. A Yamaha system and two sets of speakers were taken to more than a dozen spaces, including offices, classrooms, band rooms, and larger performance spaces. Measurements of each room's acoustical characteristics were recorded and analyzed to see how the acoustic parameters changed when the digital simulation of a concert hall sound field was running. Comparisons of the frequency responses, reverberation times, and clarity indices will be presented. [Work supported by UNL College of Engineering Summer Undergraduate Engineering Research Experience Grant.]

3:50

2pAA7. Design challenges for the integration of virtual acoustics in music practice rooms. Ron Freiheit (Wenger Corp., 555 Park Dr., Owatonna, MN 55050, ron.freiheit@wengercorp.com)

The use of virtual acoustics has provided a new level of practice experience for the musician. By integrating the sound isolation of music practice rooms with the signal processing of an active acoustic system (with time-variant gain before feedback), musicians can now benefit from the experience of practicing in multiple acoustic environments. The variability of the acoustic environment allows the musician to clearly hear their intonation and articulation, which may be difficult to discern in a small practice room. For the effective implementation of virtual acoustics in small spaces (typically 15 square meters) a number of technical issues had to be resolved including sound-field immersion, reducing input level due to proximity of the source to the microphone transducers, and a simplified system control allowing musicians to focus on the art of practice and not the technology. This paper deals with these issues and the subsequent design decisions to address them.

4:10

2pAA8. Composition and performance as outgrowth of synthesis techniques. Richard Boulanger and Greg Thompson (Berklee College of Music, 150 Massachusetts Ave., Boston, MA 02215)

New methods of controlling and interacting with synthesizers are increasingly available to the composer. Advances in input devices enable a large range of gestural parameters to be translated into musically-relevant data, which can then be used in synthesis environments like Csound and Max/MSP. This paper will examine such input devices, including the radio baton, and show how this technology may be combined with existing synthesis techniques to produce exciting new sounds, algorithms, and compositional works.

Contributed Paper

4:30

2pAA9. Violin acoustic radiation synthesis: A source model for direct sound enhancement in musical acoustic environments. Jacob Waxman and Mark Bocko (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, jw001j@mail.rochester.edu)

Within the context of immersive acoustic environments (real or virtual) for purposes of musical performance, it is useful to recreate the acoustic field radiated by real musical instruments. Given the popularity of wave field synthesis for direct sound enhancement and other methods of holophonic sound imaging, source modeling is a desirable development for such musical applications. There exist careful directivity measurements of the sound radiation of the violin in the frequency range from 1 to 5 kHz, over which the far field directivity changes rapidly as a function of frequency [L. M. Wang, "Radiation Mechanisms from Bowed Violins," Ph.D. thesis, Pennsylvania State University, 1999]. Source models based on these measurements using a cylindrical harmonic decomposition are presented. With this preliminary approach to violin acoustic radiation modeling, issues regarding further auralization of this synthesized field are subsequently addressed.

TUESDAY AFTERNOON, 16 NOVEMBER 2004 PACIFIC SALON 1, 1:25 TO 5:10 P.M.

Session 2pAB

Animal Bioacoustics: Marine Mammal Acoustics: Session in Honor of Ron Schusterman II

Colleen Reichmuth Kastak, Chair
Long Marine Laboratory, University of California, Santa Cruz, 100 Shaffer Road, Santa Cruz, California 95060

Chair's Introduction—1:25

Invited Papers

1:30

2pAB1. Temporal integration in a California sea lion and a harbor seal: Estimates of aerial auditory sensitivity as a function of signal duration. Marla M. Holt (UC Santa Cruz Long Marine Lab., 100 Shaffer Rd., Santa Cruz, CA 95060), Brandon L. Southall (NOAA Fisheries Acoust. Program, Silver Springs, MD 20910), David Kastak, Ronald J. Schusterman, and Colleen Reichmuth Kastak (UC Santa Cruz Long Marine Lab., Santa Cruz, CA 95060)

Stimulus durations shorter than some critical value result in elevated signal-detection thresholds due to temporal integration (or temporal summation) properties of the auditory system. These properties are important from a theoretical perspective in terms of the trade-offs of stimulus duration and intensity on sensitivity. From a methodological perspective, temporal integration is important because absolute detection thresholds measured using signal durations shorter than the temporal integration period may underestimate hearing sensitivity. In this study, aerial sound-detection thresholds were estimated at 2500 and 3530 Hz in a California sea lion (*Zalophus californianus*) and a harbor seal (*Phoca vitulina*). Thresholds were measured for each frequency at seven stimulus durations ranging from 100 to 500 ms using behavioral psychophysics in a hemianechoic chamber. In general, thresholds increased as tone duration decreased for durations shorter than approximately 300 ms. For tone durations longer than 300 ms, thresholds were not different from those measured with the longest duration tested. These results suggest temporal integration times of approximately 300 ms for these species, which are consistent with data collected on other mammals. Our findings suggest that tone durations longer than 300 ms should be used in estimating pinniped auditory sensitivity.

1:50

2pAB2. Noise-induced temporary threshold shifts in pinnipeds: Effects of noise energy. David Kastak (UCSC Long Marine Lab., 100 Shaffer Rd., Santa Cruz, CA 95060), Brandon Southall (UCSC Long Marine Lab and NOAA Fisheries Acoust. Program, Silver Spring, MD 20910), Marla Holt, Colleen Reichmuth Kastak, and Ronald Schusterman (UCSC Long Marine Lab., Santa Cruz, CA 95060)

Auditory pure-tone thresholds were obtained in air and in water from three pinnipeds before and immediately after exposure to octave-band noise. Noise exposure durations were 1.5, 12, 22, 25, or 50 min, and noise levels were 65, 80, or 95 dB referenced to each subjects pure-tone threshold. In air and in water, pre- and postnoise thresholds were obtained at the center frequency of the octave band. In water, thresholds were also obtained at a frequency octave higher than the octave-band center frequencies. Maximum threshold shifts for each species were about 15 dB in air and in water. Under all exposure conditions hearing sensitivity recovered within 24 h. In both media, a curvilinear function best predicted the magnitude of threshold shift from noise energy flux density. At TTS magnitudes greater than 10 dB, this function predicts a 2-dB increase in threshold shift for a 1-dB increase in noise energy, agreeing well with data collected from other mammals.

2:10

2pAB3. Objective measures of steady-state auditory evoked potentials in cetaceans. James J. Finneran (US Navy Marine Mammal Program, SPAWARSYSCEN San Diego, Code 2351, 53560 Hull St., San Diego, CA 92152) and Dorian S. Houser (BIOMIMETICA)

Although behavioral methods provide the most direct means of assessing hearing capability, the time, access, and cost required to train individual marine mammal subjects has limited large-scale application of this technique. As an alternative to behavioral testing, auditory evoked potentials (AEPs) have been measured in a number of marine mammal species. AEP measurements using steady-state amplitude modulated tones allow rapid estimates of hearing threshold without extensive subject training. These stimuli result in sinusoidal AEPs, allowing frequency domain measures of AEP amplitude. Unfortunately, AEPs from near-threshold stimuli possess very low signal-to-noise ratios, making discrimination of AEPs from noise difficult. The most common procedures used for AEP threshold estimates in marine mammals have featured a subjective component, requiring an experienced observer to assess the presence or absence of the AEP. In this talk, the use of objective techniques to determine the presence of an AEP will be discussed. The focus of the talk will be on parametric and nonparametric methods within the frequency domain and the importance of AEP phase information. Steady-state AEPs measured in bottlenose dolphins will be used to evaluate the different techniques and compare to behavioral response measures. [Work supported by ONR and SSC San Diego ILIR.]

2:30

2pAB4. Hearing thresholds of a stranded infant Rissos dolphin. Paul E. Nachtigall, Michelle M. Yuen, T. Aran Mooney, and Kristen A. Taylor (Marine Mammal Res. Program, Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734, nachtiga@hawaii.edu)

The underwater hearing of an infant male Rissos dolphin that stranded off the coast of southern Portugal was measured using evoked auditory potentials (AEPs). Hearing thresholds were measured from envelope following responses to amplitude modulated pure tones ranging from 4 to 150 kHz. Acoustic signals were presented within the calibrated rehabilitation pool. Evoked responses were passively gathered from human EEG electrode sensors imbedded within rubber suction cups that were gently attached to the animal with slight suction and conductor gel. One sensor was placed behind the blowhole and the other reference sensor on the back. Tones were presented 1 m directly in front of the animal under water while the animal was held in a neutrally buoyant position. Unlike the previously published audiogram for an older Rissos dolphin, the audiogram, obtained by presenting 18 different frequencies, showed that this very young animal heard tones up to 150 kHz in a manner similar to other odontocetes. These data, collected within 4 days, are the first measure of the hearing of a neonate marine mammal, verify that very young dolphins likely hear better than older animals, and show the value of using AEPs to obtain hearing thresholds of stranded animals undergoing rehabilitation.

2:50

2pAB5. *In vivo* imaging correlated with otoacoustic emissions as a metric for ear disease in seals. D. R. Ketten (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543; Harvard Med. School, Boston, MA 02114, dketten@whoi.edu), W. F. Dolphin (Boston Univ., Boston, MA 02115), R. William (Woods Hole Aquarium, Boston, MA 02110), J. Arruda, and J. O'Malley (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Otoacoustic emissions (OAEs) coupled with auditory brainstem responses (ABR) can help differentiate central, sensorineural, and conductive hearing losses. Pinnipeds with moderate bore canals are good OAE candidates, but OAE utility for assessing marine mammal inner ear health is not known. We examined three juvenile harbour seals (*Phoca vitulina*) with ear disease with computerized tomography (CT), OAE, and ABR. Combining CT, OAE, and ABR allowed simultaneous ear pathology documentation, quantification, and intracanal probe microphone position determination. Hearing was tested bilaterally from 500 Hz to 15 kHz. CT/OAE/ABR results were assessed independently. In two animals, CT showed middle ears occluded with fluid but normal auditory nerve anatomy, suggesting short-term circumscribed infection with no retrograde neuronal loss. OAE found moderately elevated response levels consistent with conductive hearing loss. ABR confirmed normal brainstem functioning. In the third animal, no OAEs or ABRs were obtainable up to 70 dB re 1 μPa, suggesting retrograde loss through brainstem level. CT for this animal showed inner, middle, and external ear occlusions consistent with aggressive, long-term disease. These data show volume and site of auditory pathologies are strongly correlated with OAE results in pinnipeds. [Work supported by Seaver Institute and ONR N00014-93-1-0940; N00014-94-1-1081.]

3:10–3:25 Break

3:25

2pAB6. Noise exposure metrics for auditory and nonauditory damage in aquatic animals. Mardi C. Hastings (Office of Naval Res., Code 341, 800 N. Quincy St., Arlington, VA 22217, mardi_hastings@onr.navy.mil)

The total acoustic energy flux is often used to correlate impacts of different types of sounds having various durations on marine animals. To calculate this metric, one must know both the acoustic pressure and particle velocity. But, in practice, the acoustic pressure is measured usually at just one point and the particle velocity is unknown. The total energy flux is then estimated by assuming the stimulus is an ideal plane wave and calculating the sound exposure level. The inner ears of many aquatic animals, however, respond directly to acoustic particle motion. In addition, because aquatic animals are acoustically coupled to the surrounding water, acoustic particle motion is also critical in estimating sound exposures that may cause damage to tissues outside the auditory system. Thus in noise impact studies, both pressure and local particle velocity must be measured to accurately estimate the total acoustic energy dose received by an aquatic animal. Several case studies taken from the literature will be presented to demonstrate how to estimate acoustic particle velocity from pressure gradient measurements in common aquatic animal testing environments and calculate the total acoustic energy flux.

3:40

2pAB7. Ocean bioacoustics, human generated noise and ocean policy. Michael Stocker (Seaflow, Inc., 1062 Fort Cronchite, Sausalito, CA 94965)

The recent release of the U.S. Commission on Ocean Policy (USCOP) report, just a year on the heels of the Pew Oceans Commission report, has alerted policymakers and the public about the precarious biological health of our seas. While the reports discuss ecosystem based management, ocean bioacoustics is given short treatment in both reports. The ocean is not a visual-dominant environment, rather it is an acoustic environment. Most animals in the sea depend on sound, but we know next to nothing about how living organisms use it. We do know from recent studies that ocean habitats are being seriously compromised by human generated noise in evidence through stranded whales and, more recently, high fish mortality and low productivity in fishing areas due to seismic exploration and civil engineering. Due to the ubiquity of sounds and noises in all of our ocean enterprises, legislating anthropogenic sound promises to be a Byzantine endeavor. This paper examines some of the known challenges to crafting ocean noise policy.

3:55

2pAB8. A potential explanation for marine mammal strandings. L. Crum, S. Kargl, and T. Matula (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

The stranding of whales following naval exercises has been a topic of increasing interest. A recent article [P. D. Jepson *et al.*, "Gas-bubble lesions in stranded cetaceans," Nature (London) **425**, 575–576 (2003)] has suggested that some version of decompression sickness may be involved. Considerable additional evidence has been presented that indicates both acute and chronic effects of these bubbles. We suggested some time ago [L. A. Crum and Y. Mao, "Acoustically enhanced bubble growth at low frequencies and its implications for human diver and marine mammal safety," J. Acoust. Soc. Am. **99**, 2898–2907 (1996)] that the acoustic fields produced by naval sonars could induce bubble formation under certain conditions, particularly if there were a high level of nitrogen supersaturation in these animals. Recent evidence of high supersaturation levels [D. S. Houser, R. Howard, and S. Ridgway, "Can diving-induced tissue nitrogen supersaturation increase the chance of acoustically driven bubble growth in marine mammals?," J. Theor. Biol. **213**, 183–195 (2001)] has stimulated us to perform a series of experiments that suggests that even modest acoustic intensities can trigger bubble formation under supersaturated conditions. Once bubbles are nucleated, these local supersaturation levels then result in bubble growth to macroscopic sizes, and their potential deleterious bioeffects. The results of these preliminary experiments will be presented. [Work supported by APL IR&D funds.]

4:10

2pAB9. High frequency components in cetacean echlocation signals. Thomas G. Muir, Ronald W. Toland, Steven R. Baker (U.S. Naval Postgrad. School, Monterey, CA 93943), Diane J. Blackwood, Lew A. Thompson, and Preston S. Wilson (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713)

The rich literature of high-resolution biosonar capability is sometimes baffling as to attainable sonar resolution. This has led us to measurements in San Diego Bay, on captive research dolphins, utilizing an extremely wide band piezo-composite hydrophone, with a frequency response extending to 2 MHz. The results indicate that cetacean echolocation signals contain frequency components, above ambient noise, that can extend to the neighborhood of 500 kHz. The study was conducted on two bottlenosed dolphins and one beluga whale. Attempts were made to determine if these animals were actually using these high frequency components in echolcation, but they were not completely successful. However, measurements with rho-c matched, acoustic, low pass filter panels, placed between the blindfolded subjects (on a bite bar) and their test targets, showed that as the detection tasks became more difficult, the animals increased the intensity of the high frequency content of their transmissions (to ~500 kHz) and their pulse repetition frequencies also increased. [Work supported by the U.S. Navy ONR.]

4:25

2pAB10. The modulation rate transfer function of a Rissos dolphin, *Grampus griseus*, using the envelope following response.. T. Aran Mooney, Paul E. Nachtigall, and Michelle M. Yuen (Marine Mammal Res. Program, Hawaii Inst. of Marine Biol., 46-007 Lilipuna Rd., Kaneohe, HI 96744, mooneyt@hawaii.edu)

Auditory evoked potentials (AEPs) were used to obtain a modulation rate transfer function (MRTF) of a young, stranded Rissos dolphin (*Grampus griseus*). The animal's neurological response to sound was stimulated by triangle-shaped broadband clicks, $\frac{1}{2}$ ms in duration, played at presentation rates from 100 to 2000 Hz. These stimuli were played from a transducer 1 m in front of the subject, within a calibrated rehabilitation pool. The AEP responses were recorded using human EEG sensors attached by two soft suction-cups, placed just behind the blowhole and in front of the dorsal fin, respectively. By playing the stimuli at different rates, from low to high frequency, it was possible to determine the maximum and most efficient modulation rate, to be used in an AEP audiogram. The MRTF was similar to other cetaceans as it is low-pass in shape. Corner frequencies were somewhat lower than other published marine mammal MRTFs, dropping steeply after 1000 Hz. At this frequency, the evoked potential was highest, 230 nV, compared to noise levels in the tens of nV. Thus, 1000 Hz was determined to be the modulation rate used in determining the animals AEP audiogram.

4:40

2pAB11. Directional sensitivity in bottlenose dolphins: Evoked potential study. Vladimir Popov and Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prosp., 119071 Moscow, Russia)

ABR threshold-versus-azimuth function was measured in two bottlenose dolphins. The measurements were done in a circular pool 6 m in diameter and 40 cm deep. The animal was supported by a stretcher, the head in the center of the pool. The transducer was moved around the animal's head at the radius of 1.2 m. ABRs were recorded using electrodes mounted in suction caps fastened at the head surface. ABR thresholds were measured at azimuth of the sound source of $\pm 90°$ from the midline and frequencies from 8 to 128 kHz. Both acuteness of the directional diagram and the lowest-threshold azimuth value depended on stimulus frequency. Acuteness increased with the frequency increase. The lowest-threshold azimuth was near midline at high frequencies and deviated laterally up to 45–60° with lowering the frequency. The maximum azimuth-dependent thresholds shift (20–35 dB) was at high frequencies (90–128 kHz); at lower frequencies (8–16 kHz), the azimuth-dependent threshold shift decreased down to 10–15 dB. [Work supported by the Russian Foundation for Basic Research.]

4:55

2pAB12. Comparison of behavioral and auditory evoked potential (AEP) audiograms of a false killer whale (*Pseudorca crassidens*). Michelle Yuen, Paul E. Nachtigall, Marlee Breese (Hawaii Inst. of Marine Biol., Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, myuen@hawaii.edu), and Alexander Ya. Supin (Severtsov Inst. of Ecol. and Evolution, 119071 Moscow, Russia)

Behavioral and auditory evoked potential (AEP) audiograms of a false killer whale were measured using the same subject and experimental conditions from 2001 to 2004. The objective was to compare and assess the validity of auditory thresholds collected by psychometric and electrophysiological techniques. Behavioral audiograms used 3-s pure-tone stimuli from 4 to 45 kHz. AEP audiograms used 20-ms sinusoidally amplitude-modulated (SAM) tone bursts from 4 to 45 kHz. Electrophysiological responses were received through gold disk electrodes mounted in rubber suction cups placed on the animals dorsal surface. Psychometric data were reliable and repeatable with the region of best sensitivity for the behavioral audiograms between 16 and 24 kHz, and with peak sensitivity at 20 kHz. The AEP measures produced thresholds that were consistent over time, with ranges of best sensitivity from 16 to 22.5 kHz and peak sensitivity at 22.5 kHz. Behavioral thresholds were lower than AEP thresholds. Signal type and duration differences may explain the discrepancies between thresholds obtained under the two conditions. These data indicated that behavioral and AEP techniques can be used successfully and interchangeably to measure cetacean hearing sensitivity. [Work supported by ONR.]

TUESDAY AFTERNOON, 16 NOVEMBER 2004 — SHEFFIELD ROOM, 1:30 TO 3:30 P.M.

Session 2pAO

Acoustical Oceanography and Underwater Acoustics: Acoustic Sensing of Internal Waves II

James F. Lynch, Chair

Woods Hole Oceanographic Institute, 203 Bigelow Building, Woods Hole, Massachusetts 02543

Contributed Papers

1:30

2pAO1. Fluctuations of high-frequency sound field in shallow water in the presence of internal waves. Mohsen Badiey (Univ. of Delaware, Newark, DE 19716), Boris Katsnelson, and Serguey Pereselkov (Voronezh Univ., Voronezh, 394006 Russia)

The structure of space-time fluctuations of a high-frequency (several kHz) sound field in a shallow water waveguide is considered. The sound field fluctuations due to internal and surface waves show different temporal variability. The perturbation theory within the framework of ray description is applied in order to derive some analytical estimation of fluctuations of arrival time and vertical angle distribution of high frequency broadband acoustics signals. The parabolic equation is used for numerical modeling of space-time fluctuations and vertical sound field intensity distribution. The experimental measurements of ocean dynamics from recent acoustic experiments are used as input data in numerical simulation of sound field fluctuations. [Work was supported by ONR and CRDF].

1:45

2pAO2. Internal- and surface-wave-induced fluctuations and frequency spreading in shallow water acoustic propagation over short ranges. Stephen D. Lynch, Gerald L. D'Spain, and Eric Terrill (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093)

During a 2001 ocean acoustics experiment approximately 40 miles west of San Diego, a moored, underwater source at 30-m depth transmitted a set of narrow-band tones in the 100–1000-Hz band to a 32-element, 992-m-long bottom hydrophone array deployed from R/P FLIP. Also deployed about FLIP were multiple, fixed vertical thermistor strings with ambient pressure sensors, and CTD casts. These data provide observations of the effects of a fluctuating ocean environment on acoustics in shallow water (180 m) over short ranges (2.5 km). A series of very long fast Fourier transforms of the received signals reveal time dependence in the frequency spreading about the transmitted frequency. Time series of spatial and temporal coherence estimates show temporal variations in the relatedness of the received signals. The environmental data are used along with physics-based models of ocean surface and internal waves to predict the signal fluctuation and frequency spreading characteristics. Results of

comparison of acoustic observations with oceanographic-data/model-based predictions under various assumptions will be presented. [Work supported by ONR, Code 321(US).]

2:00

2pAO3. Vertical line array beamforming of signal and noise in shallow-water regions. Tim Duda, James Lynch (AOPE Dept., M.S. 11, Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Phil Abbot (Ocean Acoust. Services and Instrumentation Systems, Inc., Lexington, MA 02421), and Ruey-Chang Wei (Natl. Sun Yat-sen Univ., Kaohsiung City, Taiwan)

The familiar mode-stripping and coupling processes are prevalent in shallow water, shaping the vertical directional content of noise from distributed sources and of signals from discrete sources. Directional patterns can help in the determination of the physical processes, including internal waves, controlling acoustic behavior in a given region. VLA beamformer output from the New England PRIMER and the South China Sea ASIAEX studies are shown. Beamformed output from moored source signals and of noise vary rapidly for ASIAEX, less so for PRIMER. Fluctuations of signal and noise directional content have limited coherence. Correlated fluctuations of these will not alter the signal/noise ratio for a given beam, whereas uncorrelated fluctuations will. Such correlation depends upon the distance between the mode-coupling structure and the receiver. Predictions of pattern variability are complex because many environmental degrees of freedom are involved. For example, mode coupling of distant sources will spread the pattern from near horizontal (low mode), but not always for nearby sources; mode coupling of noise will energize low angle beams, although most effectively near noise sources. Implications of waveguide characterization based on beam energy variability will be discussed and compared with direct environmental waveguide measurements and with other signal properties. [Work supported by ONR.]

2:15

2pAO4. Acoustic mode coupling effects from propagation through nonlinear internal waves. Laurel K. Reilly-Raska (Rensselaer Polytech. Inst., Troy, NY 12180, reilll@rpi.edu), James F. Lynch, John A. Colosi (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), and William L. Siegmann (Rensselaer Polytech. Inst., Troy, NY 12180)

Existing literature demonstrates that costal and shelf break regions frequently contain significant nonlinear internal wave (NIW) activity and fronts. Our previous work, which incorporated stochastic modeling techniques in conjunction with azimuthally varying NIWs in an adiabatic setting, showed significant variations in intensity. However, NIWs can strongly affect energy coupling between acoustic modes. We seek to understand the evolution of coupled mode propagation through a field of randomly varying NIWs. We will examine the effects of coupling due to propagation through the intersection of two or more NIWs crossing at oblique angles. We use a fully 3D parabolic equation method and other tools to examine the ensemble of effects. [Work supported by ONR.]

2:30

2pAO5. The relative effects of sound-speed variability in the water column versus the seabed on normal-mode propagation in shallow water. George V. Frisk (Dept. of Ocean Eng., Florida Atlantic Univ., Dania Beach, FL 33004, gfrisk@seatech.fau.edu), Cynthia J. Sellers (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), and Luiz L. Souza (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Normal-mode propagation in shallow water is influenced by variations in the acoustic properties of both the water column and the seabed. Specifically, the sound speed in the water may fluctuate due to the passage of an internal wave, while the sound speed in the bottom may change due to variable geological features. Assessing the relative effects of the water column versus the seabed on the characteristics of modal propagation is critical to the understanding of both the forward and inverse problems in shallow-water acoustics. In this paper, perturbation theory is combined with a Pekeris waveguide model to provide an analytic formalism for evaluating the delicate interplay between the water column and the seabed in shallow water propagation. In particular, the relative contributions of sound-speed fluctuations in the water and the bottom to variations in the modal eigenvalues are determined. The results are affected by both the strengths of the fluctuations and the magnitudes of the background modal eigenfunctions in the water and the seabed. The complexity of the problem is illustrated with synthetic and experimental data. [Work supported by ONR.]

2:45

2pAO6. Coupled mode theory for sound propagation through random internal wave fields. John Colosi (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

In the late 1970's Dozier and Tappert were able to formulate a theory for the range evolution of deep-water acoustic normal-mode energies utilizing (1) a perturbation method up to second-order mode transitions; (2) a phase-randomized internal wave field; (3) the Markov approximation; and (4) zero cross-mode coherence. Since 1989, vertical receiver array observations of broadband signals at 75 and 250 Hz have shown distinct time-resolved wavefronts at multimegameter ranges, thereby invalidating the zero cross-mode coherence assumption. In this work, the theory is modified to remove the zero cross-mode coherence assumption, thereby yielding range evolution equations for both the mode energies and the cross-mode coherences. Using a Lorentzian internal wave spectrum which fits the Garrett–Munk spectrum very well, analytic expressions for the coefficients of the evolution equations are obtained. Predictions using the theory agree with Monte Carlo numerical simulations at 75 Hz at roughly the 1–2-dB level. It is found that cross-mode coherences decay very slowly with range, and that the modal energy equations nearly decouple from the cross-mode coherence equations. The theory may offer a means to acoustically measure internal wave spectra.

3:00

2pAO7. Random horizontal refraction at long-range sound propagation in the ocean. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA/Environ. Technol. Lab., 325 Broadway, Boulder, CO 80305), Alexander G. Voronovich, and Valery U. Zavorotny (NOAA/Environ. Technol. Lab., Boulder, CO 80305)

Internal gravity waves (IWs) are a major source of sound speed fluctuations in the ocean. For refracted waves in deep water, 3-D and 4-D acoustic effects are caused by cross-range gradients and time dependence of the sound speed. In this paper, IW-induced 3-D and 4-D acoustic effects are studied assuming a continuum of random IWs with the Garrett–Munk spectrum. Fluctuations in azimuthal arrival angles and frequency of tonal signals as well as travel time biases due to horizontal refraction are quantified with a theory which allows one to calculate directly statistical moments of acoustic observables as a weighted integral of an appropriate statistical moment of environmental perturbations. Predictions of the theory are compared to results of Monte Carlo simulations of 3-D and 4-D acoustic effects in the ray approximation. Magnitude of the effects associated with random horizontal refraction at megameter ranges is found to be sensitive to a choice between forms of the Garrett–Munk spectrum which are often viewed as equivalent. Implications of this observation on acoustic characterization of IW fields in deep water are discussed. [Work supported by ONR.]

3:15

2pAO8. Space-frequency sound field distribution in the neighborhood of a temperature front. Boris Katsnelson, Alexander Tshoidze (Voronezh Univ., 1, Universitetskaya sq., Voronezh, 394006, Russia), James Lynch, and Ta-Wei Wang (Woods Hole Oceanogr. Inst, Woods Hole, MA 02543)

In this paper, the properties of the sound field in the area close to a temperature front (TF) in shallow water are studied. It is shown that for a realistic temperature front providing the variation of the sound speed profile, interesting effects due to horizontal refraction can be observed if the

source/receiver are placed at distances of several hundred meters (up to 1 km) from the TF. It is shown that strong interferences in the sound field take place both in the range (horizontal plane) and frequency domains (resonance-like spectra of received signals). Effects of mode separation due to horizontal refraction can be experimentally observed (in principle) as well as the aforementioned intensity gain. These effects are similar to observed horizontal refraction phenomena in the presence in internal solitons. Numerical modeling is carried out, and an experimental setup is discussed for typical conditions in shallow water areas. [Work supported by RFBR (03-05-64568) and CRDF (REC 010-0)].

TUESDAY AFTERNOON, 16 NOVEMBER 2004 ROYAL PALM SALON 6, 1:25 TO 4:15 P.M.

Session 2pEA

Engineering Acoustics, Signal Processing in Acoustics, Psychological and Physiological Acoustics and Committee on Standards: Hearing Aids

Gary W. Elko, Cochair
Avaya Labs, 233 Mt. Airy Road, Basking Ridge, New Jersey 07920

James G. Ryan, Cochair
Gennum Corporation, 232 Herzberg Road, Kanata, Ontario K2K 2A1, Canada

Chair's Introduction—1:25

Invited Papers

1:30

2pEA1. Integrated circuit technology in hearing aids. Steve Armstrong (Gennum Corp., 970 Fraser Dr., Burlington, ON L7L 5P5, Canada)

In recent years, digital technology has enabled the provision of many new hearing aid features such as noise reduction, feedback suppression and environment classification. As silicon technology evolves, more and more features will be possible within the existing package size and power budget. This paper provides an overview of integrated circuit technology and its application to hearing aid design. The architecture of a digital hearing aid circuit is described in light of the system constraints. The talk will conclude with a brief discussion of expected technological advances and their impact on hearing aid design.

2:00

2pEA2. Noise reduction for digital hearing aids. Volker Hohmann and Birger Kollmeier (Universitaet Oldenburg, Medizinische Physik, D-26111 Oldenburg, Germany)

The normal-hearing system extracts monaural and binaural features from the signals at the left and right ears in order to separate and classify sound sources. Robustness of source extraction is achieved by exploiting redundancies in the source signals (auditory scene analysis). ASA is closely related to the "Cocktail Party Effect," i.e., the ability of normal-hearing listeners to perceive speech in adverse conditions at low signal-to-noise ratios. Hearing-impaired people show a reduced ability to understand speech in noisy environments, stressing the necessity to incorporate noise reduction schemes into hearing aids. Several algorithms for monaural, binaural and multichannel noise reduction have been proposed, which aim at increasing speech intelligibility in adverse conditions. A summary of recent algorithms including directional microphones, beamformers, monaural noise reduction and perceptual model-based binaural schemes will be given. In practice, these schemes were shown to be much less efficient than the normal-hearing system in acoustically complex environments characterized by diffuse noise and reverberation. One reason might be that redundancies in the source signals exploited by the hearing system are not used so far by noise reduction algorithms. Novel multidimensional statistical filtering algorithms are introduced that might fill this gap in the future. [Work supported by BMBF 01EZ0212.]

2:30

2pEA3. A metric for evaluating speech intelligibility and quality in hearing aids. James M. Kates (GN ReSound, 3215 Marine St., Rm. W161, Boulder, CO 80309, jkates@gnresound.dk) and Kathryn H. Arehart (Univ. of Colorado, Boulder, CO 80309)

Noise and distortion reduce speech intelligibility and quality in hearing aids, but there are no metrics that encompass both noise and distortion. This presentation introduces new intelligibility and sound-quality calculation procedures based on the Speech Intelligibility Index [ANSI S3.5-1997]. The SII involves measuring the signal-to-noise ratio (SNR) in separate frequency bands, modifying the estimated noise levels to include auditory masking, and computing a weighted sum across frequency of the modified SNR values. In the new procedure, the estimated SNR is replaced by a signal-to-distortion ratio (SDR) computed from the coherence between the input and output signals of the system under test. The SDR replaces the system noise estimate with the combined noise and nonlinear distortion. For the procedure, the signal is divided into three regions comprising the low-, mid-, and high-level signal segments. The SII is then calculated for each region using the corresponding coherence. The three coherence SII values are then combined to predict the intelligibility and the sound quality for the device under test. The coherence SII procedure is shown to be accurate for both normal-hearing and hearing-impaired listeners for additive noise, peak-clipping distortion, and center-clipping distortion. [Work supported by GN ReSound (JMK) and the Whitaker Foundation (KHA).]

3:00–3:15 Break

3:15

2pEA4. Hearing aids: real world outcomes of the engineering feats. Ruth Bentler (Dept. of Speech Pathol. and Audiol., Univ. of Iowa, 250 Hawkins Dr., Iowa City, IA 52242, ruth-bentler@uiowa.edu)

It is often the case that software and hardware designs result in profound changes in the way hearing aids function. However, designing engineers must rely on feedback from others' disciplines as to how these devices are evaluated and used in the "real world." Directional microphone systems, noise reduction algorithms, and feedback cancellation schemes are all designed to eliminate unwanted interference. How those functions work in the real world, either independently or concurrently, will be explored.

Contributed Papers

3:45

2pEA5. Audio scene analysis for hearing aids. Marie Roch, Tong Huang (Dept. of Computer Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720), and Richard R. Hurtig (The Univ. of Iowa, Iowa City, IA 52242)

It is well known that simple amplification cannot help many hearing-impaired listeners, and numerous signal enhancement algorithms have been proposed for digital hearing aids. In many cases, algorithms are most effective in specific environmental or source conditions. If one can properly detect components of the auditory scene, it is possible to dynamically apply enhancement algorithms which are appropriate for a given situation. This work illustrates this principle by describing a cohort detection scheme which serves as a control system for a frequency-domain compression algorithm. The compression algorithm preserves formant ratios and thus enhances speech understanding for individuals with severe sensorineural hearing loss in the 2–3-kHz range. By detecting speakers from broad cohorts (e.g., male, female), it is possible to adjust the compression ratio dynamically based upon characteristics of the auditory scene, resulting in a more appropriate enhancement than one based upon a static ratio. Cohort decisions are derived from the likelihood scores of a Gaussian mixture model classifier.

4:00

2pEA6. A new psychoacoustic fitting algorithm for digital hearing aids. Jeong-Guon Ih, Kyung-Hoon Park (Dept. of Mech. Eng., KAIST, Sci. Town, Taejon 305-701, Korea), and Dong-Gu Yoo (Sammi Sound Tech., Cheongju, Chungbuk 360-813, Korea)

Existing fitting methods, which employ a pure tone stimulus, such as the Fig6 method or the POGO2 method, yield the same target gains when individual hearing thresholds are identical. Therefore, the loudness perception of an individual is hardly considered. Also, the fitting procedure becomes an empirical one, which is very time-consuming for detailed adjustment. Fitting methods using fractional octave-band tone stimulus often result in excessive gains at low frequencies and too many measurements for loudness level setting are required. In this study, subjects with normal hearing are tested and the loudness perception to a certain level of band-limited white noise at the modified 14 critical bands is classified by five categories. A standard database, as the target value, is constructed by statistically processing the results. The same procedures are applied to a hearing impaired patient and the response data are used for estimating individual hearing characteristics. Acquired hearing loss data are compared with the standard database of normal hearing and then the target gains for five loudness categories are obtained for compensation. The proposed fitting method was compared with the existing methods for some patients and the results suggest that the auditory performance is better than the existing algorithms or empirical readjusting fitting.

Session 2pED

Education in Acoustics: Take 5's

Uwe J. Hansen, Chair
Department of Physics, Indiana State University, Terre Haute, Indiana 47809

Do you have a novel demonstration, a new laboratory experiment, a favorite video, a recorded sound example, or a new idea for teaching acoustics? Share it with your colleagues. A sign-up board will be provided for scheduling presentations. No abstracts are printed. Presenters are encouraged to have handouts for distribution. Multiple presenations are acceptable (not consecutively). Presentations are limited to 5 minutes. Keep them short! Keep them fun!

TUESDAY AFTERNOON, 16 NOVEMBER 2004 ROYAL PALM SALON 1, 1:30 TO 4:00 P.M.

Session 2pNS

Noise: Propulsion/Airframe Aeroacoustics II

Kevin P. Shepherd, Chair
NASA Langley Research Center, Hampton, Virginia 23681

Chair's Introduction—1:30

Invited Papers

1:35

2pNS1. Installed jet noise prediction with Jet3D. Craig Hunter (NASA Langley, MS 499, Hampton, VA 23681, craig.hunter@nasa.gov)

Much of the aircraft noise reduction research in recent decades has been concentrated at the component level in understanding the physics of noise generation, formulating noise prediction methods, and developing noise reduction strategies. A new strategy in NASA's noise reduction program is a focus on the noise effects specifically attributable to installation. This focus will also extend to developing noise reduction strategies that take advantage of installation effects. An essential requirement for an installed jet noise prediction method is that it must be able to predict noise from complex three-dimensional flows. Having this physics based capability is also desirable in order to develop a flexible noise prediction method applicable to the investigation of advanced concepts and revolutionary configurations. A jet noise prediction tool satisfying these objectives is currently under development at NASA Langley Research Center. The Jet3D methodology is based on Lighthills Acoustic Analogy and uses Reynolds-averaged Navier–Stokes (RANS) computational fluid dynamics (CFD) simulations from the PAB3D flow solver, with temperature-corrected two-equation turbulence closure and anisotropic Reynolds stress modeling. This talk describes development of the Jet3D noise prediction method and its application to installed jet configurations.

1:55

2pNS2. Engine configurations for the Silent Aircraft. Cesare Hall (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, UK, cah1003@cam.ac.uk)

The Silent Aircraft Initiative is a Cambridge-MIT Institute research project aimed at reducing aircraft noise to the point where it would be unnoticeable in the urban areas around airports. This talk investigates viable engine and installation options that can meet the Silent Aircraft noise requirements. To reduce the transmitted jet noise sufficiently, it is found that a very large exhaust nozzle area is required. If this is achieved with a conventional turbofan it leads to a very high engine diameter and low fan pressure ratio. The large diameter results in greater engine weight and unacceptably high installation drag at cruise. The low fan pressure ratio makes the engine more susceptible to operability problems such as fan stall and vibration, which lead to a further increase in weight. This presentation demonstrates how these problems can be mitigated with a propulsion system that has a variable cycle combined with a novel installation that is embedded into the airframe. Several suitable engine concepts are considered in detail and each configuration is assessed in terms of its relative performance, weight, noise emission, and operability.

2:15

2pNS3. Acoustic shielding of engine noise by the Silent Aircraft airframe. Anurag Agarwal (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, UK)

The "Silent Aircraft Initiative" is a project funded by Cambridge-MIT Institute (CMI). Its aim is to discover ways of reducing aircraft noise to the point where it would be virtually imperceptible to people outside the airport perimeter in a typical urban environment. The present design of the Silent Aircraft is in the form of a flying wing with a large wing planform and a propulsion system that is embedded in the rear of the airframe. Thus, a large part of the forward-propagating noise from the intake duct of the engines is expected to be shielded from observers on the ground by the large upper surface of the wing. In this talk, the use of boundary element methods to predict the attenuation of sound due to shielding by the wing is discussed. The reciprocity principle is invoked to study the noise contributions from several engine-intake locations at given observer locations on the ground. This procedure yields the optimum location of the engine intake for maximum noise shielding. Finally, a technique to incorporate the effect of background mean flows on sound propagation is briefly discussed.

2:35–2:50 Break

2:50

2pNS4. Innovative aircraft design for step changes in noise reduction. Zoltan S. Spakovszky (MIT Gas Turbine Lab., 77 Massachusetts Ave, Cambridge, MA 02139, zolti@mit.edu)

Low-noise integrated propulsion system concepts are proposed for functionally silent aircraft with the goal to reduce airframe and propulsion system noise by as much as 30 dB. Silent in this context means sufficiently quiet that the aircraft noise is less than that of the background noise in a typical well-populated environment. The theme of the technical approach for this multi-disciplinary problem is based on a systems view rather than an individual component view of the airframe and interacting propulsion system. Simple analytical modeling and existing semi-empirical noise prediction methods and scaling laws are used to predict the acoustic signature of low-noise airframe and propulsion system concepts envisioned for a functionally silent aircraft. The design study and acoustic analysis is based on an aerodynamically clean blended-wing-body-type airframe configuration. A distributed propulsion system is proposed to facilitate airframe boundary layer ingestion and to take advantage of shielding effects by embedding engines in the airframe. Ultra-high bypass ratio engines are necessary to reduce jet noise and multiple small engines or a multi-fan engine system is envisioned to enable integrating the propulsion system with the airframe. Results from the noise assessment studies are discussed and preliminary design implications for a functionally silent aircraft are given.

3:10

2pNS5. Systems analysis of advanced quiet aircraft. Erik Olson, Geoffrey Hill, Sherilyn Brown, Karl Geiselhart (NASA Langley Res. Ctr., MS 248, NASA-Langley Res. Ctr., Hampton, VA 23681-2199, erik.d.olson@nasa.gov), and Cecile Burg (Georgia Inst. of Technol., Hampton, VA 23681-2199)

To achieve future goals for the reduction of aircraft community noise, it may be necessary to use unconventional engine and wing arrangements. Prediction of the resulting noise benefits cannot be made without taking into account the effect of the configuration on the whole aircraft system, including the impact on the aerodynamics, weights, and systems integration. Systems studies have been conducted that quantify the impact of various configurations on both the performance and noise of the aircraft. In this paper, the results for several systems studies are presented. In one study, a large matrix was assembled for candidate configuration options for a blended wing–body (BWB) aircraft–including engine, inlet, and nozzle types and placement options—and ranked based on prioritization between noise benefits and performance penalties. In another study, a pair of concepts was developed and evaluated with the goal of simultaneously reducing the noise produced and reducing or eliminating the production of harmful chemical emissions. Finally, a BWB configuration was studied to determine the sensitivity of the overall noise to the levels of the various engine and airframe sources, quantify the airframe shielding benefits using experimental data, and explore the effects of wing trailing-edge extensions on the shielding.

Contributed Papers

3:30

2pNS6. Observations of aeroacoustic modifications of a supersonic jet by using a spherical reflector. Kunisato Seto and Md. Tawhidul Khan Islam (Dept. of Mech. Eng., Saga Univ., 1 Honjo-machi, Saga, 840-8502, Japan)

Jet screech was eliminated by placing a spherical reflector at the nozzle exit of an underexpanded supersonic jet. The placement of the reflector minimized the sound pressure and this muted sound did not excite the unstable disturbance at the exit of nozzle and the loop of feedback mechanism disappeared, thus the generation of jet screech was cancelled. The proposed method was very effective in reducing overall sound pressure in the upstream region of the nozzle exit. Therefore, it could be a promising countermeasure against acoustic fatigue to protect the fuselage of an aircraft by incorporating it into a streamline form. The flow characteristic of the jet for the proposed system was observed by the Schlieren apparatus with a high speed video camera. A slit path was cut to the reflector for visualizing the jet properly. The shock structure of the jet was slightly modified and became more stable. [The research was supported by a grant from JSPS of the government of Japan.]

3:45

2pNS7. Control of supersonic jet noise with a spherical reflector. Md. Tawhidul Khan Islam and Kunisato Seto (Dept. of Mech. Eng., Saga Univ., 1 Honjo-machi, Saga, 840-8502, Japan)

Jet screech is considered to be generated from a feedback cycle in an underexpanded supersonic jet. Experiments were carried out to control the screech tone by placing a spherical reflector at the nozzle exit. The reflector controlled the location of the sound image-source and minimized the sound pressure at the nozzle exit. The weak sound could not excite the unstable disturbances at the exit and the feedback mechanism was cancelled and finally screech tone was eliminated. The new technique sup-

pressed not only the screech tones but also the broadband noise and reduced the overall sound pressure of the jet. The performance of the proposed technique was checked with different sizes of reflectors by placing the reflectors at different upstream positions of the nozzle exit for different pressure ratios and the result was very good. [The research was supported by a grant from JSPS of the government of Japan.]

TUESDAY AFTERNOON, 16 NOVEMBER 2004 PACIFIC SALONS 4 & 5, 1:00 TO 5:15 P.M.

Session 2pPA

Physical Acoustics and Biomedical Ultrasound/Bioresponse to Vibration: Sonoluminescence, Sonochemistry and Sonofusion

Thomas J. Matula, Chair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, Washington 98105-6698

Invited Papers

1:00

2pPA1. What are the limits of energy focusing in sonoluminescence? Seth Putterman, C. Camara, B. Kappus, C. K. Su, E. Kirilov, and A. Chakravarty (Phys. Dept., UCLA, Los Angeles, CA 90095)

Sonoluminescence (“SL”) is an amazing marker for the extraordinary degree of acoustic energy focusing achieved in a cavitating bubble. Local energy dissipation exceeds Kirchhoff’s law by 10^{15} and the ambient acoustic energy density concentrates by 12 orders of magnitude to create picosecond flashes of broadband ultraviolet light. For single bubbles driven at 30 kHz, SL is nature’s smallest blackbody. Therefore the bubble’s interior is such a dense plasma that the photon-matter mean-free path is shorter than the wavelength of light. Excitation of a vertical column of fluid ($\sim$50 Hz), so as to create a water hammer, upscales flash energy by a factor over one million, achieving peak powers approach 1 W. At 1 MHz the spectrum resembles Bremsstrahlung from a transparant plasma with a temperature $\sim$1 MK. At 10 MHz the collapsed size of the SL bubble approaches 10 nm, which raises the possibility that the SL parameter space may extend to the domain of quantum mechanics. At 30 MHz experiments are under way to excite sonoluminescence with sound fields in excess of 3000 atm. The strongest cavitation collapses may be realized with Greenspan’s ultrasonic resonators that reach fields in excess of 20 atm without cavitating. When bubbles are seeded with an external laser a massive cavitation event ensues.

1:30

2pPA2. Additional evidence of nuclear emissions during acoustic cavitation. Rusi Taleyarkhan, Colin West, JaeSeon Cho, and Richard Lahey, Jr. (Purdue Univ., W. Lafayette, IN 47907-1290, rusi@purdue.edu)

Time spectra of neutron and sonoluminescence emissions were measured in cavitation experiments with chilled deuterated acetone. Statistically significant neutron and gamma ray emissions (with more than 25 standard deviation accuracy) were measured with a calibrated liquid-scintillation detector. The neutron and sonoluminescence emissions were found to be time correlated over the time of significant bubble cluster dynamics. The neutron emission energy was at and below 2.45 MeV and the neutron emission rate was up to $\sim$400 000 n/s. Measurements of tritium production were also performed and these data implied a neutron emission rate due to D-D fusion which agreed with what was measured. In contrast, control experiments using normal acetone did not result in statistically significant tritium activity, or neutron or gamma ray emissions. The talk will highlight significant details of the acoustic chamber design, characterization and qualification along with the nuclear data obtained.

2:00

2pPA3. The temperatures of single-bubble sonoluminescence. Kenneth S. Suslick and David J. Flannigan (School of Chemical Sci., Univ. of Illinois at Urbana-Champaign, 600 S. Mathews Ave., Urbana, IL 61801)

We observe extraordinarily intense single-bubble sonoluminescence (SBSL) from concentrated sulfuric acid (H2SO4) containing noble gases. Strong atomic Ar emission and extensive vibronic progressions from sulfur monoxide (SO) are also present in the SBSL spectra. The Ar atom excited states observed are too high in energy to be thermally populated and must be excited by high energy particle impact, consistent with Ar atom SBSL from an emissive shell surrounding an optically opaque plasma core, just as in a star or thermonuclear explosion. From relative intensities of Ar lines, we find that the observed effective emission temperature during SBSL is 15 200$\pm$1900 K. SBSL emission temperatures can be systematically controlled over the range from $\sim$1500 to $\sim$20 000 K by changing the applied acoustic pressure or the thermal conductivity of the dissolved gas.

2:30

2pPA4. High-frequency, high-power ultrasonic chemical reactors. Michael Hoffmann, Timothy Lesko, and Agustin Colussi (California Inst. of Technol., Eng. & Appl. Sci. 138-78 CIT, Pasadena, CA 91125)

A novel high-frequency (612 kHz), high-power (4 kW), pilot-plant scale sonochemical reactor was developed and used to study the degradation of organic chemical pollutants in aqueous solutions. The degradation rates of trichloroethylene, dichloromethane, and phenol were found to exceed those of similar frequency, small-scale bench reactors by factors ranging from 2.5 to 7. In addition, there

is linear dependence between the observed sonolytic rate constants and the applied power density. The addition of ozone during sonication did not affect the first-order degradation rate constants for phenol degradation compared to the linear combination of sonication and ozonation. However, enhancement in the degradation rates of the total organic carbon (TOC) were observed. The enhanced reactivity of sonolysis coupled with ozonation is due to the sonolytic formation of hydrogen peroxide in water, which in turn reacts with ozone to form a highly reactive ozonide intermediate, dihydrogen trioxide, that reacts in a similar fashion to hydroxyl radical. However, its lifetime in aqueous solution is found to be substantially longer, and thus it is more likely to react with refractory organic compound fragments.

3:00

2pPA5. The effect of surfactants on inertial cavitation activity and sonoluminescence intensity in aqueous solutions. Muthupandian Ashokkumar, Judy Lee, Sandra Kentish, Franz Grieser (School of Chemistry, Univ. of Melbourne, VIC 3010, Australia), and Thomas J. Matula (Univ. of Washington, Seattle, Washington 98105)

The understanding and influencing of bubble clusters in an ultrasonic filed is important for many applications. We have investigated the various concentration-dependent effects of sodium dodecyl sulfate (SDS), an anionic surfactant, in aqueous solutions on bubble coalescence, stability of bubble nuclei and rate of bubble growth in an ultrasonic field. For example, the addition of low concentrations of SDS has been found [J. Phys. Chem. B, **101**, 10845 1997] to enhance the intensity of multibubble sonoluminescence (MBSL). Further investigations on the effect of SDS on MBSL intensity have revealed that the SL enhancement observed is highly dependent on the ultrasound pulse length. We have observed that there exists a critical (ultrasound) pulse length below which the addition of SDS in fact lowers the MBSL intensity relative to that observed from water. The inertial cavitation activity has been experimentally measured under similar conditions to that of the SL experiments and has been found to correlate with MBSL results. Both MBSL and inertial cavitation activity results suggest that the presence of SDS at higher concentrations could also enhance the rate of bubble growth by rectified diffusion. This presentation will focus on how specific surfactants influence bubble cluster dynamics and bubble coalescence.

3:30–3:45 Break

3:45

2pPA6. Acoustically assisted cleaning in microelectronics below 65 nanometers (nm): An urgent need to understand mechanisms and effects. Gary Ferrell (SEZ America, Inc., 2632 Bayshore Pkwy., Mountain View, CA 94043)

1000-kHz ultrasonics (megasonics) has been an important technology in controlling killer defects in modern microelectronics. In the construction of, say, a microprocessor, perhaps 10 percent of the 500 process steps may involve acoustically aided cleaning. Feature sizes are currently at 65 nm and will be at 32 nm by 2009. In our laboratory, we have been utilizing multiple-bubble sonoluminescence (MBSL) as a probe into cavitation processes in the hopes of learning to control bubble dynamics and growth. Strong evidence has been found which links MBSL with the removal of 100 nm and larger particles on single-crystal silicon substrates during wafer cleaning. Improved understanding of the bubble interaction with the surface features must be developed in order to maximize cleaning and eliminate damage to nanoscale features. Surface tension modification, overpressure, and nucleation control are some of the experimental controls being developed. A novel nanoscale optical damage probe is in development to further aid in understanding the forces impinging on surface features.

Contributed Papers

4:15

2pPA7. Temporal evolution of sigle bubble sonoluminescence spectra: From bubble creation to stabilization. Jeremy Young (NASA John Glynn Res. Ctr., 21000 Brookpark Rd., Cleveland, OH 44135) and Thomas Matula (Univ. of Washington, Seattle, WA 98105)

Single bubble sonoluminescence (SBSL) is generated by creating and levitating a bubble at the pressure antinode of a standing wave. Previously, it was determined that the SBSL total light intensity evolves during the stabilization process [T. J. Matula and L. A. Crum, Phys. Rev. Lett. **80**, 865–868 (1998)]. The evolution process was used to confirm a theoretical hypothesis that chemical diffusion effects were important for stable SBSL. In this talk we report how the emission spectrum evolves over time, from bubble creation to stabilization. SBSL was generated in a 100-ml spherical cell filled with degassed (air) filtered water. Boiling from a nichrome wire created bubbles which were then forced to the pressure antinode. Spectra were recorded with a multichannel PMT (Burle Industries) using 16 color channels in a 4×4 grid with 20-nm bandwidth interference filters covering the range from 250 to 625 nm. Continuous recording of the spectra (with a 192-ms integration time) was performed until the bubble stabilized. Spatial, crosstalk, filter, and tube response calibrations were performed. Temporal evolution measurements showed evidence of an early-stage emission band near 300 nm, suggesting that hydroxyl emission becomes swamped by the continuum during stabilization. [Research was funded by NASA.]

4:30

2pPA8. Stable multibubble sonoluminescence. Larry R. Greenwood, Gerald J. Posakony, Leonard J. Bond, Morris S. Good (Pacific Northwest Natl. Lab., P.O. Box 999, Richland, WA 99354), Salahuddin Ahmed, Michael D. Wojcik, Warren W. Harper, and Marino Morra (Pacific Northwest Natl. Lab., Richland, WA 99354)

A multibubble standing wave pattern can be generated from an acoustic wave reflected from a flat surface. By adding a second transducer at 90 deg from the transducer generating the standing wave, a three-dimensional volume of stable single bubbles can be established. Further, the addition of the second transducer operating at the same frequency stabilizes the bubble pattern so that individual bubbles may be studied. The size of the bubbles and the separation of the standing waves depend on the frequency of operation. Two transducers, operating at frequency of 630 kHz, provided the most consistent results for the configuration used in this study. The bubbles exhibit bright sonoluminescence. Spectral measurements are in progress. Effect of the shape of the configuration will be discussed

along with the standing wave patterns, spectral data, and pictorial results of separation of individual bubble sonoluminescence in a multibubble sonoluminescence environment.

4:45

2pPA9. Physical analysis of a stable cavitation bubble structure at high acoustic intensity. Bertrand Dubus, Alexei Moussatov, Christian Granger (IEMN dpt ISEN, UMR CNRS 8520, 41 bd Vauban, 59046 Lille Cedex, France, bertrand.dubus@isen.fr), Cleofe Campos-Pozuelo (CSIC, Madrid, Spain), Christian Vanhille (Universidad Rey Juan Carlos, Madrid, Spain), Robert Mettin, Topi Tervo, and Werner Lauterborn (Gottingen Univ., Germany)

A cavitation bubble structure stable at high acoustic intensity (from 1.8 to more than 8.2 W/cm^2) has been experimentally observed [A. Moussatov *et al.*, Ultrason. Sonochem. **10**, (2003)]. At the vincinity of an axisymmetrical radiating surface, big streamers of bubbles get ejected from the surface and build up a bubble structure of conical shape denoted CBS. In this paper, results on the observation and analysis of the CBS are reported for 20-kHz horn-type transducers with different sonotrode diameters. It is found that: (i) the CBS is always a zone of high chemical activity, even when the bubble structure is not observed due to high speed streaming; (ii) the geometry of the CBS is determined by nonlinear acoustic wave propagation. These results are supported by various experimental data: chemiluminescence measurements, high speed movies (2250 frames/s) under CW scattered light or LED flashes synchronized with driving signal and measurement of the acoustic pressure and of the time-averaged acoustic pressure. [Work supported by CNRS-CSIC cooperation project and European Union (FEDER 2).]

5:00

2pPA10. The influence of ultrasound power on multibubble sonoluminescence intensity from aqueous solutions containing surface active solutes. Devi Sunartio, Muthupandian Ashokkumar, and Franz Grieser (School of Chemistry, Univ. of Melbourne, VIC 3010, Australia)

Water soluble surface active solutes have been found to affect the intensity of multibubble sonoluminescence (MBSL) in aqueous solutions. For example, the presence of charged surfactants enhances the MBSL intensity relative to that observed from pure water, whereas the presence of volatile surface active solutes decreases the MBSL intensity in reference to pure water [J. Phys. Chem. B **101**, 10845 (1997)]. Further investigations on how ultrasound power influences the effect of surface active solutes on MBSL intensity have shown that ultrasound power plays an important role in governing the above mentioned effects. The relative enhancement in MBSL intensity by charged surfactants has been found to vary with changes in ultrasound power level; the relative enhancement decreases with an increase in the ultrasound power level. Also, the extent of MBSL quenching by alcohols has been found to increase with an increase in the applied ultrasound power level; no SL quenching at lower power levels and $>80\%$ SL quenching at higher power levels have been observed. The influence of ultrasound power on the population of "active" bubbles and the cavitation bubble temperature, in aqueous solutions containing surface active solutes, will be discussed in order to rationalize the observed experimental data.

TUESDAY AFTERNOON, 16 NOVEMBER 2004 ROYAL PALM SALON 5, 1:30 TO 5:00 P.M.

Session 2pSA

Structural Acoustics and Vibration: Structural Acoustics in MEMS

Karl Grosh, Chair

Mechanical Engineering Department, University of Michigan, 2350 Hayward Street, Ann Arbor, Michigan 48109-2125

Invited Papers

1:30

2pSA1. Viscothermal wave propagation, including acousto-elastic interaction. Willem Beltman (Intel, 2111 NE 25th Ave., M/S JF3-254, Hillsboro, OR 97124, willem.m.beltman@intel.com)

Standard wave equation models neglect the effects of viscosity and thermal conductivity on acoustic wave propagation. For wave propagation in narrow tubes or thin layers, like in MEMS devices, this might not be accurate. This presentation outlines models that take the effects of inertia, viscosity, thermal conductivity and compressibility into account. Three classes of models are described and characterized based on the use of dimensionless parameters. The most important parameter is the shear wave number, an unsteady Reynolds number that indicates the ratio between inertial and viscous effects. It is shown that for most applications the low reduced frequency model is sufficient and the most efficient. The wave propagation models are coupled to the structural models to capture the acousto-elastic interaction. For simple geometries, analytical solutions can be found for these coupled analysis cases. For more complex geometries, a finite element model was developed, based on the low reduced frequency model, in which viscothermal acoustic finite elements are coupled to structural elements. Examples of applications are presented.

2:00

2pSA2. Fluid–structure interaction in a physical model of the human cochlea. Michael J. Wittbrodt, Charles R. Steele, and Sunil Puria (Mech. and Computation Group, Mech. Eng., 262 Durand Bldg., Stanford Univ., Stanford, CA 94305, wittbrod@stanford.edu)

Understanding the mammalian cochlea requires an understanding of the complex fluid structure interaction of an elastic partition separating two fluid channels. Using a combination of microfabrication and macrofabrication technologies, a passive physical model was developed. Compared to most MEMS devices the elastic partition is large. The width is tapered from 100 to 500 μ over the 36-mm length. Orthotropic properties are achieved with 9000 discrete aluminum fibers supported by either 1 or 5 μ of a polyimide thin film. Two fluid channels are macromachined from plastic and filled with saline. A magnet-coil system excites the fluid channel. The model demonstrates a traveling wave which peaks at a characteristic place. Normalized responses show gains of 2.5 and 3 at 29 and 23 mm from the basal end with phase lags of 3 pi and 5 pi for 8 and 18 kHz, respectively. Calculations using the WKB asymptotic approximation confirm the general character of the responses measured. The presence of fluid provides an efficient means of transporting the wave energy to a characteristic place on the elastic partition followed by a sharp roll-off in response. [Work supported in part by grants DC03085 and DC05454 from the NIDCD of NIH.]

2:30

2pSA3. Sensors at small scales for applications in fluid structure interaction problems. Vasundara V. Varadan (Eng. Sci. Elect. Eng., The Pennsylvania State Univ., University Park, PA 16804, vvvesm@engr.psu.edu)

This talk will present an overview of the state of the art in the development of sensors at the nano and micro scale that are particularly suited for the study of small scale effects in fluid structure interaction problems. The parameters of interest pertain to pressure, shear stress, thermal effects, material surface modifications, and sensors and devices that play a role in microfluidics for applications in materials analysis on a chip as well as biomedical uses. In several applications wireless telemetry of data is a necessity and special challenges arise if devices are immersed in a liquid. Power sources for such devices as well as the telemetry system is also a research issue. Passive sensors such as remotely excited rf surface acoustic wave sensors are also of great interest. Real time correlation of data from several sensors measuring either the same or different parameters is also needed to provide parametric data for adaptive computer models that start with incomplete system models but rely on real time experimental data to evolve the computation. Active collaboration is needed between researchers working on device design and development and those studying fluid–structure interaction problems either experimentally or numerically.

3:00

2pSA4. Aspects of acoustics in MEMS devices. Jeffrey Dohner, Mark Jenkins, and Timothy Walsh (Sandia Natl. Labs., P.O. Box 5800, Albuquerque, NM 87185)

In this talk we will present an overview of aspects of structural acoustics in MEMS devices at Sandia. Recent results concerning viscous wave motion in a rotational micromechanical system will be presented. In addition, experimental approaches for the characterization of acoustic effects in MEMS devices will be discussed. We will also present an overview of massively parallel numerical simulation capabilities for large-scale and small-scale (MEMS) structural acoustic analysis. Since large numbers of degrees of freedom are typically present in structural acoustic analysis, massively parallel computations are essential in solving practical application problems. Two formulations for the fluid will be considered: a standard linear velocity potential formulation, and a nonlinear wave formulation. The applications of the two formulations will be discussed. A parallelization scheme will be presented that allows for arbitrary decomposition of the wet interface. Finally, numerical results of application problems will be presented. [Sandia is a multiprogram laboratory operated by Sandia Corporation, a Lockheed Martin Company for the United States Department of Energy's National Nuclear Security Administration under Contract DE-AC04-94AL85000.]

3:30–3:45 Break

3:45

2pSA5. An all surface-machined MEMS microphone. Gary W. Elko (Avaya Labs, 233 Mt. Airy Rd., Basking Ridge, NJ 07920, gwe@ieee.org), Flavio Pardo, Daniel Lopez, and David Bishop (Bell Labs, Lucent Technologies)

An all-surface machined MEMS microphone for telephony applications was constructed a few years ago at Bell Labs. This microphone was built-up from multiple layers of standard polysilicon. After etching, the polysilicon layers were mechanically actuated in an origami-like fashion, to form a condenser microphone shaped in a tetrahedral structure. The overall size of the microphone was roughly 300 μ on a side. The enclosed back reference volume was designed in concert with a unique multiple cantilever beam design to realize a diaphragm resonance frequency of approximately 20 kHz. A discrete component preamplifier based on a modulated carrier detection circuit to reduce $1/f$ preamplifier noise was constructed. The predicted overall noise level of the microphone was approximately 45 dBA. The measured noise was significantly higher, due most likely to stray capacitance of the lead wires to the external discrete preamplifier. Some design issues related to mechanical-thermal noise will be discussed and some suggestions to mitigate self-noise will be given.

Contributed Papers

4:15

2pSA6. Fluid–structure waves in a micromachined variable impedance waveguide. Robert White, Niranjan Deo, and Karl Grosh (Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48109, grosh@umich.edu)

A micromachined fluid–structure system, intended to demonstrate a unique cochlear-like acoustic sensing scheme, has been fabricated and analyzed. The system consists of a 0.11-mm-deep Pyrex fluid chamber, anodically bonded to a silicon structure housing a tensioned membrane. The membrane varies in width from 0.14 to 1.82 mm. Both isotropic LPCVD silicon nitride membranes and orthotropic nitride/polyimide membranes have been fabricated. Silicone oils of either 5 or 20 cSt viscosity are used. Laser vibrometry measurements show strong fluid–structure traveling waves. Wave speeds are between 50 and 300 m/s in the 4- to 35-kHz band. These traveling fluid–structure waves exhibit maximum structural motion at a location determined by their frequency. Wave decay rate is influenced by the viscosity of the fluid and, after the waves become evanescent, by membrane orthotropy. Results from finite element computations of an orthotropic membrane coupled to a compressible viscous fluid are compared with experiment. The fluid is modeled using either a two-dimensional thin-film approximation or the three-dimensional linearized Navier–Stokes equation. Both models accurately predict the

observed fluid–structure response. Nondimensional parameters controlling system performance, and regions of applicability of the models, will be discussed. [Work supported by ONR, NIH, and NSF.]

4:30

2pSA7. A theory for flexural vibration of thin plates with thermoelastic damping. Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ 08854-8058, norris@rutgers.edu)

Thermoelastic damping in thin plates can be the dominant loss mechanism under certain circumstances, e.g., as demonstrated by recent measurements for MEMS paddle oscillators. However, modeling of thermoelastic damping in flexural vibration of plates has lagged, despite the classical theory of Zener for beams. This talk shows how the thinness of the plate permits a very accurate approximation to the fully coupled system of stress and temperature. In this asymptotically valid theory, the flexural plate equations are modified by damping terms of two types. The first are viscosity-like terms in the plate equation of motion. These depend on the local curvature, being zero at saddles. The damping also modifies the boundary moment and shear force. Both the bulk and boundary effects are important in estimating the Q for resonant modes. The theory is derived from a generalized form of Hamilton's principle, using Kirchhoff's kinematic assumption and asymptotic approximations for small thermal coupling valid for all relevant materials. Modal damping can be estimated simply for certain boundary conditions, including fixed edges. Plates with free or partially free edges present a more interesting challenge, and various alternative estimates of Q are presented and illustrated numerically for rectangular plates and MEMS structures.

4:45

2pSA8. Development of film bulk acoustic wave resonators based on piezoelectric aluminum nitride. Hisanori Matsumoto, Kengo Asai, and Mitsutaka Hikita (Hitachi, Ltd., Central Res. Lab., 1-280, Higashi-koigakubo, Kokubunji-shi, Tokyo 185-8601, Japan, h-matsu@crl.hitachi.co.jp)

Film bulk acoustic wave resonators (FBARs) have high quality factors (Q) and are smaller than conventional surface acoustic wave devices, so they are particularly suitable for constructing the bandpass filters of mobile phones. This report covers the simulation and fabrication of FBARs consisting of an aluminum nitride (AlN) piezoelectric layer sandwiched between molybdenum (Mo) electrode layers. The simulation was based on Mason's equivalent circuit model. Other than material Q, the AlN material constants for the FBAR model were those reported by Tsubouchi and Mikoshiba [K. Tsubouchi and N. Mikoshiba, IEEE Trans. Sonics Ultrason. **32**, 634–644 (1985)]. The material Q of AlN was assumed to be 1000. Thickness values for the AlN and Mo films were determined by simulation, after which the FBARs were fabricated. Reactive radio-frequency sputtering and direct-current sputtering were adopted for deposition of the AlN and Mo films, respectively. In terms of frequency properties, our FBARs achieved Q of 800 and an effective electromechanical coupling coefficient of 5.7%. The transmission loss was greater than that calculated through simulation. This result suggests that the material Q of AlN is less than 1000. This may be due to suboptimal film quality.

TUESDAY AFTERNOON, 16 NOVEMBER 2004 — CALIFORNIA ROOM, 2:00 TO 5:00 P.M.

Session 2pSC

Speech Communication: Measuring and Modeling Voice and Talker Characteristics (Poster Session)

Roger W. Chan, Chair

University of Texas Southwestern Medical Center, Otolaryngology-Head Neck Surgery, 5323 Harry Hines Boulevard, Dallas, Texas 75390-9035

Contributed Papers

All posters will be on display from 2:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:00 p.m. to 3:30 p.m. and contributors of even-numbered papers will be at their posters from 3:30 p.m. to 5:00 p.m.

2pSC1. A search for listener differences in the perception of talker identity. Robert E. Remez, Stephanie C. Wissig, Daria F. Ferro, Kate Liberman, and Claire Landau (Dept. of Psychol., Barnard College, 3009 Broadway, New York, NY 10027)

Talkers differ along several dimensions that listeners can resolve. These include acoustic effects of native variation in the scale and shape of the articulatory anatomy, and the effects of age and use on the tissues of the vocal tract. Linguistically, talkers differ in phonetic habits occasioned by dialect and idiolect, and differ paralinguistically in manner of affective expression conveyed vocally. However, listeners are themselves likely to vary in sensitivity to intertalker variation. In this study we aimed to identify differences in sensitivity to talker variation as a function of linguistic experience. Speech samples were produced by female talkers 15–17 years old drawn from two dialect groups, one from Brooklyn, NY, and one from Bloomington, IN. Each talker produced sentences in a list-reading task. Listeners in our tests were also native either to Brooklyn or to Bloomington, and each was far more familiar with one dialect than the other. Tests of apparent similarity of talkers in each set were conducted with listeners from the same and different dialect group. The results of similarity scaling analyses calibrate the contribution of sensitivity to idiolectal contrast within and across dialect in the perception of a talker's characteristics. [Research supported by NIDCD.]

2pSC2. Relative contribution of temporal cues and spectral profile to voice gender discrimination. Sherol Chinchilla-Rodriguez, Geri Nogaki, and Qian-Jie Fu (House Ear Inst., 2100 W. 3rd St., DAIP, Los Angeles, CA 90057, schinchilla@hei.org)

Despite limited access to spectral and temporal cues, cochlear implant (CI) users are somewhat able to discriminate voice gender. The present study explored the relative contributions of spectral and temporal cues to normal-hearing (NH) subjects' voice gender discrimination, while listening to multi-channel simulations of CI processing. The output spectrum was either matched (relative to normal) or upwardly shifted to simulate the spectral shift associated with CIs; the envelope filter in each channel was varied to examine the contribution of temporal cues. Voice gender discrimination was tested with two talker sets, in which the mean fundamental frequency ($F0$) between male and female talkers was either widely or narrowly separated. Results showed that for both talker sets, 16 spectral channels were needed before subjects could use the spectral profile to identify voice gender; when the speech spectrum was shifted, 32 channels were needed. Given enough temporal cues, the spectral profile had a relatively small effect on discrimination when the $F0$ was widely separated between male and female talkers. When there was little $F0$ separation between male and female talkers, the spectral profile had a much stronger effect; however, 16 or more channels were needed before listeners could attend to the profile. [Work supported by NIDCD-RO1-DC004993.]

2pSC3. Temporal change in memory of human voice. Hiroshi Kido (Dept. of Commun. Eng., Tohoku Inst. of Technol., Sendai, Japan, kidoh@tohtech.ac.jp) and Hideki Kasuya (Utsunomiya Univ., Utsunomiya, Japan)

This study investigates the extent to which human subjects accurately retain memory of speaker individuality and voice quality of utterances without receiving instruction to memorize them. Experiments constituted three phases. In phase one, 27 subjects listened to two target utterances in a paired-comparison test that required only a similarity judgment of voice quality. The target utterances were one sentence read by two males with average and characteristic voice quality, respectively, who were selected from 109 males. In phase two, performed 21 days after phase one, the same subjects evaluated Japanese expressions of voice quality [Proc. ICSLP-98, No. 1005] of the two previous target utterances based on memory only and judged with a degree of confidence whether each of seven speech stimuli (two targets and five additional speakers' utterances) was identical to one of the two targets. In the final phase, conducted immediately after phase two, the subjects repeated the same phase-two task but after listening to the two target utterances. Data processing based on the signal detection theory revealed that temporal change in memory of speaker individuality and voice quality was relatively small over a period of 21 days. Experiments over a longer period of time are now underway.

2pSC4. Multiregression analysis of autoregressive with exogenous input speech synthesis parameters and voice qualities. Hideki Kasuya, Masayoshi Kawamata (Utsunomiya Univ., 7-1-2 Yoto, Utsunomiya, Japan, kasuya@klab.jp), and Hiroshi Kido (Tohoku Inst. Technol., Sendai, Japan)

This study investigates the relationship between acoustic parameters utilized in the formant-based ARX (autoregressive with exogenous input) speech synthesis model (J. Acoust. Soc. Jpn., **58**, 386–397) and perceived voice qualities of synthetic speech. The acoustic parameters manipulated were $F0$ baseline, $F0$ range, spectral tilt of glottal flow (TL), formant scaling parameter (FS), and speaking rate (SR). Japanese expressions associated with voice qualities were high-pitched/low-pitched, masculine/feminine, hoarse/clear, calm/excited, powerful/weak, youthful/elderly, thick/thin, and tense/lax (Proc. ICSLP-98, No. 1005). A sentence utterance of an average speaker selected from a database of 109 male speakers was analyzed using the ARX method. Each of the five acoustic parameters of the utterance was manipulated at three levels, producing 243 samples of synthetic speech $(3\times3\times3\times3\times3)$. Ten subjects evaluated the voice qualities of each of the 243 synthetic stimuli with regard to the eight Japanese expressions. Multiregression analysis showed that $F0$ range, $F0$ baseline, and FS were primary acoustic correlates of high-pitched/low-pitched and masculine/feminine, SR and $F0$ range for calm/excited, and $F0$ range, SR and $F0$ baseline for thick/thin. Significant relations were not found for the remainder of the Japanese expressions, which was thought to be associated in part with irregularities of glottal flow.

2pSC5. Fibrous proteins and tensile elasticity of the human vocal fold. Roger W. Chan (Otolaryngol. Head and Neck Surgery, Grad. Program in Biomed. Eng., Univ. of Texas Southwestern Medical Ctr., Dallas, TX 75390-9035), Neeraj Tirunagari, and Min Fu (Univ. of Texas Southwestern Medical Ctr., Dallas, TX)

Viscoelastic response of the human vocal fold under tension has been reported previously, demonstrating nonlinearity and hysteresis of stress–strain curves. However, the importance of various matrix molecules for tensile elasticity of the vocal fold is not well understood. It is important to correlate the biomechanical behavior of the vocal fold with its histological microstructure, so as to examine the relative contributions of the matrix proteins such that cultured tissue constructs for repairing voice disorders may be engineered accordingly. This study attempted to test the hypothesis that collagen and elastin play a major role in determining the tensile elasticity of the vocal fold. Uniaxial tensile elastic properties of the vocal cover and vocal ligament were quantified using a servo-control lever system. The specimens were also studied histologically with elastin van Gieson stain. Digital image analysis of the vocal fold cover showed that a more profoundly nonlinear stress–strain curve and a higher elastic modulus (tangent Young's modulus) were associated with higher densities of collagen and elastin in the specimens. These findings suggested that both collagen and elastin fibers likely contribute significantly to elasticity of the vocal fold under tension, thereby regulating vocal fold length changes and fundamental frequency control. [Work supported by NIH.]

2pSC6. Source spectra for excised and latex phonatory models. Fariborz Alipour (Dept. of Speech Pathol. & Audiol., Univ. of Iowa, Iowa City, IA 52242) and Ronald C. Scherer (Bowling Green State Univ., Bowling Green, OH 43403)

An excised larynx model and a latex physical model were used to study acoustic spectra of the phonatory source as a function of subglottal pressure, glottal adduction, and vocal-fold length. A supraglottal vocal tract was not used, indicating that the acoustic signal corresponded to the output glottal flow. Each model was mounted over an -inch tracheal tube through which flowed pressurized, heated, and humidified air. The subglottal pressure and EGG signal (excised model) were recorded on a personal computer. The mean flow rate, mean subglottal pressure, and SPL were recorded manually, and adduction was specified by the use of interarytenoid shims. The output audio signals from the larynx models were recorded on a DAT recorder. Spectral information of the audio signal was obtained via FFT analysis (MATLAB). Preliminary data indicate that the spectral slope did not have a constant dB/octave rate, and spectral slope had a primary dependence on subglottal pressure and a secondary dependence on adduction. Other acoustic details and the differences between the two models will be discussed. [Work supported by NIDCD Grant Number DC03566.]

2pSC7. Vocal tract influence on medial surface dynamics of the vocal folds. Michael Doellinger, David Berry (The Laryngeal Dynam. Lab., UCLA School of Medicine, 31-24 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095-1794), and Douglas Montequin (Univ. of Wisconsin, Madison, WI 53792-7375)

In previous work, quantitative measurement of the medial surface dynamics of the vocal folds was reported using a hemilarynx methodology. The technique was applied to excised larynges from both humans and canines, as well as to *in vivo* canine larynges. In the present investigation, a vocal tract was attached to the excised hemilarynx preparation, and systematic changes were made in vocal tract shape to study its influence on the medial surface dynamics. In particular, the width of the epilarynx was varied across experiments. Previously, in a similar experiment, phonation threshold pressure was investigated as a function of epilarynx width. It has also been shown that epilarynx width has an influence on glottal volume velocity and the acoustic output [I. R. Titze and B. H. Story, J. Acoust. Soc. Am. **101**, 2234–2243]. However, the direct influence of the epilarynx width on vocal fold vibration has never been quan-

tified. The present investigation considered the impact of epilarynx width on the medial surface dynamics, including the underlying empirical eigenfunctions which make up the vibration patterns. Further, quantitative measures such as displacement and velocity were reported and compared. [Work supported by NIH/NIDCD Grant No. R01 DC03072.]

2pSC8. The application of chaotic dynamics, synchronization, and parameter estimation in an asymmetric vocal fold system. Y. Zhang and J. J. Jiang (Dept. of Surgery, Div. of Otolaryngol. Head and Neck Surgery, Univ. of Wisconsin Med. School, Madison, WI 53792-7375)

Methods from nonlinear dynamics, including Poincare map, Lyapunov exponent and dimension, are applied to describe the vibrations of a vocal fold model with tension, stiffness, and mass imbalances. Bifurcation diagrams illustrate the effects of these imbalance parameters. When tension, stiffness, and mass imbalance parameters deviate from the normal value of 1, chaotic vibrations with positive Lyapunov exponents may occur. Furthermore, the technique of feedback synchronization allows us to manipulate a simulator to approach the vibratory patterns of an original vocal fold system. The minimal glottal area is applied as a feedback variable connecting between two systems. The simulator and the original system are synchronized when their state differences asymptotically converge to zero. The effects of noise and parameter mismatches on synchronization are investigated. Finally, a parameter estimation scheme based on feedback synchronization is applied. Despite noise perturbations and large initial parameter mismatches, the original system parameters can be reproduced in the simulator with parameter controls, and two chaotic vocal fold systems can be synchronized. A parameter estimation scheme shows the potential application to extract asymmetric biomechanical parameters of the vocal folds from a time series of the glottal area. [Work supported by NIDCD Grant No. R01 DC006019-01.]

2pSC9. An acoustic and electroglottographic study of V[glottal stop]V in two indigenous American languages. Christina Esposito (Dept. of Linguist., UCLA, 3125 Campbell Hall, Los Angeles, CA 90095) and Rebecca Scarborough (UCLA, 3125 Campbell Hall, Los Angeles, CA 90095)

Both Pima, a Uto-Aztecan language spoken in Arizona, and Santa Ana del Valle Zapotec (SADVZ), an Otomanguean language spoken in Oaxaca, Mexico, have sequences of two vowels separated by an intervening glottal stop. In both languages, this V?V sequence becomes reduced in certain occurrences, with the perceptual effect of the loss of /?/ in Pima and the loss of V2 in SADVZ. The purpose of this study is to provide an acoustic and electroglottographic (EGG) description of these sequences in both full and reduced forms, prompted by varying speech rate. Two acoustic measures of phonation type (H1–H2, H1–A3) and two EGG measures (OQ and peak closing velocity) were made for each vowel at the midpoints and adjacent to /?/. For Pima, an issue of interest is what properties of the /V?V/ sequences (when V1=V2) allow them to be distinguished from phonemic long vowels in the reduced forms where /?/ is lost. It is hypothesized that /?/ will be preserved as vowel glottalization. For SADVZ, an important question is what property of V?(V) (where V2 is devoiced), V?, and creaky vowels allow them to be distinguished from each other in reduced forms, given that they are all characterized by glottalization.

2pSC10. The feature [stiff] interacts with intonation to affect vocal-fold vibration characteristics. Helen M. Hanson (36-585, MIT RLE, 77 Massachusetts Ave., Cambridge, MA 02139, hanson@speech.mit.edu)

As part of a larger study of the effect of prosody on segmental cues, previous work has shown that in a high pitch environment, $F0$ is significantly increased relative to a baseline following voiceless obstruents, but $F0$ closely follows the baseline following voiced obstruents. When a syllable carries a low or no pitch accent, $F0$ is increased only slightly following all obstruents. It is suggested that this difference occurs because demands of the segmental and prosodic levels of speech production conflict. In particular, results support a theory that the primary feature of voiced or voiceless obstruents is [.-stiff] or [+stiff] vocal folds, respectively. Enhancing features such as [-.spread] or [+spread] vocal folds are secondary. In a high pitch environment, the feature [-.stiff] conflicts with the need to raise pitch. Because of enhancing gestures, the prosodic level can override the segmental. Likewise, in a low pitch environment, the feature [+stiff] conflicts, so the vocal folds will not be stiffened for voiceless obstruents. If so, one might expect that stop consonants in low pitch environments will have longer or stronger prevoicing than in high pitch environments, and preliminary data show that speakers do show this tendency. [Work supported by NIH Grant DC04331.]

2pSC11. Acoustic characteristics of whispered vowels. Michael Kiefte (School of Human Commun. Disord., Dalhousie Univ., Halifax, NS B2Y 1P9, Canada)

It is well known that whispered speech is able to convey information that is normally associated with pitch. For example, it is possible to whisper the question "You are going today?" without any syntactic information to distinguish this sentence from a simple declarative. It has been shown that pitch change in whispered speech is correlated with the simultaneous raising or lowering of several formants [e.g., Kallail and Emanuel, J. Speech Hear. Res. **27**, 245–251 (1984)]. Data will be presented from 81 native speakers of English from the Halifax region of Nova Scotia (35 men and 46 women) who were asked to phonate and whisper the vowels /i,ɪ,e,ɛ,æ,≳,o,ʊ,u,ʌ,ɚ,ɔɪ,aʊ,aɪ/ at three different pitches across a range of roughly a musical 5th. Formant frequency variability is much greater for whispered vowels with different intended pitches resulting in much greater between-category overlap. Listeners' categorizations of these stimuli will be reported as well as results from a discriminant analysis based on either static or dynamic spectral information. [Work supported by SSHRC.]

2pSC12. Perception of source spectral slope. Jody Kreiman and Bruce R. Gerratt (Head/Neck Surgery, UCLA School of Medicine, 31-24 Rehab Ctr., Los Angeles, CA 90095-1794)

Researchers broadly agree that the spectral slope of the voice source is an important concomitant of voice quality, but it is unclear which specific aspects of spectral slope are perceptually important. To examine this issue, 50 voice samples were inverse filtered, and a large variety of measures of spectral slope were calculated for the resulting source pulses. Factor analysis was applied to determine which of these measures of slope are in fact independent. Synthetic copies of the voices were generated. Listeners used a method of adjustment task to manipulate these independent aspects of spectral slope in the synthesizer so that the synthetic copies matched the natural voice samples. Patterns of listener agreement and variability provide information about the perceptual relevance of different acoustic measures of source spectral slope. [Research supported by NIDCD.]

2pSC13. Effects of spectral tilt on the perception of naturally spoken vowels. James M. Hillenbrand (Speech and Hearing Ctr., MS5355, Western Michigan Univ., Kalamazoo, MI 49008)

Experiments using synthetic signals with static spectral patterns have shown that spectral tilt can have an influence on vowel identity that is independent of the formant-frequency pattern [e.g., M. Kiefte and K. Kluender, J. Acoust. Soc. Am. **109**, 2294–2295 (2001)]. The purpose of this experiment was to determine whether spectral tilt affects vowel identity for naturally spoken utterances. Test signals consisted of filtered and unfiltered versions of 300 /hVd/ utterances selected from a large, multi-talker database [Hillenbrand *et al.*, J. Acoust. Soc. Am. **97**, 3099–3111 (1995)]. Phonetically trained listeners ($N=24$) identified the unfiltered signals and the same signals conditioned by a $+9$ dB/oct filter and a -9 dB/oct filter, presented in a single random order. The effects of the two filters were clearly audible to listeners, but there was no effect of spectral tilt on vowel recognition accuracy. These findings do not invalidate earlier results with synthetic speech; rather, they suggest that when natural cues to vowel identity are preserved, overall spectral tilt plays little if any role in vowel recognition (within the limits of the ±9 dB tilt manipulation). The findings would also seem to present a significant challenge to whole-spectrum models of vowel perception, though they are by no means incompatible with this approach. [Work supported by NIH.]

2pSC14. The Davis addendum to the Tomatis effect. Dorinne S. Davis-Kalugin (Davis Ctrs., Inc., 98 Rt. 46 W, Budd Lake, NJ 07828, davis@thedaviscenter.com)

The relationship of the ear and voice is defined in three laws, known as the Tomatis effect. Specifically, these laws state: (1) The voice only contains the harmonics that the ear can hear. (2) If you give the possibility to the ear to correctly hear the distorted frequencies of sound that are not well heard, these are immediately and unconsciously restored into the voice. (3) The imposed audition sufficiently maintained over time results in permanently modifying the audition and phonation. Using a time domain frequency analysis of the voice, the spontaneous otoacoustic emissions from the ear were evaluated and compared with a frequency analysis obtained through voice analysis. One hundred percent correlation between the stressed frequencies of the ear and voice was noted. An addendum to the Tomatis Effect describes these effects. First, the ear emits the same stressed frequencies that are emitted by the voice. Second, when complementary or supplementary frequencies of stressed frequencies are introduced via sound vibration to the ear, vocal patterns regain coherence.

2pSC15. A comparison of three algorithms for estimating aspiration noise in dysphonic voices. Mina Goor (Dept. of Elec. and Computer Eng., Univ. of Florida, Gainesville, FL 32611), Rahul Shrivastav (Univ. of Florida, Gainesville, FL 32611), and John G. Harris (Univ. of Florida, Gainesville, FL)

Dysphonic voices are frequently characterized by increased aspiration noise. Several algorithms have been proposed to quantify the noise present in such voices. Yet, an independent analysis of these algorithms has not been reported. These algorithms differ in a number of aspects such as the theory underlying these measurements, the procedures used for estimating the noise, and the nature of their output. Three different algorithms for estimating noise in dysphonic voices were implemented in MATLAB and their output for synthetic and natural voice samples compared. These algorithms include: (1) the pitch-predictive signal-to-noise ratio reported by Milenkovic [Workshop on Acoustic Voice Analysis: Proceedings (1994)], which analyzes signals in the time domain and provides a time waveform of the aspiration noise: (2) the harmonics-to-noise ratio reported by deKrom [J. Speech Hear Res., **36**, 254–266 (1993)], which performs a cepstral analysis to provide the spectrum of the aspiration noise; and (3) the glottal-noise excitation reported by Michaelis, Frohlich, and Strube [J. Acoust. Soc. Am. **103**, 1628–1639 (1998)], which measures the correlation of the Hilbert transform of different frequency bands. The result of the study will help identify the algorithm(s) most suitable in the prediction of the listener judgments of voice quality. [Research supported by NIH/R21DC006690-01.]

TUESDAY AFTERNOON, 16 NOVEMBER 2004 PACIFIC SALON 3, 1:00 TO 3:00 P.M.

Session 2pSP

Signal Processing in Acoustics and Underwater Acoustics: Signal Processing Arrays with Many Elements in Novel Configurations or Novel Environments Part II

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Contributed Papers

1:00

2pSP1. Simulation of synchronized animal calling with a distributed sensor network. Efosa Ojomo, Praveen Mudindi, Isaac Amundson, and Kenneth D. Frampton (Dept. of Mech. Eng., Vanderbilt Univ., Nashville, TN 37235)

A distributed sensor network has been programmed to simulate synchronized calling exhibited by many species of frogs and insects. The distributed sensor network consists of numerous sensor nodes consisting of a microprocessor, wireless communications and a sensor board containing a buzzer and microphone. The goal was to program the sensor array to mimic the synchronized calling behavior (inspired in no small part by the recent appearance of cicada brood X). To this end, a single leader node was programmed to begin calling (i.e., buzzing). The remainder of the node array was programmed with an algorithm which allows them to detect other buzzing nodes and to synchronize their own buzz to that of the group. After start-up transients, the entire array calls in synchrony. Of particular interest is the system's transient behavior system when perturbed or when a second leader node begins calling out of synch with the array. The hardware and software algorithms will be described. Furthermore, numerous nodes will be distributed throughout the audience and a demonstration of system behavior will be provided. While this is a rather whimsical application of distributed array processing, it does demonstrate the unique system behaviors that can arise in truly distributed processing.

1:15

2pSP2. Optimized multistatic sonobuoy array patterns. Donald R. DelBalzo (Neptune Sci., Inc., 40201 Hwy., 190 E, Slidell, LA 70461, delbalzo@neptunesci.com), David P. Kierstead (Daniel H. Wagner, Assoc., Vienna, VA 22180), and Erik R. Rike (Neptune Sci., Inc., Slidell, LA 70461)

Standard patterns for monostatic sonobuoy fields were developed during the Cold War for deep, uniform underwater environments, where a simple median detection range defined a fixed spacing between sonobuoys (usually along staggered lines). Oceanographic and acoustic conditions in littoral environments are so complex and dynamic that spatial and temporal variability of low-frequency signal and noise fields destroys the basic homogeneous assumption associated with standard tactical search concepts. Genetic algorithms (GAs) have been applied to this problem to produce near-optimal, nonstandard search tracks for monostatic mobile sensors that maximize probability of detection in such inhomogeneous environments. For the present work, a new capability, SCOUT (sensor coordination for optimal utilization and tactics) was developed to simulate multistatic distributed-sensor geometries and to optimize the locations of multistatic active sonobuoys in a complex, littoral environment. Both standard and SCOUT-derived tactics were evaluated for cumulative detection probability and compared. The results show (a) that the standard pattern is not optimal even for a homogeneous environment and (b) that standard patterns are grossly ineffective in inhomogeneous environments. [Work was sponsored by NAVAIR.]

1:30

2pSP3. Performance analysis of angle-or-arrival techniques in shallow water bathymetry measurements. Wen Xu, Marc Parent, and Fran Rowe (RD Instruments, 9855 Businesspart Ave, San Diego, CA 92131, wxu@edinstruments.com)

Conventional bathymetric side scan system measures the phase difference between two parallel rows of transducers and converts it to bottom echo signal direction and then depth. A main concern is that the phase measurements are easily dispersed by noises and interferences, particularly in shallow water applications. There have been considerable efforts to deal with the problem, for example, by adding more transducers and implementing signal angle-of-arrival estimation. In this work, experimental data anlaysis for a multiple-row system is conducted, disclosing shallow water echo signal spatial structure. In addition to a stable bottom return, there often exist returns from surface scattering or multi-path reflecting between surface and bottom, more spatially-dispersed depending on the surface wave condition. By comparing the differential phase approach, which is blind to interferences, to the angle-of-arrival approach, which exploits the multi-path signal structure, it is concluded that the differential phase approach can yield systematic bias toward surface in bottom depth measurements. Performance of the angle-or-arrival technique is then investigated. The Cramer-Rao bound is derived and evaluated in the presence of two signal sources, followed by discussions on source separation (resolution) and individual source strengths.

1:45

2pSP4. Low-frequency passive performance of a towed volumetric array during maneuvers. Jerrold Dietz, James Edgerton, Bruce Newhall, Juan Arvelo, Jr., and Catherine Frazier (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099)

For a large towed volumetric array, array element deformations significantly degrade system performance both during tow-ship maneuvers as well as during straight tows. Using a least squares fit for the shape estimation sensors we can generate accurate array shapes that allow us to maintain signal gain through array maneuvers. Array signal gain was calculated throughout maneuvers and is observed against measures of the array distortion. These calculations are compared for a linear, planar and volumetric array. We will discuss the experimental results in addition to environmental modeling to determine the best achievable gains. [This work was supported by the Defense Advanced Research Project Agency (DARPA) under the Robust Passive Sonar (RPS) project.]

2:00

2pSP5. Boundary array size for teleconferencing. Dwight F. Macomber (Belar Electron. Lab., Inc., Devon, PA)

Boundary arrays for teleconferencing require large numbers of sensors to be effective. The paper describes efforts to estimate the number of microphones required for an effective teleconference boundary array. Measurements of many source-to-sensor room transfer functions (RTF) were made in a conference room of moderate size. The direct arrivals of these RTFs were time aligned in software to estimate the transfer function of a delay-and-sum boundary array. Using the image method of Allen and Berkley, another set of RTFs was generated for a virtual room acoustically similar to the measured room. These RTFs were time aligned to create a second virtual array. The S/Rs of these two virtual arrays compare favorably. The image method was used to estimate transfer functions of virtual boundary arrays of different sizes for a 60-cubic-meter room. Standard beamforming and matched-filter processing were used. TIMIT speech was convolved with the virtual array transfer functions to generate audio used for evaluation. A listening test ranked the auditory performance of four single-microphone reference systems and eight test arrays in the virtual room. For rooms of roughly 100 cubic meters, boundary arrays with approximately 100 microphones may provide good subjective performance.

2:15

2pSP6. Acoustic array data compression via Karhunen-Loeve transform. Frank A. Boyle and Thomas H. Phipps (Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029, boyle@arlut.utexas.edu)

Acoustic data compression has been explored in several contexts and several techniques exist. The applicability of each technique depends on the type of data processing that is intended. For example, audio data is often compressed via perceptual coding methods (e.g., mp3) in which quantization noise is distributed according to psychoacoustic principles. The Karhunen-Loeve (KL) transform presents an opportunity to compress surveillance array hydrophone-level data while preserving relevant features in beamformed displays. A KL codec was formulated and applied to acoustic test data from a horizontal line array. The results appear promising in that features of interest are preserved with significant data compression. The presentation will include a description of the algorithm as well as examples with actual data. [Work funded by ONR.]

2:30

2pSP7. Nonlinear weighting techniques to replace tradeoffs between mainlobe width and sidelobe suppression. Dale B. Dalrymple (Signal Processing Systems Div. of Information Systems Labs., 10070 Barnes Canyon Rd., San Diego, CA 92121, ddalrymple@islinc.com)

When the discrete Fourier transform is used to transform data from one domain to another as in narrowband beamforming or spectral analysis a weighting function can be applied as a vector multiplication before the transform or as a convolution after the transform to control mainlobe width in opposition to sidelobe suppression. Nonlinear operations applied to transform outputs or weighted versions of the transform outputs can combine mainlobe width control with sidelobe suppression. Applying weights before the transform requires the execution of multiple transforms for this process. The use of small kernals in the frequency domain on the output of a single transform provides a more efficient implementation. The set of small kernal summed cosine weights provides a variety of responses that can be combined nonlinearly retaining the best traits of each. Kernals can also be calculated adaptively.

2:45

2pSP8. Applications of beamforming and acoustic holography in an anechoic tank with surface reflections. Sea-Moon Kim, Youngchol Choi, and Yong-Kon Lim (Yuseong, P.O. Box 23, Daejeon 305-600, Korea)

Recently, our institute, KRISO/KORDI, constructed a small anechoic tank which has an anechoic lining at the four walls and bottom. Because surface reflections still occur, special care must be taken for acoustic measurements and array signal processing, for example, source localization and acoustic holography. In this paper the effects of the surface reflections to the array signal processing methods are investigated. First, estimated errors by both methods are analyzed for various measurement parameters. Beamforming experiments with vertical line arrays are performed. Holographic reconstruction of simple and complex acoustic sources is also performed to compare the true and estimated results.

Session 2pUW

Underwater Acoustics: Propagation and Modeling

Kevin B. Smith, Chair
Department of Physics, Naval Postgraduate School, Code PH/SK, Monterey, California 93943

Contributed Papers

1:00

2pUW1. Beam summation algorithm for wave radiation and propagation in stratified media. Tal Heilpern and Ehud Heyman (School of Elec. Eng., Tel Aviv Univ., Tel Aviv 69978, Israel, heyman@eng.tau.ac.il)

An efficient Gaussian beams summation (GBS) algorithm for tracking source excited wavefields in plane stratified media is introduced. It has two important features: (a) it involves an efficient calculation of the GB propagators, and (b) it involves rather sparse lattice of beams. For (a) we approximate the medium using layers with constant wavespeed gradient, and derive an efficient recursive algorithm for tracking the GB through such medium. This model not only reduces the number of layers, and thereby the algorithm complexity, in comparison with the conventional uniform layers model, but it also eliminates the reflection artifacts at layers interfaces. Property (b) is achieved by determining the beam expansion parameters for an efficient discretization of the source-excited beam spectra using a sparse lattice of beams. The algorithm has been validated and calibrated via thorough numerical comparisons with closed form ray solutions for source-excited fields in layered media. Perfect agreement between these independent solutions has been obtained in regions where the ray solution is valid, but the beam formulation also provided smooth and physically meaningful solutions in caustic regions where the ray solution fails.

1:15

2pUW2. Parabolic propagation in a weakly range-dependent duct: Approaching higher orders systematically. Robert F. Gragg (Naval Res. Lab., Code 7140, Washington, DC 20375, robert.gragg@nrl.navy.mil)

This work illustrates a technique that exploits energy-conserving transformations to split a CW field into a pair of components that propagate via uncoupled parabolic equations in opposite directions along the axis of a weakly inhomogeneous waveguide. A systematic series of these transformations produces this splitting at successively higher orders while avoiding backscatter. In the interest of clarity, the simplest possible nontrivial case is considered: waves of vertical displacement on a stretched membrane with a smooth density inhomogeneity along the y direction that forms a duct in the x direction. (This case is truly two-dimensional and its only environmental variable, density, is continuous.) This transformation technique provides an efficient means of incorporating the effects of weak environmental inhomogeneity in higher-order parabolic propagation. [Work supported by ONR.]

1:30

2pUW3. Modeling acoustic nonlinearities in marine sediments. B. Edward McDonald (Naval Res. Lab, Code 7145, Washington, DC 20375)

Investigations into the theoretical and numerical treatment of acoustic nonlinearities in marine sediments are motivated by mine countermeasures and other Naval interests related to explosions in shallow water. Agreement between theory and data for the nonlinearity parameter B/A has been demonstrated in early papers concerning saturated sands. In the more general case involving varying depth profiles of air and silt contents, however, it is observed that B/A values vary so greatly that deterministic nonlinear modeling approaches may have to be augmented to address environmental uncertainty. Comparisons are given between B/A values derived from static consolidation tests and available *in situ* data. Numerical profiles from the NPE shock propagation model are given to demonstrate the effects of environmental uncertainty. [Work supported by the ONR.]

1:45

2pUW4. A single-scattering solution that handles large contrasts across interfaces. Elizabeth T. Kusel, William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180), and Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

Single-scattering and energy-conservation approximations have both proven to be effective for solving range-dependent ocean acoustics problems. The energy-conservation approach is usually used in acoustic models since it provides greater efficiency. The single-scattering approach is more general and more promising for problems involving elastic layers. A range-dependent medium is approximated in terms of a series of range-independent regions separated by vertical interfaces. The single-scattering solution is usually implemented in terms of an iteration formula, which may diverge when there is a large contrast in the material properties across an interface. Convergence can be improved by artificially splitting a vertical interface into a series of slices with smaller contrasts. For the idealized problem of scattering from a single stair step, a significant error may be introduced by neglecting multiple scattering between slices. However, the approach should be accurate for problems involving gradual range dependence, such as sloping interfaces. [Work supported by the Office of Naval Research.]

2:00

2pUW5. Coupled perturbed modes for a wedge waveguide. Chris J. Higham and Chris T. Tindle (Phys. Dept., Univ. of Auckland, Private Bag 92019, Auckland, New Zealand)

A transformation of the normal modes is described to accommodate the density jump found in range-dependent penetrable bottom problems. The transformation is necessary for the application of perturbation theory to a coupled mode model of a wedge waveguide. The combination of conventional coupled modes and perturbation theory yields an efficient modal theory for shallow water acoustics of range dependence. The method is applied to upslope propagation in the ASA benchmark 2 wedge [J. Acoust. Soc. Am. **87**, 1499–1510 (1990)].

2:15

2pUW6. Acoustical propagation modeling using the three-dimensional parabolic equation based code 3DWAPE within a multiprocessing environment. Kaelig Castor (CEA/DASE, BP 12, FR-91 680 Bruyeres-le-Chatel, France), Frédéric Sturm (Laboratoire de Mecanique des Fluides et d'Acoustique, 69134 Ecully Cedex, France), and Pierre Franck Piserchia (CEA/DASE, FR-91 680 Bruyeres-le-Chatel, France)

In some particular oceanic environments involving bathymetric slopes and horizontal sound speed gradients, the azimuthal coupling can be significant. Fully three-dimensional models are thus needed to account for horizontal refraction. These models need usually high computational re-

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sources, especially for broadband calculations and/or for long-range paths where the 3-D effects are clearly accentuated. In this study, numerical simulations using the 3-D parabolic equation model 3DWAPE [F. Sturm, Acust. Acta Acust. **88**, 714–717 (2002)] are presented. The calculations are performed on a massively parallel computer providing a high computational efficiency. The message-passing interface (MPI) communication library is used. Two parallelization levels are considered. The first one allows a broadband-signal propagation by distributing independently on different processors the calculations for each frequency. The second one is based on a segmentation of the propagation matrices. Simple methods are used to perform an optimized multiprocessor implementation by equalizing processor workload, and split the number of processors between the two parallelization levels. An analysis of both speedup and efficiency of the algorithm is presented for several configurations. Computational time comparisons are shown for the 3-D ASA benchmark. The algorithm can also be applied to a realistic environment involving sound speed profiles and bathymetry data sets.

2:30

2pUW7. Sound propagation in shallow water with surface waves. Chris T. Tindle (Phys. Dept., Univ. of Auckland, Auckland, New Zealand) and Grant B. Deane (Scripps Inst. of Oceanogr., San Diego, CA)

Wavefront modeling can be used to calculate acoustic pulse propagation in shallow water with real surface waves. The method uses ray tracing but phase and amplitude are found from approximations to a phase integral. The lowest order approximation is the conventional geometric ray. Higher order approximations describe caustics, shadow zones and focusing. Reflection from sloping curved boundaries also leads to a range dependent horizontal wave number and requires modification of the phase integral and phase derivatives. Results are in good agreement with an experiment in which 10-kHz short pulses were transmitted in shallow water with 0.8-m peak-to-trough surface waves and a sloping bottom. [Work supported by ONR.]

2:45

2pUW8. A new primitive ray-tracing (PRT) model to compute forward and inverse acoustic propagation in a two-dimensional, range-dependent, sound-speed field. Sergey V. Vinogradov, Jerald W. Caruthers, Hans E. Ngodock (Dept. of Marine Sci., Univ. of Southern MS, 1020 Balch Blvd., Stennis Space Ctr., MS 39529), and Natalia A. Sidorovskaia (Univ. of Louisiana-Lafayette, Lafayette, LA 70504)

A primitive ray-tracing (PRT) model has been developed as part of a computational system to support potential tomographic observations. It is intended to estimate the variability in acoustic arrivals due to mesoscale oceanic circulation. The underlying algorithm is fully range dependent; it utilizes both vertical and downrange sound-speed gradients to compute ray trajectories. The ‘‘primitiveness’’ of the model means that it is simple enough to be inverted, while most existing forward acoustic models are too sophisticated computationally for that purpose. The inversion of this PRT takes in travel times and estimates possible oceanic structure, which further is being assimilated into an ocean circulation model. This paper provides details and results of acoustic computational experiments, along with a description of the PRT's application in a postulated tomographic observational system for the northeastern Gulf of Mexico.

3:00–3:15 Break

3:15

2pUW9. High-accuracy absorbing boundary conditions for high-order parabolic wave equations. Murthy N. Guddati and A. Homayoun Heidari (Dept. of Civil Eng., North Carolina State Univ., Campus Box 7908, Raleigh, NC 27695-7908)

Using a new systematic approach, a novel set of arbitrarily high-accuracy absorbing boundary conditions (HABC) is introduced for high-order parabolic equations. HABC is derived by finite-element discretization of the boundary, followed by applying a special integration scheme and imaginary stretching. The resulting boundary condition is then coupled with the interior and can be solved with the same numerical method as the interior, e.g., finite differences. The accuracy of the HABC is controlled by two parameters: (1) the number of absorbing layers (the order of the HABC); and (2) a reference phase velocity for each layer. The latter controls the wave number for which the HABC absorbs the wave field exactly. Hence, the absorption range, i.e., the distribution of the reflection coefficient, can be controlled by the coefficient matrices of the HABC. A specially designed explicit finite-difference scheme is used to solve the HABC. The added computational cost due to the HABC is negligible for practical purposes. The performance of the proposed boundary condition is shown through numerical examples for high-order parabolic equations, for different orders of the HABC and different sets of phase velocity parameters. Effective absorption of both propagating and evanescent waves is illustrated in the results. [Work supported by NSF.]

3:30

2pUW10. Fluctuations in sound transmission through velocity profiles that are periodic in range and in geo-time. Jacob George and Robert L. Field (NRL Code 7185, Stennis Space Ctr., MS 39529)

The lunar M2 tides cause variations in sound velocity profiles that are periodic in range and in geo-time. Acoustic transmissions through such an environment exhibit fluctuations at the primary tidal frequency as well as its overtones, due to nonlinear dependence on sound speeds. The number of tidal cycles within the source-receiver range is found to be a major factor that determines the magnitude and nature of the fluctuations. This has been investigated using parabolic equation (PE) models and by modal analysis. In the WKB model the important phase factor consists of a range integral of the horizontal wave number of each mode. A study of this integral has revealed a dramatic decrease in fluctuations when the number of tidal cycles within a constant source-receiver range increases. This result directly impacts transmissions that are at variable angles measured from the direction of the tides. Pulse transmission through the tidal environment has also been investigated, and the results are similar to those described above. [Work supported by ONR.]

3:45

2pUW11. Modeling the acoustical focusing properties of shoaling surf. Grant Deane (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0238) and Chris Tindle (Univ. Auckland, New Zealand)

Shoaling surf focuses sound, creating high intensity caustics that impact the performance of underwater communications systems and sonars in the very near shore region. Simple expressions for the time-varying caustic locus and amplitude can be derived from the Kirchhoff–Helmholtz scattering integral. The analysis shows that a focus is formed at a range equal to one-half the crests radius of curvature when the wave is directly over a shallow water source, and moves toward the crest and decreases in intensity as the wave shoals shoreward. The analysis is compared with numerical scattering calculations based on measured surface gravity wave profiles, and also with experimental data in which 10-kHz, single cycle pulses were transmitted over a 40 m path in 6 m of water just north of Scripps Pier. The impact of the caustics on sonar performance will be discussed. [Work supported by ONR.]

4:00

2pUW12. Time-domain sound propagation through a ship wake. Xiao Di, Lee Culver, and David Bradley (Appl. Res. Lab, The Penn State Univ., University Park, PA 16802)

A time-series model has been developed to visualize high-frequency pulses propagating through the bubbly wake of a ship. The wakes can have very large void fractions, and thus strong inhomogeneities in sound speed, scattering cross section, and acoustic attenuation. The model uses the Green's function PE to calculate the pressure field at frequencies within the bandwidth of the acoustic pulse, followed by Fourier synthesis to calculate time-domain solutions over a range of times. Special care was required to preserve the phase of the pressure at each frequency. Our concern is with quasihorizontal propagation and source–receiver ranges from about 100 to 1000 m, which contain effects of bubbles on propagation including attenuation, refraction, and scattering. We show predictions using a simple model for the distribution of bubbles in a ship wake. We find a very interesting frequency dependence for refraction and scattering. [Work supported by ONR Code 321.]

4:15

2pUW13. A geometrical model for surface ship wake. Boris Vasiliev (DRDC-Atlantic, 9 Grove St., Dartmouth, NS B2Y 3Z7, Canada, boris.vasiliev@drdc-rddc.gc.ca)

High-frequency geometrical measurements of a surface ship wake collected by other research groups will be presented. The general trends in the width data agree with the previously published measurements; however, the trends in the depth data differ from those published in the literature. On-going efforts in data analysis to develop a high-frequency surface wake model that accounts for wake width, depth and persistence will be discussed.

4:30

2pUW14. Identifying modes that produce late arrivals for low-frequency long-range propagation in shallow water. Harry DeFerrari, Irina Rypina, and Ilya Udovydchenkov (RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149)

Model and data comparisons are presented for acoustic propagation over a frequency range from 100 to 3200 Hz ($D=145$ m, $R=10$ km). Above 0.8 kHz most all energy is reflected from the water–sediment interface. Arrival times of waterborne paths are in close agreement with normal mode and PE predictions when observed sound speed profiles are used as inputs. Generally, the received signal can be closely approximately with $f/50$ modes since steeper SRBR modes are attenuated. For frequencies below 0.5 kHz, arrivals from waterborne paths are still present, but an additional group of later arrivals is also observed. The timing of these arrivals suggests either late SRBR or more complicated modes from energy passing well into the bottom. Unfortunately, geoacoustic properties of the subbottom are not known for depth greater than a few meters. Bottom properties at depth are assumed and then propagation models are used to determine the modes that most likely produce the late arrivals. Some modes having group velocities that correspond to the arrival time of the late arrivals penetrate a few tens of meters into the bottom. Coherence in time and space is computed and compared for bottom penetrating and waterborne paths.

4:45

2pUW15. Probability density function methods for uncertainty analysis in underwater acoustics. Kevin R. James and David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, 2019 Lay Auto Lab., 1231 Beal Ave., Ann Arbor, MI 48109)

Forward modeling of underwater acoustic propagation is generally successful when environmental parameters and boundary conditions are known. Unfortunately, such information is seldom available at the requisite level of precision, and any imprecision introduces uncertainty into sound field predictions. Quantifying this sound-field uncertainty is important for applications of acoustic propagation models such as matched-field processing. This presentation describes a method for quantifying the underwater-sound-field uncertainty arising from imperfect knowledge of the environment and its boundaries. It is based on formulating and solving a transport equation for the joint probability density function (PDF) of the real and imaginary parts of a harmonic sound field. The appropriate equation is obtained by combining spatial derivatives of the PDF with physical laws drawn from guided wave mechanics. The inputs for solving the PDF-transport equation are known or assumed distributions of the uncertain parameters. Solutions can be readily reduced to expected values, uncertainties, and confidence intervals for the predicted sound field. Results for simple test cases involving range-independent isospeed underwater sound channels are considered, and compared to solutions obtained analytically or through Monte Carlo simulations. [Work sponsored by ONR.]