Session 4aAAa

Architectural Acoustics, Noise, Engineering Acoustics, and Musical Acoustics: Sound Quality—When Sound is the Essential Quality

Richard H. Lyon, Cochair

RH Lyon Corp, 691 Concord Avenue, Cambridge, Massachusetts 02138

William M. Hartmann, Cochair

Department of Physics, Michigan State University, 4230 BPS Building, East Lansing, Michigan 48824

Chair's Introduction—8:00

Invited Papers

8:05

4aAAa1. Sound quality (SQ) of concert halls: Physical and subjective attributes. Leo L. Beranek (975 Memorial Dr., Cambridge, MA 02138, beranekleo@ieee.org)

Each new concert hall has the following stated goal: ''Acoustics equal to the best in the world.'' The owner can specify the number of seats, areas of public spaces, lighting intensities, etc. But, the attributes of acoustical quality cannot as yet be specified. Most acoustical consultants seem to feel that a ''seat of the pants'' experience is the only possible specification. But the architect's goal is a monument to himself and he believes the acoustical consultant should achieve the ''best in the world'' goal without visible means. Numbers for specifications are needed. In this paper 40 years of pertinent research are described: What are the critical physical attributes of good acoustics, how do we measure them, and how can they be translated into architectural specifications? Four steps have been involved: (1) interviews of conductors and music critics to determine (a) their acoustical rank orderings of a large number of halls and (b) which acoustical characteristics do they believe are important, viz., reverberance, strength of sound, etc.; (2) a determination of which physical measures correlate with their beliefs plus others that are physiologically important; (3) measurements of those physical quantities in the rank-ordered halls; and (5) the correlation of the measured values with the subjective quality ratings.

8:25

4aAAa2. Assessing the sound quality of a grand piano for different tuning standards. Hugo Fastl (Dept. of Human Machine Commun., Munich Univ. of Technol., Germany, fastl@ei.tum.de)

Grand pianos are tuned to a standard frequency of 440 Hz for a4. Eighty years ago, the tuning standard was lower, around 432 Hz. In Germany, a group of music lovers insists that the sound quality of a grand piano when tuned to 432 Hz is much superior to that of the same instrument when tuned to 440 Hz. Therefore, well-controlled psychoacoustic experiments were performed to check the validity of that argument. Using a Welte–Steinway reproduction grand piano of the Deutsches Museum Muenchen, with the exception of the tuning, all other features of the pieces of music used as stimuli could be kept the same. The advantage of using the Welte–Steinway lies in the fact that reproductions of the music of famous (deceased) artists are available, who at their time performed at a lower tuning standard. Recordings of the music played at 432 vs 440 Hz were made on DAT and presented to the subjects via headphones for sound quality evaluation. Psychophysical procedures like semantic differential or preference scaling by ''random access'' that have proven successful for the assessment of sound quality in the context of car interior sounds were used in the experiments.

8:45

4aAAa3. Sound quality and loudspeakers. Neil A. Shaw and Michael A. Klasco (Menlo Sci. Acoust., Inc., P.O. Box 1610, Topanga, CA 90290, menlo@ieee.org)

The sound quality for consumer, professional, and industrial products has been found to be an important part of the product design. For many products, sound is a byproduct of the normal operation of the product. In the case of loudspeakers, the production of sound is the normal operation of the product—its job is to produce sound. A survey of the current popular means of describing and measuring loudspeakers is presented. The introduction of some new means for measuring and describing raw transducers and systems is reviewed. Areas that are still problematic are also discussed.

9:05

4aAAa4. Basic requirements for realistic and unprejudiced evaluation of musical instruments. Klaus Wogram (PTB, Braunschweig, Germany)

In the past, the laboratory of musical acoustics of the PTB had to develop methods of subjective and objective evaluations of all kinds of musical instruments for a competition which was launched by the German ministry of economics. For this work the following questions had to be answered: (1) What makes a musical instrument play well from physical and subjective viewpoints? (2) How does the musician play the instrument, and what does he think when playing? (3) What is the influence of room acoustical parameters on sound quality? (4) How strong is the influence of prejudices concerning the brand, color, and type? (5) How can we measure the main acoustical parameters objectively? (6) What is the correlation between objective and subjective results? This presentation will give answers to these questions based on 10 years of experience with such evaluations by the author.

9:25

4aAAa5. Evaluating the sound quality of reproduction systems and performance spaces. David Griesinger (Lexicon, Inc., 23 Bellevue Ave., Cambridge, MA 02140)

Evaluation of sound reproduction systems through rapid A/B tests has led to enormous and rapid progress in system design. The sound quality of performance spaces is almost always evaluated through long term listening, and spaces are compared through remembered characteristics. Progress has not been rapid. This paper will present methods for rapid A/B comparisons of spaces and will demonstrate the results of such comparisons. The results can be very different from remembered impressions.

9:45

4aAAa6. Sound quality of brass-wind musical instruments. Robert W. Pyle, Jr. (11 Holworthy Pl., Cambridge, MA 02138, rpyle@post.harvard.edu)

Confronted with the phrase "sound quality" applied to brass-instrument sound, the typical player will respond, "Oh, you mean *tone*." For the player, this one word lumps together all spectrum-dependent features of the sound. The player may deliberately vary tone quality depending on the type of music being performed, the other performers involved, and the performance space. After a review of those features of tone quality common to all brass instruments and the differences between the various members of the family, some more subtle questions will be addressed. How and to what extent can the player control tone color without changing equipment? How does the player choose his or her equipment? How does the instrument maker offer choices to the player? How does the tone as heard in the audience differ from that heard by the player? Time permitting, there will be consideration of the difficulties in establishing a meaningful vocabulary for the discussion of tone quality.

10:05–10:20 Break

Contributed Papers

10:20

4aAAa7. Research opportunities in evaluating sound system performance. Bob Thurmond (G. R. Thurmond and Assoc., 9315 Springdale Rd., Austin, TX 78754)

Sound reinforcement has been both ubiquitous and indispensable in our society for many years. With such a volume of experience, it would seem that the accompanying body of knowledge should be sufficient to produce a successful design for every situation. However, the quality of actual systems installed today still varies from good to terrible. A few obvious aspects of this problem have been examined, and papers published, but no serious attempt to list or organize all contributors to sound performance quality has ever been made. In addition, no evaluative techniques or analytical strategies have been established to set standards. There are several valid reasons for this. The actual performance of a sound system is surprisingly complex, both in terms of the number of factors contributing to the overall performance and in the complexity within each factor. As a result, each system is quite unique and difficult to describe in precise terms. Few rigorous investigations have been applied. The results have been meager thus far and are essentially unknown to most practitioners. Research opportunities abound, and their results could have a significant impact. In this paper the current situation is summarized and topics for future study are suggested.

10:35

4aAAa8. Virtual acoustic prototyping. Marty Johnson (Vib. and Acoust. Labs., Dept. of Mech. Eng., Virginia Tech, VA 24061-0238)

In this paper the re-creation of 3-D sound fields so the full psychoacoustic impact of sound sources can be assessed before the manufacture of a product or environment is examined. Using head related transfer functions (HRTFs) coupled with a head tracked set of headphones the sound field at the left and right ears of a listener can be re-created for a set of sound sources. However, the HRTFs require that sources have a defined location and this is not the typical output from numerical codes which describe the sound field as a set of distributed modes. In this paper a method of creating a set of equivalent sources is described such that the standard set of HRTFs can be applied in real time. A structural-acoustic model of a cylinder driving an enclosed acoustic field will be used as an example. It will be shown that equivalent sources can be used to recreate all of the reverberation of the enclosed space. An efficient singular value decomposition technique allows the large number of sources required to be simulated in real time. An introduction to the requirements necessary for 3-D virtual prototyping using high frequency Statistical Energy Analysis models will be presented. [Work supported by AuSim and NASA.]

10:50

4aAAa9. An overview of suitable measurement and assessment techniques for distributed mode loudspeakers. Nick Hill (NXT, Huntingdon PE29 7HJ, UK) and Peter Mapp (Peter Mapp Assoc., Colchester C03 4JZ, UK, Petermapp@btinternet.com)

The sound field generated by a distributed mode loudspeaker (DML) can exhibit some characteristics that may be unfamiliar to conventional cone loudspeaker designers. In particular, the modal vibration of the diaphragm gives rise to a wide range of directivity patterns dependent on the specific design of the unit. This paper outlines some of the possible behaviors and the measurement techniques that should be employed to capture each case. In particular, the features of a diffuse sound field are discussed. It is shown that this can give rise to a pressure response that may vary significantly over a small angle and frequency scale. Here a multipoint measurement, such as a spatial average or acoustic power, gives a more representative assessment than measuring just the acoustic sound pressure level at a single point. In addition, the fine detail of the response may be characterized by a spatially averaged correlation function, giving a measure of the diffusivity that is complementary to more standard measures of energy distribution.

11:05

4aAAa10. Computer modeling of a large fan-shaped auditorium. Heather Smith and Timothy Leishman (Dept. of Phys., Brigham Young Univ., N283 ESC, Provo, UT 84602, hm73@email.byu.edu)

A research project was recently undertaken to analyze the acoustical characteristics of a 21 000-seat fan-shaped auditorium. Careful geometric modeling of the hall has been a significant part of this study. Because of its size, shape, and other architectural features, computer modeling has presented some interesting challenges. For example, it has been shown experimentally that the concavely oriented rows of (moderately absorptive) seats produce significant scattering that aggregates toward the focal point of the hall. This paper will discuss how the seat scattering and scattering from other bodies have been included in the model. Other challenges in modeling a hall this size will also be discussed.

11:20

4aAAa11. Radiant exchange in partially specular architectural environments. C. Walter BeamerIV and Ralph T. Muehleisen (Dept. of Civil, Environ., and Architectural Eng., Univ. of Colorado, Boulder, CO 80309)

The radiant exchange method, also known as radiosity, was originally developed for thermal radiative heat transfer applications. Later it was used to model architectural lighting systems, and more recently it has been extended to model acoustic systems. While there are subtle differences in these applications, the basic method is based on solving a system of energy balance equations, and it is best applied to spaces with mainly diffuse reflecting surfaces. The obvious drawback to this method is that it is based around the assumption that all surfaces in the system are diffuse reflectors. Because almost all architectural systems have at least some partially specular reflecting surfaces in the system it is important to extend the radiant exchange method to deal with this type of surface reflection. [Work supported by NSF.]

11:35

4aAAa12. Citadel in Teotihuacan. Sergio Beristain (P.O. Box 12-1022, Col. Narvarte, 03001 Mexico, D.F. Mexico, sberista@hotmail.com)

Teotihuacan, the largest archaeological site nearby Mexico City, is also a place where traditions are maintained through some ceremonies on specific dates, by the Sun and Moon pyramids, and history telling by the pyramids in day to day light and sound shows. This enormous site has a large square in the south known as The Citadel (La Ciudadela), a place some 200×300 meters (m), surrounded by 2.2 m high pyramid basements, and two pyramids to the East (one in front of the main one dedicated to the good Quetzalcoatl). Near the center of this large square sits a 2.2 m basement 18×20 m where some special sound events (theatre, dance, music, etc.) are occasionally presented. Sound level measurements have proved that due to the site conditions, the sound level decreases 3–4 dB on the average per doubling distance, which makes it suitable for large audiences with the only problem of some minor echoes in small portions of the audience area.

THURSDAY MORNING, 13 NOVEMBER 2003 WEDGWOOD ROOM, 9:00 TO 10:05 A.M.

Session 4aAAb

Architectural Acoustics and Noise: Distinguished Lecture: Analysis of Community Response to Transportation Noise a Quarter Century After Schultz (1978)

Jack E. Randorff, Chair

Randorff and Associates, Inc., 11 West Canyon View Drive, Ransom Canyon, Texas 79366

Chair's Introduction—9:00

Invited Paper

9:05

4aAAb1. Analysis of community response to transportation noise a quarter century after Schultz (1978). Sanford Fidell (Fidell Assoc., Inc., 23139 Erwin St., Woodland Hills, CA 91367)

Transportation noise is a vexing and intrinsically controversial problem that has plagued societies since the beginnings of urban civilization. A function relating cumulative noise exposure to the prevalence of noise-induced annoyance [T. J. Schultz, ''Synthesis of social surveys on noise annoyance,'' J. Acoust. Soc. Am. **64**, 377–405 (1978)] is the foundation for contemporary analyses of transportation noise effects on communities. The expenditures of billions of dollars in airplane ticket and fuel taxes for the construction of an airport infrastructure and for the mitigation of noise impacts in the United States are governed by policies ostensibly supported by a successor to Schultz's original ''synthesis'' curve. Many have grown so comfortable with the last quarter century's paradigm for transportation noise assessment and regulation, however, that they have lost sight of its underpinnings and limitations. A review of the historical and modern states of the art identifies persistent unresolved problems in the prediction and explanation of community response to transportation noise that are not fully addressed by descriptive dosage-effect analysis.

Session 4aAAc

Architectural Acoustics and Noise: Progress in Measurement of Transportation Noise Since Ted Schultz

Sanford Fidell, Chair
Fidell Associates, Inc., 23139 Erwin Street, Woodland Hills, California 91367

Invited Papers

10:15

4aAAc1. Shoehorning in the jet age. Leo L. Beranek (975 Memorial Dr., Cambridge, MA 02138, beranekleo@ieee.org)

The introduction into commercial service of jet air transports preceded the codification of U.S. federal aviation noise regulatory policy by nearly two decades. The immediate problem that had to be solved to permit jet operations in 1958 at New York airports was to determine that the noisiness of jet airplane would be no greater than that of the largest propeller-driven aircraft then in operation. This paper describes the efforts made for the Port of New York Authority to meet and enforce that mandate. BBN made noise measurements of many propeller aircraft takeoffs in the community off the main runway at Idlewild at distances of 2.5 miles and greater from the start of take-off roll. Each measurement was associated with airplane type, gross weight, and altitude. The Boeing 707 was similarly measured at Boeing's airport. Relative annoyance judgments were made in the laboratory to establish equivalent noisiness of jet and propeller spectra, and ''Perceived Noise Levels'' were determined by a process similar to loudness calculations. Boeing was forced to equip the 707 with multi-tube mufflers and, and jet take-off procedures had to be modified to maintain equivalent perceived noise levels. The substantive findings and politics of these efforts are discussed.

10:40

4aAAc2. Origins and application of the European Union Position paper on dose response relationships between transportation noise and annoyance. Bernard F. Berry (Berry Environment Ltd.—BEL, Shepperton, UK)

Dose–response relationships of the sort pioneered by Schultz figure prominently in the current European noise regulation policy. A position paper developed by the Working Group on Dose-effects (part of the EU Expert Network) reviewed a range of potential health effects, but decided that annoyance and sleep disturbance remain the most prevalent and sensitive effects of transportation noise exposure, and those for which the best data were available. A Position paper providing guidance on the dose–effect relations to be used for the assessment of numbers of people annoyed by noise from transportation sources (rail, road and air) may be found at http://europa.eu.int/comm/environment/noise/home.htm This presentation explains the context in which the Paper was developed, outlines the process by which the dose–response relationships were derived and summarizes the key recommendations. Finally some observations are made as the Position Paper is being applied, and related future developments are discussed. The author acknowledges with deep gratitude the assistance of Dr. Henk Miedema of TNO in preparing this paper.

11:05

4aAAc3. Correcting DNL so it works—better. Paul Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821)

The day–night average sound level (DNL), first developed by the U.S. Environmental Protection Agency, is commonly used to quantify and assess environmental noise. A keystone to noise assessment is the dose–response relationship. However, the dose–response relationship is not an absolute; there is great scatter to the data on which it is based. In an attempt to reduce the scatter to the DNL data, the EPA suggested the use of normalized DNL. Normalized DNL is the basic DNL value with a number of adjustments added to account for specific characteristics and factors of the sound. This paper reviews and analyze the concepts inherent in normalized DNL and provide an updated set of normalization factors that can reduce the scatter to dose–response relationships. Several of these normalizations are contained in the new ISO 1996-1:2003.

11:30–12:00

Panel Discussion

Session 4aAB

Animal Bioacoustics: Neurobiology of Communications

George D. Pollak, Chair

Section of Neurobiology, University of Texas, Austin, Texas 78705

Invited Papers

8:30

4aAB1. Audiomotor integration for active sensing in the echolocating bat, *Eptesicus fuscus*. Cynthia F. Moss, Shiva R. Sinha, and Kaushik Ghose (Neurosci. and Cognit. Sci. Prog., Inst. for Systems Res., Univ. of Maryland, College Park, MD 20742)

Echolocating bats probe the environment with sonar signals that change as they seek, pursue and intercept insect prey on the wing. Coordinating its sonar vocalizations with flight dynamics in response to changing echo information, the bat exhibits a dazzling display of sensorimotor integration. Our work aims at understanding the mechanisms supporting audiomotor integration for echolocation in the FM-bat, *Eptesicus fuscus*. Behavioral studies measure adaptive responses of free-flying bats engaged in complex spatial tasks. The directional aim of the bat's sonar beam and temporal patterning of cries provide explicit data on the motor commands that feed directly back to the auditory system for spatially-guided behavior. Neural studies focus on the superior colliculus (SC), a midbrain structure implicated in species-specific orienting behaviors. A population of SC neurons shows echo-delay tuning, a response property believed to play a role in target range coding. Microstimulation of the SC elicits head and pinna movements, along with sonar vocalizations. SC recordings from tethered, vocalizing bats reveal bursts of neural activity preceding each sonar cry. Collectively, these results suggest that the bat SC plays a functional role in the auditory information processing and orienting behaviors that operate together in echolocation. [Work supported by NSF, NIMH and Whitehall Foundation.]

9:00

4aAB2. Processing and representation of social communication sounds in the brainstem auditory system of bats. George D. Pollak (Section of Neurobiology, Univ. of Texas, Austin, TX 78712, gpollak@mail.utexas.edu)

While bats are best known for their abilities to orient and capture prey via echolocation, they are also highly social animals who use a rich repertoire of species-specific sounds for social communication. This talk explores how communication signals are progressively transformed and represented in the ascending auditory system. One principal transformation that distinguishes the inferior colliculus from lower nuclei is a change from processing that emphasizes response homogeneity among the neuronal population in each lower nucleus, to one that emphasizes heterogeneity and selectivity in the inferior colliculus. Collicular neurons are selective in that each neuron fails to respond to some, or even all calls, even though those calls have energy that encroaches upon their excitatory response regions, and are heterogeneous since each collicular neuron responds to a different subset of calls. The transformation from homogeneity to heterogeneity may largely be a consequence of the difference in the ways that the various excitatory and inhibitory inputs distribute along frequency contours in lower nuclei compared to the inferior colliculus. One important consequence is that those features endow the population in the inferior colliculus with the ability to respond to any signal with a unique and pronounced spatiotemporal pattern of activity. [Work supported by NIH Grant No. DC 00268.]

Contributed Papers

9:30

4aAB3. Computational model of modulation detection by the bullfrog. Andrea M. Simmons and Kyler Eastman (Depts. of Psych. and Neurosci., Brown Univ., Providence, RI 02912, Andrea_Simmons@brown.edu)

Bullfrog eighth-nerve fibers operate as envelope detectors, showing significant phase-locking to amplitude modulation (AM) rates as high as 800 Hz. A computational model that estimates stimulus period from all-order interval histograms (autocorrelation functions) aligned in time across frequency channels mimics fiber responses. The use of autocorrelation implies a mechanism based on delay lines or neural coincidence detection. The goal of this study was to determine the relevance of such a model in predicting responses of neurons in bullfrogs auditory midbrain (torus semicircularis) to AM stimuli. Modeled output of peripheral fibers was passed through a series of delay lines varying in latency, and then compared with actual midbrain data. There is a great diversity in representation of AM in the auditory midbrain, and this diversity is related to response latency and recording location. Neural responses from the cell sparse zone in the caudal midbrain were well-matched by a delay line mechanism, although the model was poorer in predicting response properties of other midbrain areas. [Work supported by NIH and the Brown University Brain Science Program.]

9:45

4aAB4. Otoacoustic emissions measured in rhesus monkeys (*Macaca mulatta*). Dennis McFadden, Edward G. Pasanen (Dept. of Psych., Univ. of Texas, Austin, TX 78712-0187, mcfadden@psy.utexas.edu), Jessica Raper, and Kim Wallen (Emory Univ., Atlanta, GA 30322-2470)

In humans, otoacoustic emissions (OAEs) are stronger in females than in males and stronger in right ears than in left. The physiological bases for these differences are unknown, but several lines of circumstantial evidence suggest that the sex difference is attributable to androgenizing mechanisms operating during prenatal development. Specifically, it appears that exposure to high levels of androgens during prenatal development diminishes the strength of the cochlear amplifiers and thus the strength of the OAEs. Sex and ear differences in OAEs have not been well studied in species other than humans. Accordingly, click-evoked OAEs and distortion-product OAEs were measured in nine female and nine male rhesus monkeys. For CEOAEs, but less clearly for DPOAEs, females exhibited sig-

nificantly stronger OAEs than males. There was no consistent ear difference for either sex for either type of OAE. In order to better study the early components of the CEOAE waveform, a nonlinear procedure [Molenaar *et al.*, Hearing Res. **143**, 197–207 (2002)] was used to collect CEOAEs along with our standard (linear) procedure. This colony also contains animals of each sex that were treated with androgenic or antiandrogenic agents during prenatal development, and OAEs are also currently being measured on those animals. [Work supported by NIDCD.]

10:00

4aAB5. Otoacoustic emissions measured in spotted hyenas (*Crocuta crocuta*). Dennis McFadden, Edward G. Pasanen (Dept. of Psych., Univ. of Texas, Austin, TX 78712-0187, mcfadden@psy.utexas.edu), Mary L. Weldele, Stephen E. Glickman, and Ned J. Place (Univ. of California, Berkeley, CA 94720)

From birth, female spotted hyenas exhibit highly masculinized bodies and behaviors. Their external genitalia greatly resemble those of males, and they are behaviorally dominant over males. This marked masculinization raised the question of whether the otoacoustic emissions (OAEs) of female spotted hyenas also would be masculinized. Click-evoked OAEs were measured in six female and six male hyenas at two click levels. Also, distortion-product OAEs were measured at four or more primary levels in three frequency regions: 2, 3.5, and 5.0 kHz. Both CEOAEs and DPOAEs were strong in both sexes in spotted hyenas. In humans, both CEOAEs and DPOAEs are stronger in females than males and stronger in right ears than left. Unlike humans, both the CEOAEs and DPOAEs in female spotted hyenas were weaker than those in males, and unlike humans, OAEs were not stronger in right ears. The implication is that the same androgenizing processes that masculinize the body and behavior of female hyenas also masculinize those elements of the cochlea responsible for OAEs. That implication is being tested by measuring the OAEs of other hyenas in the Berkeley colony that were treated with antiandrogenic agents during fetal development. [Work supported by NIDCD.]

10:15–10:30 Break

10:30

4aAB6. Distortion product otoacoustic emissions provide clues to hearing mechanisms in the frog. Pantelis Vassilakis[a] (Dept. of Physiological Sci., Univ. of California, Los Angeles, CA 90095-1606, pantelis@acousticslab.com.) and Peter M. Narins (Univ. of California, Los Angeles, CA 90095-1606)

Cubic distortion product otoacoustic emissions (DPOAEs) were recorded from 10 *Rana pipiens* and 10 *Rana catesbeiana*, 5 males and 5 females each. The I/O curves obtained from the amphibian papilla (AP) of both species are very similar to the respective mammalian curves, indicating that, like in the mammalian cochlea, there may be an amplification process active in the frog AP. The DPOAE level dependence on primary levels is also similar to the mammalian case, suggesting a mechanical structure in the frog inner ear may be functioning analogously to the mammalian basilar membrane. DPOAE audiograms were obtained for primary frequencies spanning the animals hearing range and levels determined by the previous experiments. *R. catesbeiana* produce stronger emissions than *R. pipiens* and, consistent with previously reported sexual dimorphism in the mammalian and anuran auditory systems, females from both species produce stronger emissions than males. Additionally, the $2f_1-f_2$ DPOAE is generated primarily at the DPOAE frequency place, while the $2f_2-f_1$ DPOAE is generated primarily at a frequency place between the primaries. This difference in mammalian and frog DPOAEs may be linked to an anatomical difference that results in the acoustic energy following opposite paths through the mammalian and frog inner ears. [Work supported by NIH Grant No. DC-00222 to Peter M. Narins.]

[a]Currently at De Paul Univ., School of Music, Chicago, IL 60614.

10:45

4aAB7. Do goldfish miss the fundamental? Richard R. Fay (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626, rfay@luc.edu)

The perception of harmonic complexes was studied in goldfish using classical respiratory conditioning and a stimulus generalization paradigm. Groups of animals were initially conditioned to several harmonic complexes with a fundamental frequency (f_0) of 100 Hz. In some cases the f_0 component was present, and in other cases, the f_0 component was absent. After conditioning, animals were tested for generalization to novel harmonic complexes having different f_0's, some with f_0 present and some with f_0 absent. Generalization gradients always peaked at 100 Hz, indicating that the pitch value of the conditioning complexes was consistent with the f_0, whether or not f_0 was present in the conditioning or test complexes. Thus, goldfish do not miss the fundmental with respect to a pitch-like perceptual dimension. However, generalization gradients tended to have different skirt slopes for the f_0-present and f_0-absent conditioning and test stimuli. This suggests that goldfish distinguish between f_0 present/absent stimuli, probably on the basis of a timbre-like perceptual dimension. These and other results demonstrate that goldfish respond to complex sounds as if they possessed perceptual dimensions similar to pitch and timbre as defined for human and other vertebrate listeners. [Work supported by NIH/NIDCD.]

11:00

4aAB8. Low-frequency vocalizations in the Florida manatee (*Trichechus manatus latirostris*). Katherine Frisch (Florida Marine Res. Inst., 100 8th Ave. SE, St. Petersburg, FL 33701) and Stefan Frisch (Univ. of South Florida, Tampa, FL 33620)

Vocalizations produced by Florida manatees (*Trichechus manatus latirostris*) have been characterized as being of relatively high frequency, with fundamental tones ranging from 2500–5000 Hz. These sounds have been variously described as squeaks, squeals, and chirps. Vocalizations below 500 Hz have not been previously reported. Two captive-born Florida manatees were recorded at Mote Marine Laboratory in Sarasota, Florida. The analysis of these vocalizations provides evidence of a new category of low-frequency sounds produced by manatees. These sounds are often heard in conjunction with higher-frequency vocalizations. The low-frequency vocalizations are relatively brief and of low amplitude. These vocalizations are perceived as a series of impulses rather than a low-frequency periodic tone. Knowledge of these low-frequency vocalizations could be useful to those developing future management strategies. Interest has recently increased in the development of acoustic detection and deterrence devices to reduce the number of manatee watercraft interactions. The design of appropriate devices must take into account the apparent ability of manatees to perceive and produce sounds of both high and low frequency. It is also important to consider the possibility that acoustic deterrence devices may disrupt the potentially communicative frequencies of manatee vocalizations.

11:15

4aAB9. A dolphin lower jaw is a hydroacoustic antenna of the traveling wave. Vyacheslav A. Ryabov (Karadag Natural Reserve, Natl. Acad. of Sci. of Ukraine, Kurortnoe, Feodosia 98188, Crimea, Ukraine)

The purpose of the work is the analysis of a possible function of mental foramens as channels through which the echo passes in the lower jaw fat body and the determination of a role of channels and a skull in formation of the directivity of the dolphin echolocation hearing. Concrete problems were studying of the lower jaw morphology, modeling and calculation of a dolphin, *tursiops truncatus* p., echolocation hearing beam pattern. The outcomes of the work indicate those morphological structures of the lower jaw; the left and right half represents two hydroacoustic receiving antennas of the traveling wave type, TWA farther. The mental foramens of a dolphin lower jaw represent nonequidistant array of waveguide delay lines, and determine the phase and amplitude distribution of each of the antenna's array. The beam pattern of the echolocation hearing was calculated with the usage of the TWA model, and the allowance of flat sound wave diffraction. The beam pattern shape is naturally determined by the echolocation hearing functionality. It is equally well adapted both for echolocation and for pulses echo detection. A steepness of the bearing characteristic is estimated; it reaches 0.7 dB per degree.

11:30

4aAB10. Florida manatee avoidance technology: A pilot program by the Florida Fish and Wildlife Conservation Commission. Katherine Frisch and Elsa Haubold (Florida Marine Res. Inst., 100 8th Ave. SE, St. Petersburg, FL 33701)

Since 1976, approximately 25% of the annual Florida manatee (*Trichechus manatus latirostris*) mortality has been attributed to collisions with watercraft. In 2001, the Florida Legislature appropriated $200,000 in funds for research projects using technological solutions to directly address the problem of collisions between manatees and watercraft. The Florida Fish & Wildlife Conservation Commission initially funded seven projects for the first two fiscal years. The selected proposals were designed to explore technology that had not previously been applied to the manatee/boat collision problem and included many acoustic concepts related to voice recognition, sonar, and an alerting device to be put on boats to warn manatees. The most promising results to date are from projects employing voice-recognition techniques to identify manatee vocalizations and warn boaters of the manatees' presence. Sonar technology, much like that used in fish finders, is promising but has met with regulatory problems regarding permitting and remains to be tested, as has the manatee-alerting device. The state of Florida found results of the initial years of funding compelling and plans to fund further manatee avoidance technology research in a continued effort to mitigate the problem of manatee/boat collisions.

11:45

4aAB11. Massive gas insufflation without effect on esophageal reflectometry profiles. David T. Raphael, Dimiter Arnaudov, and Maxim Benbassat (Dept. of Anesthesiol., Keck School of Medicine, Univ. of Southern California Med. Ctr., Los Angeles, CA 90033, draphael@usc.edu)

Time–domain acoustic reflectometry generates a "one-dimensional" image of the interior of a cavity in the form of an area–distance profile. After patient intubation with a breathing tube, the characteristic reflectometry profile consists of a constant-area segment corresponding to the length of the tube, followed either by a rapid increase in the area beyond the carina (lung) or by a sudden decrease in the area to zero (esophagus). In the cardiac arrest setting, during mistaken placement of the breathing tube into the esophagus, followed by aggressive manual ventilation, is it possible to markedly distend the esophagus, such that the esophageal profile looks like a tracheal profile? With approval of the USC IUCAC Committee, an animal study was conducted with anesthetized, tracheally intubated, and mechanically ventilated dogs. With a separate breathing tube in the esophagus, aggressive esophageal ventilation (comparable to that seen in the cardiopulmonary resuscitation setting) was accomplished with a manual resuscitation bag. A Benson Hood Labs two-microphone reflectometer was used to obtain esophageal profiles with and without the above ventilation. In this pilot study, there was no significant esophageal distention as a result of the above ventilation. [Research supported by the Alfred E. Mann Institute.]

THURSDAY MORNING, 13 NOVEMBER 2003 — TRINITY B ROOM, 8:00 TO 11:45 A.M.

Session 4aPA

Physical Acoustics and Biomedical Ultrasound/Bioresponse to Vibration: Special Session on Nonlinear Acoustics in Honor of David Blackstock

Mark F. Hamilton, Cochair
Department of Mechnical Engineering, University of Texas–Austin, Austin, Texas 78712-1063

F. Michael Pestorius, Cochair
International Field Office Europe, Office of Naval Research, PSC 802 Box 39, APO/FPO, NY, AE 09499-0700

Invited Papers

8:00

4aPA1. David Blackstock and nonlinear acoustics at UT Austin. Mark F. Hamilton (Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712-1063)

Except for a period of 15 years, from when he left Austin (with B.S. and M.S. degrees in physics) for active duty in the USAF until he returned for a "year's visit" to UT in 1969, Austin has been home to David Blackstock. In the interim he developed hearing protectors for the Air Force, earned a Ph.D. in applied physics from Harvard, conducted research in nonlinear acoustics for industry, and joined the Electrical Engineering faculty at University of Rochester. Following his return, Austin rapidly became known as a home for nonlinear acoustics. This presentation will review the development of nonlinear acoustics at UT, placed in context of the underwater acoustics activities at ARL and the academic programs under David both at ARL and on campus.

8:15

4aPA2. Soap bubbles, sparks, sonic booms, spikes, and David Blackstock. Allan D. Pierce (Boston Univ., Boston, MA 02215, adp@bu.edu)

In the late 1960s people were puzzled by some anomalous features in the measured waveforms of sonic booms recorded at the ground during flyovers of supersonic aircraft. The waveforms were supposed to look like the letter N, but that was not always the case. Sometimes there were strange upward reaching spikes just behind the leading and trailing shocks. There were a lot of explanations

4a THU. AM

kicking around, and the present author had the idea that it was somehow caused by imperfect focusing of wavefonts after traveling through regions of higher sound velocity and discussed this with David Blackstock, who made the characteristic remark, ''Let's talk experiment.'' What resulted was an experiment that would have been worthy of Lord Rayleigh. The paper that reported this, titled ''Measurements of the Refraction and Diffraction of a Short *N*-Wave by a Gas-Filled Soap Bubble,'' appeared in J. Acoust. Soc. Am. in March 1971. Besides really nailing down the physical phenomena responsible for the spikes on sonic boom waveforms, it illustrated a wealth of physical concepts and experimental techniques. The present talk discusses the background of the paper, the physics that it used, and the influence it had on subsequent research.

8:30

4aPA3. Weak sparks, *N* waves, and the calibration of microphones at ultrasonic frequencies. Wayne M. Wright (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

A fortuitous situation involving the availability of wide-band condenser microphones, curiosity about acoustic waves generated by sparks, and his own development of nonlinear acoustic theory provided two significant contributions to David Blackstock's continuing research. Theoretical understanding of the nonlinear distortion experienced by intense acoustic impulses in air, such as those generated by relatively weak sparks, led to a technique for calibration of microphones that have essentially flat frequency response up to almost 1 MHz. These sparks and calibrated microphones, in turn, provided the means to carry out several graduate student thesis projects that were supervised by David. Diffraction at edges, apertures, and discs was studied in the time domain. Model studies related to sonic boom phenomena were carried out, as was the propagation of *N* waves in waveguides. The most recent application has been a model study of focusing by an ellipsoidal reflector, such as is typically used in lithotripsy. In each case, the various contributions to the received signals were readily identified in the time-domain presentation.

8:45

4aPA4. Application of nonlinear acoustics in the atmosphere and biomedicine. Robin O. Cleveland (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215, robinc@bu.edu)

David Blackstock has made important contributions to understanding nonlinear processes in underwater acoustics, atmospheric acoustics, and biomedical acoustics. In this talk, two of these areas that David has been involved with are addressed by contrasting the evolution of sonic booms in an inhomogeneous atmosphere with the evolution of shock waves in lithotripsy (breaking of kidney stones by shock waves). The distortion of *N* waves generated by supersonic aircraft is affected by geometrical spreading and changes in the acoustic properties of the medium with altitude. In shock wave lithotripsy, the distortion of triangle waves is affected by geometrical spreading and changes in the acoustic properties due to layering of tissue types. In addition, the turbulent boundary layer near the ground and the inhomogeneous nature of tissue, means that both sonic booms and lithotripter shock waves pass through a random media. Calculations are presented using weak shock to compare the distortion that occurs in the two scenarios. Measurements of saturationlike effects are shown for a clinical shock wave therapy device. Numerical simulations for propagation through random media demonstrate the localized focusing and defocusing that can occur for both sonic booms and lithotripter shock waves. [Work supported in part by NIH and NASA.]

9:00

4aPA5. Statistical nonlinear acoustics: First studies and current state. Oleg V. Rudenko (Faculty of Phys., Moscow State Univ., 119992 Moscow, Russia, rudenko@acs366.phys.msu.ru)

Statistical physics of high-intensity noise waves has, in principle, a long-term history. Nonlinear interactions between random and regular waves studied in the 1930s were devoted to sound absorption in solids caused by energy transfer from coherent phonons (signal) to Debye's thermal quantum (noise). However, exhaustive studies of nonequilibrium nonlinear random wave processes have been carried out only in the 1970s, after experimental and theoretical nonlinear acoustics reached an advanced stage of development. David Blackstock, one of the founders of modern nonlinear acoustics, performed pioneering experiments on broadband noise spectra distortion and signal–noise interactions in tubes and outdoors. These experiments confirmed existing data and formed the fundamental basis for both future studies and applications. Along with temporal statistics, spatial ones were considered by D. T. Blackstock during his studies of nonlinear diffraction caused by orifices and screens with irregular edges. These results are also of great significance for applications. A brief overview of the first studies in statistical nonlinear acoustics is given, as well as of new results obtained in recent years.

9:15

4aPA6. Propagation of finite amplitude sound through turbulence. Bart Lipkens (Mech. Eng. Dept., Western New England College, 1215 Wilbraham Rd., Springfield, MA 01119) and Philippe Blanc-Benon (Ctr. Acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, 69134 Ecully Cedex, France)

Pestorius and Blackstock [''Propagation of finite amplitude noise,'' Finite-amplitude wave effects in fluids, Proceedings of the 1973 International Symposium on Nonlinear Acoustics, edited by L. J. Bjørnø, 1973] developed a method to simulate the propagation of finite amplitude noise in a tube. The method separates the effects of nonlinear distortion and absorption and diffraction. The nonlinear steepening of an arbitrary waveform is performed in the time domain. The absorption and diffraction effects are implemented in the frequency domain. This method has been used widely for finite amplitude wave propagation. Lipkens and Blanc-Benon [''Propagation of finite-amplitude sound through turbulence: a geometrical acoustics approach,'' C. R. Acad. Sci. Paris, Ser. II b **320**, 477–484 (1995)] used the Pestorius model to simulate the propagation of *N* waves through turbulence. Linear geometrical acoustics is used to trace rays through individual realizations of a turbulent field. A nonlinear transport equation describes the propagation of sound through turbulence. The equation is solved using the Pestorius algorithm. The results indicate that the equivalent nonlinear distortion after propagation through turbulence is always less than that for the homogeneous case. In addition, the effect of a random velocity field is more pronounced that that of a random temperature field.

9:30

4aPA7. Aeroacoustics, moving boundaries, and bursting balloons: Acoustic sources revisited. Christopher L. Morfey (Inst. of Sound and Vib. Res, Univ. of Southampton, Southampton SO17 1BJ, UK, clm@isvr.soton.ac.uk)

The use of equivalent acoustic sources to describe scattering in a nonuniform medium dates back to Rayleigh's theory of sound. The idea of equivalent sources in a uniform medium at rest was later developed by Lighthill into his ''acoustic analogy,'' capable of describing the generation of sound by turbulence and other vortical flows. In the present paper Lighthill's acoustic analogy formulation is generalized to encompass initial-value problems; the initial conditions are represented by impulsive sources and dipoles distributed over the domain, and boundary conditions are represented in the usual manner by surface sources and dipoles. David Blackstock's bursting balloon example, discussed in Chapter 3 of *Fundamentals of Physical Acoustics*, can be solved by this method. However, in situations where the medium is of nonuniform density (for example, a gas with a specified temperature distribution at the initial time), the impulsive source distribution obtained by a direct application of time windowing to the acoustic analogy is nonphysical. The apparent paradox is resolved by introducing the energy conservation equation, and reformulating the acoustic analogy with pressure, rather than density, as the wave variable.

9:45

4aPA8. Diffraction of nonlinear acoustic waves: Matched-area method. Lev A. Ostrovsky (Zel Technologies/NOAA ETL, 325 Broadway, Boulder, CO 80305)

This is a brief overview of a series of investigations undertaken by the author together with his collaborators at the Institute of Applied Physics, Russian Academy of Science. In these works, a simplified theory was developed for the acoustic waves which profile is distorted due to both nonlinearity and diffraction. The theory is based on an often existing possibility to separate the regions of nonlinear geometrical acoustics and linear diffraction and match the corresponding solutions in an intermediate area(s) where neither effect is cumulative. The problems addressed on this way include: (i) broadening of the beam pattern of an intensive acoustic radiation (with V. Fridman); (ii) behavior of a nonlinear acoustic wave near a caustic (with E. Pelinovsky and V. Fridman); (iii) radiation and focusing of nonlinear waves (with A. Sutin). In a number of cases, theoretical results were corroborated by comparison with the experimental data obtained by D. Blackstock and other researchers. This approach is believed to be an efficient tool for treating many realistic situations, and to deserve further development. This work benefited greatly from numerous contacts with Professor Blackstock, starting in the early stages of the investigation.

10:00–10:15 Break

10:15

4aPA9. From bench to bedside: Nonlinear bio-ultrasound. Kevin J. Parker (School of Eng. and Appl. Sci., Univ. of Rochester, Lattimore Hall 309, RC Box 270076, Rochester, NY 14627-0076)

In the early 1960s, David Blackstock developed the weak shock theory which characterized the development and evolution of harmonics in a finite-amplitude propagating plane wave. Later work developed an understanding of beam patterns of the fundamental and higher harmonics from piston radiators. All of these concepts were germane to the field of bio-ultrasound. However, nonlinear effects were largely unexploited in bio-ultrasound until the 1980s, when the role of finite-amplitude mechanisms was recognized as an enhancer of heating and lesion production in high intensity applications. Later, in the 1990s, the role and the advantage of higher harmonics in ultrasonic imaging were developed. In particular, the higher harmonics generated by nonlinear effects, especially in the presence of aberrating tissue, were shown to produce dramatic increases in the resolution and contrast of ultrasound images in cardiology and other applications. Today, every high-end medical ultrasound scanner has the capability of ''harmonic imaging'' for improved image quality in a number of applications. The physics and technology of this remarkable development will be reviewed, along with the important guiding role of David Blackstock and his collaborators, particularly at UT Austin and at U. Rochester.

10:30

4aPA10. Cavitation in shock wave lithotripsy. Michael R. Bailey, Lawrence A. Crum, Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Seattle, WA 98105), Andrew P. Evan, James A. McAteer (Indiana Univ. School of Anatomy, Indianapolis, IN), Tim Colonius (California Inst. of Technol., Pasadena, CA), and Robin O. Cleveland (Boston Univ., Boston, MA)

A case is presented for the important role of cavitation in stone comminution and tissue injury in shock wave lithotripsy (SWL). Confocal hydrophones and a coincidence algorithm were used to detect cavitation in kidney parenchyma. Elevated hydrostatic pressure dissolved cavitation nuclei and suppressed cell injury and stone comminution *in vitro*. A low-insertion-loss, thin, mylar film nearly eliminated stone erosion and crack formation only when in direct contact with the stone. This result indicates not only that cavitation is important in both cracking and erosion but also that bubbles act at the surface. Time inversion of the shock wave by use of a pressure-release reflector reduced the calculated pressure at bubble collapse and the measured depth of bubble-induced pits in aluminum. Correspondingly tissue injury *in vivo* was nearly eliminated. Cavitation was localized and intensified by the use of synchronously triggered, facing lithotripters. This dual pulse lithotripter enhanced comminution at its focus and reduced lysis in surrounding blood samples. The enhancement of comminution was lost when stones were placed in glycerol, which retarded bubble implosion. Thus, cavitation is important in comminution and injury and can be controlled to optimize efficacy and safety. [Work supported by NIH DK43381, DK55674, and FIRCA.]

10:45

4aPA11. Nonlinear and linear wave equations for propagation in media with frequency power law losses. Thomas L. Szabo (Boston Univ., 110 Cummington St., Boston, MA 02215, tlszabo@bu.edu)

The Burgers, KZK, and Westervelt wave equations used for simulating wave propagation in nonlinear media are based on absorption that has a quadratic dependence on frequency. Unfortunately, most lossy media, such as tissue, follow a more general frequency power law. The authors first research involved measurements of loss and dispersion associated with a modification to Blackstock's solution to the linear thermoviscous wave equation [J. Acoust. Soc. Am. **41**, 1312 (1967)]. A second paper by Blackstock [J. Acoust. Soc. Am. **77**, 2050 (1985)] showed the loss term in the Burgers equation for plane waves could be modified for other known instances of loss. The authors' work eventually led to comprehensive time-domain convolutional operators that accounted for both dispersion and general frequency power law absorption [Szabo, J. Acoust. Soc. Am. **96**, 491 (1994)]. Versions of appropriate loss terms were developed to extend the standard three nonlinear wave equations to these more general losses. Extensive experimental data has verified the predicted phase velocity dispersion for different power exponents for the linear case. Other groups are now working on methods suitable for solving wave equations numerically for these types of loss directly in the time domain for both linear and nonlinear media.

11:00

4aPA12. The profile of a weak shock wave in an unconsolidated medium. Konstantin Naugolnykh (Univ. of Colorado/Zeltech, 325 Broadway, Boulder, CO 80305, konstantin.naugolnykh@noaa.gov)

The propagation of a high intensity sound wave is determined mainly by the nonlinear and dissipative effects. If the initially sinusoidal wave is intensive, the steepness of the wave fronts increases, resulting in the occurrence of a discontinuity in each period of the wave. On the other hand, the influence of dissipative processes tends to smooth the wave profile, diminishing the gradients of velocity and temperature. Consequently, during the propagation of the intense wave its profile is forming as a result of the balance of nonlinear and dissipative effects. Many aspects of the high intensity sound wave evolution were described in the set of papers of D. T. Blackstock. The standard nonlinear medium with viscosity and thermal conductivity was considered. However, later, M. J. Buckingham has developed the theory of sound propagation in saturated marine sediments which includes a new dissipation term representing internal losses arising from interparticle contacts. The balance of nonlinearity and dissipation in such a medium has specific features that are considered in the present paper.

11:15

4aPA13. A brief history of the nonlinear acoustics of rocks. James A. TenCate (Earth and Environ. Sci., Los Alamos Natl. Lab, Los Alamos, NM 87545, tencate@lanl.gov)

Much of the early measurements done to study the nonlinearity of rocks in the late 1980s were analogs to many of the classic nonlinear acoustics experiments performed by students working under the watchful eye of David Blackstock. However, it soon became apparent that nonlinear waves in rocks did not behave as expected. Wave propagation measurements in a long waveguide (a sandstone core) did not generate the usual harmonic dependencies. The nonlinear resonance of a long thin bar of sandstone did not look at all like the nonlinear resonance of a tube of air. A host of other experiments produced equally puzzling behavior. In general, waves in the rock experience considerable nonlinear distortion, exhibit peculiar hysteresis, have memory, and have confounded researchers looking for a tidy theory to describe them. Moreover, it has recently been discovered that rocks are but one member of a larger class of materials—most all of which are granular—which all exhibit similar behavior. We describe all these experiments and how the results drove us away from classical nonlinear acoustics to new theoretical descriptions and applications. [Work supported by Office of Basic Energy Sciences, Geosciences.]

11:30

4aPA14. Evaluating prediction methods for the spectral evolution of finite-amplitude jet noise. Kent L. Gee and Victor W. Sparrow (Grad. Prog. in Acoust., The Penn State Univ., 217 Appl. Sci. Bldg., University Park, PA 16802, kentgee@psu.edu)

David Blackstock has made substantial contributions to the understanding of the propagation of broadband, finite-amplitude noise. His work with Pestorius, Theobald, Webster, and Menounou, among others, has given us an understanding of much of the underlying physics. Even so, there are still gaps in our knowledge of the noise propagation from supersonic jet flows. A recent study was undertaken to compare several methods for the far field progression of a finite-amplitude broadband spectrum. This paper will provide initial results for that comparison. For each method tested, a Gaussian spectrum and experimental jet noise data were utilized as inputs. [Work supported by Strategic Environmental Research and Development Program.]

Session 4aSA

Structural Acoustics and Vibration: Vibrations in Mechanical Systems

Jeffrey E. Boisvert, Chair
Naval Undersea Warfare Center Division Newport, Code 2133, Newport, Rhode Island 02841

Contributed Papers

9:00

4aSA1. Predicting folded beam waveguide absorber behavior using full translational and rotational degree of freedom coupling. Carl Pray, Robert Campbell, Stephen Hambric, and Andrew Munro (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804, cpray@psu.edu)

Folded beam waveguide absorbers (WGAs) have been shown to be effective low-frequency damping devices. Early WGA studies were unable to accurately predict this damping behavior. These studies used only translational degrees of freedom (DOFs), which resulted in the underestimation of the WGA damping performance. A recent study [Munro and Hambric, ''Modeling folded beam waveguide absorber behavior using translational and rotational degree of freedom frequency response function coupling,'' Proc. NOISE-CON 2003] used translational and rotational DOF frequency response functions to predict folded beam WGA behavior when attached to a thick rectangular plate, where the plate and WGA rotational DOFs were estimated using the finite-differencing method. Each plate and WGA DOF was coupled independently using frequency domain substructure synthesis (FDSS) [Jetmundsen *et al.*, ''Generalized frequency domain synthesis,'' J. Am. Helicopter Soc. 55–64, Jan (1988)], and the damping contributions due to each DOF were summed to give the total WGA damping prediction. This method gives a much improved damping estimate from previous methods but is inefficient for complex problems. In this study, all the DOFs for the plate and WGA are combined simultaneously using FDSS to predict the WGA damping behavior and plate response with folded beam WGAs attached.

9:15

4aSA2. The temporal and spatial effects of a magnetorheological elastomer in squeeze mode. Anne-Marie Albanese and Kenneth Cunefare (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332)

The behavior of a magnetorheological (MR) elastomer in an adaptive tuned vibration absorber (ATVA) subjected to transient magnetic fields was examined. ATVAs suppress vibration under broadband, variable, and multiple frequency excitations, and are effective because they can rapidly change one or more parameters to retune in very short periods of time. The device examined here features a MR elastomer whose stiffness and static displacement length are both sensitive to magnetic fields; this means that the static displacement transient behavior directly corresponded to the stiffness transient behavior. A magnetic field applied at step intervals was applied to the ATVA, and the static displacement length change and flux density levels through the ATVA were measured as a function of time. For a flux density level change of 0.12 Tesla, a static displacement length change of about 0.4 mm, which amounted to a 4% change in the overall spring length, occurred in 65 milliseconds. The cause of the long transient response is believed to be due to the magnetic–mechanical coupling.

9:30

4aSA3. Prediction of damped circular cylindrical shell vibration using energy flow analysis. Jacob Klos (Structural Acoust. Branch, M.S. 463, NASA Langley Res. Ctr., Hampton, VA 23681, j.klos@larc.nasa.gov) and Robert J. Bernhard (Purdue Univ., West Lafayette, IN 47907)

Approximate energy prediction tools are gaining popularity in the field of engineering noise control because of the need to quickly predict the high frequency dynamic response of complex systems. One approximate energy prediction tool commonly used is statistical energy analysis. However, many problems encountered in the field of engineering noise control do not satisfy the assumptions of lightly damped, lightly coupled systems. A second method, energy flow analysis, has been proposed for the case of moderate damping and coupling. The application of energy flow analysis to plates and beams is well documented in the literature. However, applications of energy flow analysis to shells of curvature are limited and need to be addressed. Two approaches for modeling radial vibration of damped circular cylindrical shells using energy flow analysis are proposed and verified in this presentation. The formulations of the solutions are discussed. Qualitative and quantitative comparisons of the responses predicted by the energy flow models to analytical predictions are made. From the comparisons, it is concluded that energy flow analysis provides a valid method to model radial vibration of circular cylindrical shells subjected to a radial excitation both above and below the ring frequency of the cylinder.

9:45

4aSA4. Are the energy analysis (EA) and the statistical energy analysis (SEA) compatible? G. Maidanik and K. J. Becker (Carderock Div., Naval Surface Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817)

The statistical energy analysis (SEA) is being used commercially to find ways of achieving noise control on vehicles of various types. A tenet of SEA states that the modal energy stored in a force-driven (master) dynamic system may not exceed the modal energy stored in a passive (adjunct) dynamic system to which it is coupled. Some recent computations using an energy analysis (EA) of generic coupled dynamic systems shows that this rule may not be obeyed in (EA). To render (EA) compatible with (SEA) a lower limit on the value of the modal overlap parameter in the adjunct dynamic system must be imposed. The value must exceed a threshold for these two analyses to be compatible. [Work supported by ONR.]

10:00

4aSA5. Vibration of in-vacuo elliptic cylindrical shells. Jeffrey E. Boisvert (Naval Undersea Warfare Ctr. Div. Newport, Newport, RI 02841, boisvertje@npt.nuwc.navy.mil) and Sabih I. Hayek (Penn State Univ., University Park, PA 16802)

The equations of motion for the vibration of elliptic cylindrical shells of constant thickness were derived using a Galerkin approach. The elastic strain energy density used in this derivation has seven independent kinematic variables: three displacements, two thickness-shear, and two thickness-stretch. The resulting seven coupled algebraic equations are symmetric and positive definite. The shell has a constant thickness, h,

finite length, L, and is simply supported at its ends, $(z=0,L)$, where z is the axial coordinate. The elliptic cross-section is defined by the shape parameter, a, and the half-length of the major axis, l. The modal solutions are expanded in a doubly infinite series of comparison functions in terms of circular functions in the angular and axial coordinates. The natural frequencies and the mode shapes were obtained by the Galerkin method. Numerical results were obtained for several h/l and L/l ratios, and various shape parameters, including the limiting case of a simply supported cylindrical shell $(a=100)$. [Work supported by ONR and the Navy/ASEE Summer Faculty Program.]

10:15–10:30 Break

10:30

4aSA6. The effect of bearing properties on the eigenvalues of a rotordynamic system. Jerry H. Ginsberg and Benjamin B. Wagner (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, jerry.ginsberg@me.gatech.edu)

Natural frequencies and associated modal damping ratios are important in the diagnosis of rotating machinery problems. This work examines how the modal properties of a rotating shaft/disk system are affected by changes in rotation rate, lubricant viscosity, and bearing clearance. The system under analysis is a uniform, elastic, rotating shaft with a single, rigid disk concentrically mounted to the shaft away from mid-span. The shaft is supported by short-length, plain journal bearings. Standard lubrication theory is used to generate the stiffness and damping matrices for the bearings. The clearance and lubricant viscosity are independently adjustable at each bearing. A Ritz series expansion is used to generate the mass, stiffness, and gyroscopic matrices describing the shaft and disk. The combined action of the bearings and shaft/disk system is represented by stationary and rotational inertia, stiffness, and internal and external damping. A nonsymmetric generalized eigenvalue problem solver is used to calculate system eigenvalues from the system matrices over the speed range of interest. The behavior of the system is quantified in terms of dependence of the real and imaginary parts of system eigenvalues on the rotation rate. The real part of the eigenvalue is proportional to the modal damping ratio, and the imaginary part is the natural frequency. The analysis results include plots of natural frequency and damping ratio versus mode for varying bearing clearance and viscosity.

10:45

4aSA7. Effect of modal overlap factor on the ray tracing analysis of the curved beam structure: A computational study. Cheol-Ho Jeong and Jeong-Guon Ih (NOVIC, Dept. of Mech. Eng., Korea Adv. Inst. of Sci. and Technol., KAIST, Daejeon, Korea)

Ray tracing method (RTM) of the geometrical acoustics area is recently known to be useful for the evaluation of spatial distribution of energy density and power flow of vibrating structures at high frequencies. In this study, the spatial distribution of vibrational energy and power flow of curved beam and its connected structures are of interest. Longitudinal, flexural, and torsional waves are considered using the Euler–Bernoulli beam theory. In the analysis of spatial distribution, the band-averaged RTM result varies smoothly in space without local fluctuations. This means that the RTM result can represent a similar tendency to the traveling wave solution. An accuracy of the analysis depends on the modal overlap factor (MOF), which is the function of the length of the beam, frequency range, and structural loss factor. The effect of changing each parameter is investigated and discussed. Similar to other high frequency methods such as statistical energy analysis and power flow analysis, the results become close to the traveling wave solutions as the modal overlap factor increases. [Work supported by the BK21 project and NRL.]

11:00

4aSA8. Sonic inspection of concrete bridge decks. R. Daniel Costley, Gary Boudreaux (Miltec Missiles and Space Co.), and William Gene Ramsey (DIAL, Mississippi State Univ.)

One technique for determining the integrity of concrete structures, such as bridge decks, involves dragging a chain across it and listening to the audible response. A distinctive, hollow sound is produced when a chain is dragged over a section of concrete containing a delamination. This technique has been automated by recording the sound produced by a dragging chain with a suitable microphone and processing these signals with a minicomputer to distinguish between ''good'' and ''bad'' sections of concrete. The equipment is mounted on a hand-pushed cart with chains attached so that they drag along the surface of the deck. In addition, the microphone is mounted in such a way, using standard noise control techniques, so that external noise is minimized. Traffic noise is filtered electronically. These improvements make the technique operator independent and allow inspections to be made in noisy environments. Another advantage is that this approach produces an objective record of the inspection, available both electronically and in hardcopy. These records can be compared to past and future inspections, allowing the inspectors to monitor the health of the structure. Results from bridge deck inspections will be presented, along with a description of the device and the signal processing techniques.

11:15

4aSA9. Landmine vibration modes. Dimitri Donskoy, Andrei Zagrai, and Alexander Ekimov (Stevens Inst. of Technol., Davidson Lab., Hoboken, NJ 07030, ddonskoy@stevens-tech.edu)

Recent experimental and theoretical studies in the field of seismo-acoustic landmine detection proved high potential of this technique. Investigations have also demonstrated that acoustic detection and discrimination of landmines is a complex problem dependent upon interaction between soil and buried mines and their respective properties. Vibration characteristics of a mines casing play a critical role in this problem. Our recent tests, for the first time, revealed the multi-modal vibrations of mines. It was observed that the resonance frequencies of the soil/mine system depend on the burial depth and soil conditions. The present study provides analysis of this phenomenon both experimentally and though physical modeling. The experimental investigations were focused on the effects of burial depth and soil moisture content on vibration response measured on the soil surface above the buried mine. The modeling efforts were based on the electro-mechanical analogy approach where each vibration mode was represented as an oscillator with its own effective parameters. The validity of this consideration was confirmed through comparison against conventional modal analysis approach. The study demonstrated good agreement between the developed model and experimental results.

11:30

4aSA10. Calculation of vibration reduction design for underground sources. Stanislav A. Kostarev (Lab. of Acoust. and Vib. Tunnel Assoc., 21 Sadovo-Spasskaya Str., Moscow 107217, Russia), Samuil A. Rybak (N. N. Andreev Acoust. Inst., Moscow 117036, Russia), and Sergey A. Makhortykh (Inst. of Mathematical Problems of Biol., RAS, Pushchino Moscow reg. 142290, Russia)

The problems of ecology situation control near intensive underground acoustical sources is considered. Studied vibration absorbers are modeled by multi-component system of connected oscillators with damping. Obtained oscillatory equation system was investigated numerically. Frequency dependencies of absorbers efficiency have been calculated. An influence of the physical-mechanical parameters of the surrounding ground on the value of vibration reduction has been determined. Some variants of principal realization of vibration absorbers for the case of underground railway are discussed. [Work supported by Russian Foundation for Basic Researches Grants Nos. 01-02-16127, 02-02-17143.]

Session 4aSC

Speech Communication: Issues in Similarity and Distinctiveness (Lecture/Poster Session)

Lynne E. Bernstein, Cochair
Department of Communication Neurobiology, House Ear Institute, 2100 West Third Street, Los Angeles, California 90057

Patricia A. Keating, Cochair
Department of Linguistics, University of California–Los Angeles, Los Angeles, California 90095-1543

Chair's Introduction—8:00

Invited Papers

8:05

4aSC1. Fundamentals of spoken word recognition. Paul A. Luce and Conor McLennan (Lang. Percept. Lab., Dept. of Psych. and Ctr. for Cognit. Sci., Univ. at Buffalo, Buffalo, NY 14260)

Researchers have made significant progress in identifying the basic principles responsible for the normal-listener's rapid and accurate identification of spoken words. In particular, there is now almost uniform consensus that spoken word recognition involves two fundamental processes: *activation* and *competition*. Most current models of recognition propose that stimulus input (i.e., a spoken word) activates a set of representations of *similar* sounding words in memory that subsequently vie for recognition. Despite the fact that *similarity* is afforded a crucial role in activating and discriminating among lexical competitors, we currently have little precise information regarding *perceived* similarity relations among spoken words. We will discuss the fundamental role of similarity in current models of spoken word recognition, past attempts to capture perceived similarity among spoken words, and ongoing efforts in our laboratory to understand more deeply the precise role of similarity in the perception of spoken stimuli. [Work supported by NIH.]

8:30

4aSC2. Two perspectives on similarity between words. Stefan A. Frisch (Dept. of CSD, Univ. of South Florida, 4202 E. Fowler Ave. PCD1017, Tampa, FL 33620, frisch@chuma1.cas.usf.edu)

This presentation examines the similarity between words from both bottom up (phonetic) and top down (phonological/psycholinguistic) perspectives. From the phonological perspective, the influence of structure on similarity is explored using metalinguistic acceptability judgments for multisyllabic nonwords. Results from an experiment suggest that subjects try to align novel words with known words in order to maximize similarities while minimizing dissimilarities. This finding parallels results from psychology on similarity judgments for visual scenes. From the phonetic perspective, the influence of similar gestures on speech error rates is examined using ultrasound measurement of tongue position. In a pilot experiment, subjects, produced tongue twisters containing words where onset and vowel phonemes had similar gestures (e.g., tip, comb) and where the onset and vowel had dissimilar gestures (e.g., tube, keep). Preliminary results suggest that misarticulations are more frequent in the context of dissimilar gestures (e.g., in the tongue twister tip cape keep tape, error rates are higher for /k/ than /t/). These errors appear to be gestural interactions rather than errors at the phonemic or featural level of phonological spellout. Together, these two experiments indicate that similarity relations between words are found at multiple levels, any which are potentially relevant to the structure of phonological systems.

8:55

4aSC3. Stimulus-based similarity and the recognition of spoken words. Edward T. Auer, Jr. (Dept. of Commun. Neurosci., House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057, eauer@hei.org)

Spoken word recognition has been hypothesized to be achieved via a competitive process amongst perceptually similar lexical candidates in the mental lexicon. In this process, lexical candidates are activated as a function of their perceived similarity to the spoken stimulus. The evidence supporting this hypothesis has largely come from studies of auditory word recognition. In this talk, evidence from our studies of visual spoken word recognition will be reviewed. Visual speech provides the opportunity to highlight the importance of stimulus-driven perceptual similarity because it presents a different pattern of segmental similarity than is afforded by auditory speech degraded by noise. Our results are consistent with stimulus-driven activation followed by competition as general spoken word recognition mechanism. In addition, results will be presented from recent investigations of the direct prediction of perceptual similarity from measurements of spoken stimuli. High levels of correlation have been observed between the predicted and perceptually obtained distances for a large set of spoken consonants. These results support the hypothesis that the perceptual structure of English consonants and vowels is predicted by stimulus structure without the need for an intervening level of abstract linguistic representation. [Research supported by NSF IIS 9996088 and NIH DC04856.]

9:20

4aSC4. Perceptual similarity co-existing with lexical dissimilarity. Andrea Weber (Dept. of Psycholinguist., Saarland Univ., 66041 Saarbruecken, Germany, aweber@coli.uni-sb.de) and Anne Cutler (Max Planck Inst. for Psycholinguist., Wundtlaan 1, 6525 XD Nijmegen, The Netherlands)

The extreme case of perceptual similarity is indiscriminability, as when two second-language phonemes map to a single native category. An example is the English had-head vowel contrast for Dutch listeners; Dutch has just one such central vowel, transcribed [E]. We examine whether the failure to discriminate in phonetic categorization implies indiscriminability in other—e.g., lexical—processing. Eyetracking experiments show that Dutch-native listeners instructed in English to ''click on the panda'' look (significantly more than native listeners) at a pictured pencil, suggesting that pan- activates their lexical representation of pencil. The reverse, however, is not the case: ''click on the pencil'' does not induce looks to a panda, suggesting that pen- does not activate panda in the lexicon. Thus prelexically undiscriminated second-language distinctions can nevertheless be maintained in stored lexical representations. The problem of mapping a resulting unitary input to two distinct categories in lexical representations is solved by allowing input to activate only one second-language category. For Dutch listeners to English, this is English [E], as a result of which no vowels in the signal ever map to words containing [ae]. We suggest that the choice of category is here motivated by a more abstract, phonemic, metric of similarity.

9:45–9:55 Break

9:55

4aSC5. The development of sensitivity to speech sound dissimilarity in humans. Curtis Ponton (Neuroscan, 7850 Paseo del Norte, El Paso, TX 79912)

The understanding of spoken language development has been expanded by examining speech-sound evoked brain activity recorded using an oddball presentation paradigm. When brain activity is recorded using this paradigm, a response known as the mismatch negativity (MMN) is generated. The MMN is regarded as a neurophysiological correlate of short-term auditory memory processes that are necessary for behavioral discrimination. Numerous investigations have demonstrated that neural generators underlying the MMN are sensitive to a wide range of acoustic contrasts, including both nonspeech and speech sounds. Recent studies have shown that for at least some speech sound contrasts, the MMN is present at birth and is insensitive to native/non-native speech contrasts. The appearance of language-specific contrasts around six months of age appears to correspond with the emergence of myelinated connections between the thalamus and auditory cortex. Combined, the neurophysiological and anatomical data suggest that this neural mechanism for detecting dissimilarity may have subcortical or cortical loci, depending on the speech contrast. Particularly for vowels, this dissimilarity processing may be the basis for developing categorization.

10:20

4aSC6. Neural and perceptual discrimination of the spectral and temporal modulations in birdsong and speech. Frederic Theunissen (Dept. of Psych., Univ. of California, 3210 Tolman, Berkeley, CA 94720-1650)

My laboratory is interested in the neural basis of complex sound perception, including vocalizations. Our neural studies have focused on the high-level auditory system of songbirds where neurons respond preferentially to birdsong. We quantified the selectivity of these auditory neurons by recording their responses to degraded versions of song. To do so, we added noise to this natural stimulus, affecting the time-frequency structure of the sound at different scales. We found that at particular time-frequency scales the noise has little effect on the response of the neurons, while at other scales it greatly reduces the neural response. We correlated the tuning of the neurons with the statistical structure in the birdsongs that allows the discrimination of songs produced by different birds. We found a good match between the optimal scale of the neurons sensitivity curve and the scale that best captures the variance in the acoustical structure of an ensemble of songs. We have done a similar analysis with speech, where the neural tuning curve in songbirds is replaced by a human perceptual tuning curve based on speech intelligibility. As in songbirds, the scale for optimal speech perception is also best for capturing the statistical structure in the speech signal.

10:45–11:00
Panel Discussion

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. Contributors will be at their posters from 11:00 a.m. to 12:00 noon.

4aSC7. Enhancement of distinctiveness in strident fricatives: An example from Mandarin Chinese. Chao-Yang Lee (School of Hearing, Speech and Lang. Sci., Ohio Univ., Athens, OH 45701) and Kenneth N. Stevens (Res. Lab. of Elec., MIT, Cambridge, MA 02139)

Strident fricatives in Mandarin may be followed by fricative vowels, which are made with the tongue body in essentially the same position as in the corresponding fricative (Ladefoged and Maddieson, 1996). Specifically, the vowel [i] may occur only after the palatalized fricative, the apical vowel after the alveolar fricative, and the retroflex vowel after the palatoalveolar fricative. Given the complementary distribution, the three phonetically distinct vowels are generally considered allophones. Under the view that the implementation of discretely specified distinctive features may be enhanced by gradient, non-contrastive articulatory gestures to maintain perceptual distinctiveness (Keyser and Stevens, 2003), it is proposed that the three vowels are derived from an underlying [i] and are modified to enhance the defining articulatory and acoustic attributes of the fricatives. Acoustic data from six speakers indicate that the defining acoustic attributes for the fricatives, i.e., spectral prominences in discrete frequency ranges, are associated with distinct tongue configurations. The articulatory and acoustic properties of the palatalized fricative is most

compatible with those of [i], whereas the tongue body backing gesture in the alveolar and palatoalveolar fricatives are consistent with lowered $F2$ frequency in the apical and retroflex vowels. [Work supported by NIH.]

4aSC8. The importance of duration in the perception of glides and vowels. Amy E. Coren (Dept. of Psych., Univ. of Texas, SEA 4.212, 108 E. Dean Keeton St., Austin, TX 78712-0198, aecoren@mail.utexas.edu) and Willis J. Warren (Univ. of Texas, Austin, TX 78712-0198)

The segmental duration for the perception of glides versus vowels was investigated. A central Texas Spanish speaker produced a single token of radio with a high vowel duration of 150 ms. The duration of the high vowel, [i], was edited at the midpoint in 10-s intervals to see at what length listeners perceive a complete vowel versus a glide, i.e., radyo versus radio. Ten listeners judged ten repitions of the ten tokens. It was determined that the steady state of the vowel must retain somewhere near 80 ms to be perceived as a full vowel.

4aSC9. Changes in $F2-F1$ as a voicing cue. Willis J. Warren (Dept. of Linguist., Univ. of Texas, Calhoun Hall 501, 1 University Station B5100, Austin, TX 78712-0198, warisill@mail.utexas.edu) and Amy E. Coren (Univ. of Texas, Austin, TX 78712-0198)

The interaction between formant transitions and vowel length was measured with respect to syllable final voicing distinctions. A synthesized *ad* VC token of 360 ms was edited in 5-ms intervals from either side, onset or offset, so that 260 ms were preserved. Ten subjects were asked to make final voicing judgments for the words ''odd'' and ''ought'' ([ad] vs [at]) when hearing the 20 edited tokens. Each token was presented five times, randomly, for a total of 1000 judgements. Results showed an overwhelming number of voiced responses when the entire offset was preserved and symmetrical voiceless results with the deletion of offset. A follow-up experiment utilized a similarly synthesized token of 460 ms. The results when adding 100 ms onto the vowel were insignificantly different than the results acquired for formant transitions, suggesting the latter are a more important cue for syllable final voicing distinctions. These findings contradict previous vowel length conclusions [L. J. Raphael, J. Acoust. Soc. Am. **51**, 1296–1303 (1972)] and further suggest that in addition to $F1$ [V. Summers, J. Acoust. Soc. Am. **84**, 485–492 (1988)], $F2$ transitions are also an important cue to final voicing distinctions in low vowel contexts.

4aSC10. Effects of blocking and presentation on the recognition of word and nonsense syllables in noise. José R. Benkí (Dept. of Linguist., Univ. of Michigan, Ann Arbor, MI 48109-1285, benki@umich.edu)

Listener expectations may have significant effects on spoken word recognition, modulating word similarity effects from the lexicon. This study investigates the effect of blocking by lexical status on the recognition of word and nonsense syllables in noise. 240 phonemically matched word and nonsense CVC syllables [Boothroyd and Nittrouer, J. Acoust. Soc. Am. **84**, 101–108 (1988)] were presented to listeners at different S/N ratios for identification. In the mixed condition, listeners were presented with blocks containing both words and nonwords, while listeners in the blocked condition were presented with the trials in blocks containing either words or nonwords. The targets were presented in isolation with 50 ms of preceding and following noise. Preliminary results indicate no effect of blocking on accuracy for either word or nonsense syllables; results from neighborhood density analyses will be presented. Consistent with previous studies, a j-factor analysis indicates that words are perceived as containing at least 0.5 fewer independent units than nonwords in both conditions. Relative to previous work on syllables presented in a frame sentence [Benkí, J. Acoust. Soc. Am. **113**, 1689–1705 (2003)], initial consonants were perceived significantly less accurately, while vowels and final consonants were perceived at comparable rates.

4aSC11. Influence of semantic similarity on spoken word recognition. Jonna L. Armbruster and Michael S. Vitevitch (Dept. of Psych., 1415 Jayhawk Blvd., Lawrence, KS 66045)

Previous research has shown that the number of phonologically similar items influences the processing of spoken words (e.g., Vitevitch, 2002; Vitevitch and Luce, 1998). The present experiment examines the influence of semantic similarity on the speed and accuracy of spoken word recognition using a lexical decision task. Semantic density refers to the number of words that are semantically associated to a given word (Nelson, McEvoy, and Schreiber, 1998). A word that has relatively many semantically associated words has high semantic density and a word that has relatively few semantically associated words has low semantic density. The results showed that words with high semantic density were responded to more quickly and accurately than words with low semantic density, suggesting that semantic information influences spoken word recognition.

4aSC12. The impact of phonetic dissimilarity on the perception of foreign accented speech. Shawn A. Weil (Sytronics, Inc., Dayton, OH and Ohio State Univ., Columbus, OH)

Non-normative speech (i.e., synthetic speech, pathological speech, foreign accented speech) is more difficult to process for native listeners than is normative speech. Does perceptual dissimilarity affect only intelligibility, or are there other costs to processing? The current series of experiments investigates both the intelligibility and time course of foreign accented speech (FAS) perception. Native English listeners heard single English words spoken by both native English speakers and non-native speakers (Mandarin or Russian). Words were chosen based on the similarity between the phonetic inventories of the respective languages. Three experimental designs were used: a cross-modal matching task, a word repetition (shadowing) task, and two subjective ratings tasks which measured impressions of accentedness and effortfulness. The results replicate previous investigations that have found that FAS significantly lowers word intelligibility. Furthermore, in FAS as well as perceptual effort, in the word repetition task, correct responses are slower to accented words than to nonaccented words. An analysis indicates that both intelligibility and reaction time are, in part, functions of the similarity between the talker's utterance and the listener's representation of the word.

Session 4aSP

Signal Processing in Acoustics, Underwater Acoustics, Speech Communication, Animal Bioacoustics, Noise and Engineering Acoustics: Detection and Classification in Acoustics III

Paul M. Baggenstoss, Chair
Naval Undersea Warfare Center, Newport, Rhode Island 02840

Invited Papers

8:00

4aSP1. Audio signal recognition for speech, music, and environmental sounds. Daniel P. W. Ellis (Dept. of Elec. Eng., Columbia Univ., 500 W. 120th St., New York, NY 10027, dpwe@ee.columbia.edu)

Human listeners are very good at all kinds of sound detection and identification tasks, from understanding heavily accented speech to noticing a ringing phone underneath music playing at full blast. Efforts to duplicate these abilities on computer have been particularly intense in the area of speech recognition, and it is instructive to review which approaches have proved most powerful, and which major problems still remain. The features and models developed for speech have found applications in other audio recognition tasks, including musical signal analysis, and the problems of analyzing the general ''ambient'' audio that might be encountered by an auditorily endowed robot. This talk will briefly review statistical pattern recognition for audio signals, giving examples in several of these domains. Particular emphasis will be given to common aspects and lessons learned.

8:45

4aSP2. Recognition of information-bearing elements in speech. Hynek Hermansky (Institut Dalle Molle d'Intelligence Artificielle Perceptive, Rue du Simplon 4, Case Postale 592, CH-1920 Martigny, Switzerland and Intl. Computer Sci. Inst., Berkeley, CA, hynek.hermansky@idiap.ch)

An acoustic speech signal carries many different kinds of information: the basic linguistic message, many characteristics of the speaker of the message, details of the environment in which the message was produced and transmitted, etc. The human auditory/cognitive system is able to detect, decode, and separate all these information sources. Understanding this ability and emulating it on a machine has been an important but elusive scientific and engineering goal for a long time. This talk critically surveys the situation in the speech recognition field. It puts automatic recognition of speech in perspective with other acoustic signal detection and classification tasks, reviews some historical, contemporary, and evolving techniques for machine recognition of speech, critically compares competing techniques, and gives some examples of applications in speech, speaker, and language recognition and identification. The talk is intended for an audience interested but not directly involved in the processing of speech.

9:30

4aSP3. Acoustic classification: An overview of theory and reality. Stephen Greineder (Naval Undersea Warfare Ctr. Div. Newport, 1176 Howell St., Newport, RI 02841, GreinederSG@Npt.NUWC.Navy.Mil)

Historically, Bayes' decision theory has formed the statistical foundation for the development of acoustic classification techniques for problems that contain large numbers of data classes. The successful realization of this approach depends directly on mitigating two important mismatches that are introduced using this classification model. First, the underlying probabilistic structure in the problem is not typically known and therefore must be estimated as part of a classifier training phase. Second, due to the high dimensionality of time series data, a feature extraction stage is added as a preprocessor to most classifiers. Attempting to extract the sufficient information in the problem while limiting the feature set size to a dimension supported by the training data is the ultimate designer's challenge. This is particularly true in the case of finite data sample sizes. This talk will trace the development evolution of a low false alarm rate multiclass operational algorithm that will provide an overview of the fundamental challenges and resulting mitigation approaches. Examples using real data will be used to demonstrate the points being presented.

10:15–10:30 Break

10:30

4aSP4. The class-specific method for classification. Paul M. Baggenstoss (Naval Undersea Warfare Ctr., Newport, RI 02840)

This talk describes a new probabilistic method for classification called the ''class-specific method'' (CSM). CSM is able to avoid the ''curse of dimensionality'' which plagues most classifiers which attempt to determine the decision boundaries in a high-dimensional feature space. Using CSM, it is possible to build a theoretically optimum classifier without a common feature space. Separate low-dimensional features sets may be defined for each class, while the decision functions are projected back to the common *raw data* space. CSM effectively extends classical classification theory to handle multiple feature spaces. It is completely general, and

requires no simplifying assumption such as Gaussianity or that data lies in linear subspaces. In real-data problems, CSM has shown orders of magnitude reductions in the false-alarm rate. CSM achieves this gain because it is able to make use of partial prior knowledge about the data classes. In contrast, the existing theory can only make use of full knowledge—that is when the parametric forms of the data probability density functions (PDFs) are known.

Contributed Papers

11:00

4aSP5. Automatic computational models of acoustical category features: Talking versus singing. David Gerhard (Dept. of Computer Sci., Univ. of Regina, 3737 Wascana Pkwy., Regina, SK S4S 0A2, Canada, david.gerhard@uregina.ca)

The automatic discrimination between acoustical categories has been an increasingly interesting problem in the fields of computer listening, multimedia databases, and music information retrieval. A system is presented which automatically generates classification models, given a set of destination classes and a set of *a priori* labeled acoustic events. Computational models are created using comparative probability density estimations. For the specific example presented, the destination classes are talking and singing. Individual feature models are evaluated using two measures: The Kologorov–Smirnov distance measures feature separation, and accuracy is measured using absolute and relative metrics. The system automatically segments the event set into a user-defined number (n) of development subsets, and runs a development cycle for each set, generating n separate systems, each of which is evaluated using the above metrics to improve overall system accuracy and to reduce inherent data skew from any one development subset. Multiple features for the same acoustical categories are then compared for underlying feature overlap using cross-correlation. Advantages of automated computational models include improved system development and testing, shortened development cycle, and automation of common system evaluation tasks. Numerical results are presented relating to the talking/singing classification problem.

11:15

4aSP6. Simple algorithm for classification of target in shallow water. Angie Sarkissian (Naval Res. Lab., Washington, DC 20375)

When applying classification algorithms to the scattering response of a target in shallow water, algorithms that attempt to deconvolve the free field target response from the environment, such as modal decomposition or time reversal, typically require multiple receivers. A simple algorithm is applied here that requires a single source and a single receiver. The impulse response of the target is Fourier transformed to the frequency domain; the magnitude of the frequency domain response is transformed back to the time domain; the resultant time domain response is truncated. The algorithm is applied to a hemispherically end-capped cylindrical shell under various shallow water conditions. The response obtained for the target in shallow water is correlated against the target free field response. [Work supported by ONR.]

THURSDAY MORNING, 13 NOVEMBER 2003 SABINE ROOM, 8:00 TO 11:40 A.M.

Session 4aUWa

Underwater Acoustics, Engineering Acoustics and Signal Processing in Acoustics: Gradient Array Acoustics I

Paul C. Hines, Cochair
Defence R&D Canada Atlantic, P.O. Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada

Daniel L. Hutt, Cochair
Defence R&D Canada Atlantic, P.O. Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada

Chair's Introduction—8:00

Invited Papers

8:05

4aUWa1. Superdirective and gradient sensor arrays. Harold M. Merklinger (Defence R&D Canada Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada)

During the late 1960s and the 1970s, underwater acoustic investigators examined superdirective and gradient sensor systems in order to enhance submarine detection capabilities for surface ships and maritime aircraft. Simple gradient processing had already been used in both in-air acoustic systems (cardioid and super-cardioid microphones) as well as radio and radar applications. Superdirective techniques were known [R. L. Pritchard, J. Acoust. Soc. Am. **25**, 879 (1953)] and sometimes exploited in radar systems. It was quickly demonstrated that simple gradient sensors and modest degrees of superdirective array processing were possible, although self-noise and the ability to calibrate hydrophones limited the processing gains achievable. Circular superdirective arrays were used extensively by the Defence Research Establishment Atlantic for noise directionality measurements in the frequency range 4 Hz to about 1 kHz and considered for naval ASW applications until the superiority of oil-filled conventional arrays became apparent. Nevertheless, the significant theoretical and practical development of spatial harmonic beamforming and direction finding was completed. Although much of this work was not considered classified, neither was it widely published. This presentation will review the concepts developed and progress made. Beamforming, noise mitigation and calibration issues are covered.

8:30

4aUWa2. Differential and gradient microphone arrays. Gary W. Elko (Avaya Labs, gwe@avaya.com), James E. West (Johns Hopkins Univ.), and Steve Thompson (Knowles Electron. LLC)

Differential microphone arrays have been in existence for more than 7 decades and are the basis of most commercial directional microphones in use today. These microphones obtain directionality by combining the acoustic pressure and the pressure-difference to form what is termed a first-order differential microphone. Differential microphones are inherently superdirectional since they can obtain broadband directional gains of up to 6.0 dB in an array that is physically much smaller than the acoustic wavelength. Differential arrays constructed by subtracting omnidirectional microphones are inherently more flexible in that the directional response can be easily and continuously varied from omnidirectional to hypercardioid. The simultaneous measurement of the acoustic pressure and particle velocity allows one to estimate the complex acoustic intensity along the axis of a microphone pair. A measure of the complex acoustic intensity vector can be obtained using a minimum of four pressure-sensing microphones. Higher-order differential microphones are also possible by using more microphone elements, but the problems of microphone calibration and signal-to-noise combine to practically realize microphones of differential order greater than third order. We will present some of the history of differential microphone array design and discuss some applications related to hands-free communication, hearing aids, and spatial audio recording.

Contributed Papers

8:55

4aUWa3. Performance of a superdirective line array in nonideal environments. Paul C. Hines (Defence R&D Canada Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada), Victor F. Humphrey (Dept. of Phys., Univ. of Bath, Bath BA2 7AY, UK), and Victor Young (Defence R&D Canada Atlantic, Dartmouth, NS B2Y 3Z7, Canada)

Superdirective line arrays can provide high gains relative to their dimensions, whenever the inter-element spacing is much less than half a wavelength. However, their performance can be degraded by system noise. System noise can result from limitations in electronic components, inter-element mismatch in gain or phase, or from scatter from array components. Using state-of-the-art electronics and digital signal processing one can drastically reduce the errors due to electronic noise as well as those due to gain and phase mismatch. Thus, acoustic scatter from the array components can determine the performance limit. Since theoretical developments typically assume plane waves incident on idealized point receivers, it is not all that surprising that array performance fails to meet theoretical expectations! This is especially true if the array is merely one component within a much larger system. Nonetheless, impressive gains can still be realized from superdirective arrays even with significant departures from the idealized model. To support this observation, we present results obtained using a 0.8 m long, 5th order superdirective receiver that is part of a much larger system used to study scattering from marine sediments.

9:10

4aUWa4. Experimental performance analysis of a superdirective line array. Victor F. Humphrey (Dept. of Phys., Univ. of Bath, Bath BA2 7AY, UK, v.f.humphrey@bath.ac.uk), Paul C. Hines, and Victor Young (Defence R&D Canada Atlantic, Dartmouth, NS B2Y 3Z7, Canada)

Superdirective line arrays can provide a significant array gain from a structure that is relatively small in terms of acoustic wavelengths. However, system imperfections, electronic noise and acoustic scatter from the array structure can degrade their performance. An acoustic calibration of a six-element line array, 0.8 m in length, has been performed over the frequency range 1 to 4 kHz in order to investigate the performance of a real array. The data is used to identify the angular variation of the hydrophone outputs and the phase difference between hydrophone pairs. These angular responses are analyzed in terms of a modal series in order to quantify the variations and help identify the source of perturbations. The effects of imperfections are also investigated by synthesising superdirective arrays of order 1 to 5 and monitoring how the array gain varies for both deterministic signals and ambient acoustic noise. These results are compared with theoretical predictions. Further evidence of the variation in performance is gained by comparing the output of different implementations of lower order arrays, synthesized from subsets of the full array. The results indicate the influences that the array structure may have on the performance of the array.

9:25

4aUWa5. Acoustic intensity in the interaction region of a parametric source. G. C. Lauchle, T. B. Gabrielson, D. J. Van Tol, N. F. Kottke (Appl. Res. Lab and Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804), and J. A. McConnell (Acoustech Corp., State College, PA 16804)

The goal of this project was to measure acoustic intensity in the strong interaction region of a parametric source in order to obtain a clear definition of the source-generation region and to separate the local generation (the reactive field) from propagation (the real or active field). The acoustic intensity vector was mapped in the interaction region of a parametric projector at Lake Seneca. The source was driven with primary signals at 22 kHz and 27 kHz. Receiving sensors were located 8.5 meters from the projector. At that range, the secondary at 5 kHz was between 40 and 45 dB below either primary. For the primary levels used, the plane-wave shock inception distance would have been at least 14 meters. Furthermore, the Rayleigh distance for the projector was about 4 meters so the measurements at 8.5 meters were in the strong interaction region but not in saturation. Absorption was negligible over these ranges. The intensity measurements were made at fixed range but varying azimuth angle and varying depth thus developing a two-dimensional cross-section of the secondary beam. Measurements of both the active and reactive intensity vectors will be presented along with a discussion of measurement error. [Work supported by ONR Code 321SS.]

9:40

4aUWa6. Performance of vector sensors in noise. Henry Cox (Lockheed Martin Orincon, 4350 N. Fairfax Dr., Arlington, VA 22203) and Arthur Baggeroer (MIT, Cambridge, MA 02139)

Vector sensors are super gain devices that can provide "array gain" against ocean noise with a point sensor. As supergain devices they have increased sensitivity to nonacoustic noise components. This paper reviews and summarizes the processing gain that is achievable in various noise fields. Comparisons are made with an omni-directional sensor and with the correlation of a pair of closely spaced omni-directional sensors. Total processing gain that consists of both spatial and temporal gain is considered so that a proper analysis and interpretation of multiplicative processing can be made. The performance of "intensity sensors" (pressure times velocity) that are obtained by multiplying the omnidirectional component with a co-located dipole is also considered. A misinterpretation, that is common in the literature, concerning the performance of intensity sensors is discussed. The adaptive cardioid processing of vector sensors is also reviewed.

9:55–10:10 **Break**

10:10

4aUWa7. Comparison of the performance of vector and sensors using optimum array processing. Arthur Baggeroer (MIT, Cambridge, MA 02139) and Henry Cox (Lockheed Martin Orincon, Arlington, VA 22203)

Vector sensors in sonar are often used for direction finding. With the appropriate processing three component vector sensors are also quite effective at nulling directional interference as well as estimating the ambient acoustic intensity vector. Here we examine optimum array processing (MVDR based) methods for both these applications. We first present the detection performance of a plane wave source operating in directional noise fields as well as noise field based a spheroidial harmonic expansion for modeling diffuse surface, seabed and ducted environments. Next, we use these optimal methods for estimating the intensity vector. Finally, we compare the performance to an array of pressure sensors using the same number of output channels. This indicates that vector sensors are effective against some types of noise fields but not uniformly better than an array of scalar sensors.

10:25

4aUWa8. Highly directional receivers using various combinations of scalar, vector, and dyadic sensors. James A. McConnell (Acoustech Corp., P.O. Box 139, State College, PA 16804)

The generalized theory of directional sensors is presented in the form of the Taylor series expansion of the acoustic pressure about a point in space. If the expansion is truncated to second order, the analysis of scalar, vector, and dyadic sensors can be made and corresponds to the zeroth-, first-, and second-order gradient of the acoustic pressure. This translates into using a sufficient number of omni-directional hydrophones or a multimode hydrophone in conjunction with the appropriate finite-differencing operations to achieve the desired beam patterns. Using the linearized Euler equation, the formulation can be recast in terms of the zeroth-order gradient of the acoustic pressure along with the zeroth- and first-order gradient of the particle acceleration. In this case, the zeroth-order terms can be measured directly with an omni-directional hydrophone and a neutrally buoyant accelerometer. The gradient of the particle acceleration can be measured indirectly using finite differences or directly by measuring the angular acceleration akin to a Rayleigh disk. Of particular interest is the use of scalar, vector, and dyadic sensors to localize sources of sound using arc-tangent-squared processing and cardioid-squared processing as opposed to conventional arc-tangent and cardioid processing. [Work supported by ONR 321SS.]

10:40

4aUWa9. Practical application of a tri-axial intensity array. Victor W. Young, Paul C. Hines, Daniel L. Hutt, and Victor F. Humphrey (DRDC Atlantic, P.O. Box 1012, Dartmouth, NS, Canada, victor.young@drdcrddc.gc.ca)

Sound intensity is a vector quantity representing the magnitude and direction of propagating energy within an acoustic field. In an underwater environment, a single omni-directional hydrophone can be used to measure instantaneous acoustic pressure and a finite difference approximation applied to the pressure signals from a pair of such hydrophones can be used to calculate particle velocity in a single direction. Because the time average of the product of instantaneous pressure and particle velocity is intensity, a pair of hydrophones is all that is required to measure a single component of the intensity vector. The complete three-dimensional intensity vector can be calculated using three orthogonal pairs of hydrophones. To evaluate this concept a tri-axial array consisting of three orthogonal pairs of omni-directional hydrophones has been developed and tested on both calibrated sources at a laboratory facility and sources of opportunity during sea trails in littoral waters. The use of this array to calculate the intensity vector and thereby localize both near-field and far-field acoustic sources and characterize the directionality of ambient noise fields will be discussed. The impact of signal-to-noise ratio and the effect of self-noise will also be examined.

10:55

4aUWa10. Cardioid processsing of reverberation data observed on the STRATAFORM with the Five Octave Research Array (FORA). John R. Preston (ARL, Penn State Univ., P.O. Box 30, State College, PA 16804)

The author recently participated in ONR's 2003 Geoclutter Experiment to study shallow water bottom reverberation and clutter in the STRATAFORM off New Jersey. The experimental effort was lead by M.I.T. and included researchers from the Naval Research Laboratory, Penn State Univ., and the Naval Underwater Weapons Center. Sources were bistatic coherent pulses from 400 to 3600 Hz. The receiver was the new Five Octave Research Array (FORA). The STRATAFORM is known to have benign surface morphology but contains many buried river channels and other sub-surface horizons. Some highlights of the reverberant returns are discussed that include the correlation of returns with a bottom mounted target and probable fish schools. The main objective of this work is to assess the directional characterization of the observed clutter. The cardioid aperture of FORA should, in theory, yield good estimates of the sources of reverberation above ~1800 Hz. Examples from the rverberation data analysis are presented using a cardioid beamforming algorithm developed by SACLANTCEN. [Work supported by ONR Code 32, Grant N00014-03-1-0113.]

11:10

4aUWa11. Separation of signals from near field noise via pressure and velocity field measurements. Peter R. Stepanishen (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882-1197)

The separation of acoustic plane wave signals from near field noise sources is addressed using both pressure and velocity measurements at the same location in the fluid. A compact broadband noise source at a known location is first assumed to generate the noise field. Using a time domain approach a simple signal processor is presented to separate the unknown signal from the noise. Effects of sensor and quantization noise are included in the analysis. A frequency domain analysis of the noise separation approach is presented along with numerical results to illustrate the sensitivity of the method to field point location, noise spectrum, and sensor and quantization noise. The separation of plane wave signals from spatially distributed near field noise sources is then addressed. A signal processor using both pressure and velocity measurements at a single point is developed to separate a plane wave signal from a spectrally pure evanescent noise field generated at a planar interface close to the sensors. Numerical results are presented for several examples to illustrate the noise separation process for a wide range of signals. In particular, the case of overlapping signal and noise spectrums is addressed.

11:25

4aUWa12. Wideband direction finding via shielded gradient beamspace techniques. Terry J. Brudner and Terry L. Henderson (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

Monopulse techniques have been used for over 50 years in the radar community to estimate the direction of arrival (DOA) of incoming echoes. In recent years, a variant of the monopulse technique has been developed, termed the shielded gradient technique, which allows DOA estimation for signals of arbitrary bandwidth. The technique maps the array-output M-vector into a frequency-invariant B-dimensional beamspace. The work presented here describes the shielded gradient beamspace model in its higher-order form, and develops wideband DOA estimation algorithms analogous to the narrow-band MUSIC, root-MUSIC, and ESPRIT algorithms. The performance of these new algorithms is studied through simulation and application to measured, in-water sonar data. They are also compared via simulation to existing wideband DOA estimation algorithms. [Work supported by the Internal Research and Development Program under Contract No. FEE-800.]

THURSDAY MORNING, 13 NOVEMBER 2003 PECOS ROOM, 8:00 TO 11:45 A.M.

Session 4aUWb

Underwater Acoustics: Modeling of Propagation and Scattering

Nicholas C. Makris, Chair

Department of Ocean Engineering, Massachusetts Institute of Technology, 77 Massachusetts Avenue, Cambridge, Massachusetts 02139

Contributed Papers

8:00

4aUWb1. Ocean acoustic wave propagation and ray method correspondence: Internal wave fine structure. Nicholas R. Cerruti, Katherine C. Hegewisch, and Steven Tomsovic (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814, ncerruti@wsu.edu)

Acoustical wave fields can only detect structures in the ocean's sound-speed fluctuations that are on the order of the smallest acoustic wavelength component. On the other hand, geometrical ray-tracing methods are sensitive to infinitely fine structures in the wave modeling of these fluctuations. Hence, a proper model seeking agreement between ray methods and wave propagation must, at a minimum, filter out the fine oscillations of the internal wave field. Starting from an efficient numerical scheme for generating the internal waves introduced by Brown and Colosi [J. Acoust. Soc. Am. **103**, 2232 (1998)], which reproduces the Garrett–Munk spectrum, we introduce a smoothing technique that removes the unphysical portion of the internal wave modeling. The key is to find a smoothing that does not significantly alter the propagated wave field, yet eliminates as much of the "micro-folding" phenomena that was discussed by Simmen *et al.* [J. Acoust. Soc. Am. **102**, 239 (1997)]. We give a characterization of the extent of smoothing necessary as a function of acoustic frequency and propagation range, and show how it improves the correspondence. Finally, we note that the smoothing allows detailed ray methods to be implemented further in range. [Work supported by ONR.]

8:15

4aUWb2. Covariance of the forward propagated field through a waveguide containing random inhomogeneities. Purnima Ratilal and Nicholas C. Makris (MIT, Cambridge, MA 02139)

Analytic expressions are derived for the spatial covariance of the field from a point source after forward propagation though a waveguide containing random surface and volume inhomogeneities. It is shown that the depth-averaged second moment and expected power of the forward propagated field can be obtained analytically. The mean forward propagated field has also been obtained analytically in terms of modal attenuation and dispersion coefficients in a waveguide [P. Ratilal and N. C. Makris **112**, 2403 (2002)]. The covariance between two receiver depths of the forward propagated field through the entire random medium can then be determined by invoking the equi-partition of modal energy after significant multiple scattering. It is expressible as a sum of modal covariance terms. Each term depends on (1) the modal extinction cross-section [P. Ratilal and N. C. Makris, **110**, 2924–2945 (2001)] of an expected elemental inhomogeneity of the medium, and (2) the scatter function variance of an elemental inhomogeneity which couples each mode to all the other modes.

8:30

4aUWb3. Energy-conserving and single-scattering parabolic equation solutions for elastic media. Elizabeth T. Kusel, William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180, kusele@rpi.edu), Michael D. Collins, and Joseph F. Lingevitch (Naval Res. Lab., Washington, DC 20375)

Parabolic equation techniques are efficient for solving nonseparable wave propagation problems. When the properties of the medium vary gradually in range, parabolic equation solutions are also very accurate for many problems. The key to achieving accuracy and efficiency simultaneously is to apply energy-conservation or single-scattering corrections to account properly for range dependence. This approach has proven to be very effective for acoustic media. Some progress has been made on the elastic case [J. Acoust. Soc. Am. **94**, 975–982 (1993); **94**, 1815–1825 (1993)], but this problem has not been fully resolved. In this paper we will discuss some recent progress in the formulation of the elastic parabolic equation [W. Jerzak, J. Acoust. Soc. Am. (submitted)], a single-scattering approach for a vector wave problem [J. Acoust. Soc. Am. **104**, 783–790 (1998)], and how they are being used to improve the accuracy of parabolic equation solutions for problems involving elastic sediments. [Work supported by ONR.]

8:45

4aUWb4. Generalization of the rotated parabolic equation to variable slopes. Donald A. Outing, William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180, outind@rpi.edu), and Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

The parabolic equation method is very efficient for solving range-dependent propagation problems. This approach is also accurate when range dependence is treated properly. This problem was resolved for fluid media by applying energy-conservation [J. Acoust. Soc. Am. **89**, 1058–1075 (1991)] and single-scattering [J. Acoust. Soc. Am. **91**, 1357–1368 (1992)] corrections. Since these corrections were less successful for problems involving elastic layers, other approaches such as mapping [J. Acoust. Soc. Am. **107**, 1937–1942 (2000)] and rotating [J. Acoust. Soc. Am. **87**, 1035–1037 (1990)] coordinates were investigated. In this presentation, the rotated parabolic equation solution is generalized to problems involving variable slope. The medium is divided into a series of regions with constant slope. When changes in slope are encountered, the field is propagated beyond the change and then used to interpolate and extrapolate onto a computational grid that is rotated relative to the previous grid. This approach is implemented and tested for the fluid problem. It should also be applicable to the elastic problem, but it will be necessary to apply a change

of variables each time the slope changes since the dependent variables are the tangential and normal displacements. [Work supported by ONR.]

9:00

4aUWb5. Analysis of measured broadband acoustic propagation using a parabolic equation approach. Mason Gray, D. P. Knobles, and Robert Koch (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

A broadband parabolic equation (PE) approach is employed to simulate data taken from two Shallow Water Acoustic Measurement Instrument (SWAMI) bottom mounted horizontal line array (HLA) experiments in shallow water environments off the east coast of the U.S. and in the Gulf of Mexico. In both experiments the HLA was deployed along an isobath. Light bulbs were imploded at known depths and ranges in both the range-independent (array end fire) and range-dependent (array broadside) directions. For the east coast experimental data, the PE model is used to infer a seabed geoacoustic description in both the range-dependent and range-independent directions. Also, comparisons of modeled time series were made for the range-independent case with a broadband normal mode model to validate the PE calculations. In the Gulf of Mexico experiment, the sediment geoacoustic profile is well known from previous inversions and geophysical measurements. This known seabed description was used to simulate the range-dependent data. A broadband energy-conserving coupled mode approach is also employed to model the range-dependent propagation. This allows the physical mechanisms associated with range-dependent propagation to be examined in a quantitative manner for this shallow water environment. [Work supported by ONR.]

9:15

4aUWb6. Sound scattering by 3-D, time-dependent internal gravity wave fields in the ocean. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA/Environ. Technol. Lab., Boulder, CO 80305-3328, Oleg.Godin@noaa.gov), Alexander G. Voronovich, and Valery U. Zavorotny (NOAA/Environ. Technol. Lab., Boulder, CO 80305-3328)

In a wide range of sound frequencies, internal gravity waves (IWs) are the major cause of underwater sound scattering and resulting fluctuations in acoustic quantities. In this paper, IW-induced variations in acoustic travel times, grazing and azimuthal arrival angles, and frequency spectra are analyzed and their implications for acoustic oceanography are emphasized. Statistical properties of IW-induced acoustic fluctuations are studied within ray and modal representations of the acoustic field. Ray perturbation theory is developed to extend results previously obtained using eikonal perturbation theory to rays which encounter an arbitrary number of caustics on their way from a source to a receiver. Effects of mode coupling on acoustic travel time variance and bias are quantified in range-dependent and horizontally-inhomogeneous environments. Time-dependent, IW-induced variations of sound speed are shown to result in a detectable wander of frequency of the received signal emitted by a CW sound source at rest. Acoustic frequency variance on refracted rays is proportional to spatial density of IW energy. Feasibility of quantifying internal wave energy density and its variation in time through measurements of spectra of acoustic signals received along resolved ray paths from a highly stable source is discussed. [Work supported by ONR.]

9:30

4aUWb7. Nonlinear modes interaction in an acoustic waveguide. Kaelig Castor, Philippe Roux, W. A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA 92037), and B. E. McDonald (Naval Res. Lab., Washington, DC 20375)

The nonlinear interaction of lower order modes in an acoustic waveguide is investigated. For finite-amplitude wave propagation, nonlinear effects redistribute the modal amplitudes as a function across frequency and wave number. Therefore, the nonlinear propagation in a waveguide can produce an arrival structure quite different from the classical result. Simulations with the NPE code [McDonald and Kuperman, J. Acoust. Soc. Am. **81**, 1406–1417 (1987)] were used to study this frequency mode coupling in a realistic ocean waveguide.

9:45

4aUWb8. Scattering in a 3-D wedge with a rough bottom boundary. D. P. Knobles (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

Scattering problems in ocean waveguides may be formulated in the context of a set of coupled inhomogeneous integral equations. A particular problem of interest is the physics of low frequency reverberation in an ocean waveguide with a bottom described by a rough interface superimposed on a sloped boundary that defines the water–sediment interface. An idealization of such a waveguide is a 3-D wedge with a rough bottom. The Helmholtz equation for the propagation of sound for such an environment is examined by solving the basic coupled integral equations. Reverberation time series for a bistatic source–receiver geometry are computed by Fourier synthesis of the frequency response of the sum of the direct, refracted, and scattered components of the acoustic field. An objective of the computations is to examine the interplay between the refracted component of the field that results from the wedge geometry and the scattering components that result from the rough bottom. Although this picture of reverberation is incomplete since the physics of volume scattering is ignored, it allows one to examine the basic physics of reverberation in a waveguide that has variations in both range and azimuth. [Work supported by ONR.]

10:00–10:15 Break

10:15

4aUWb9. The coupling of elastic, surface-wave modes by a slow, interfacial inclusion. John G. Harris (Ctr. QEFP, Northwestern Univ., 2137 N. Sheridan Rd., Evanston, IL 60208-3020, j-harris8@northwestern.edu) and Gareth Block (Univ. of Texas–Austin, Austin, TX 78713-8029)

The coupling of in-plane, elastic, surface waves guided by a homogeneous layer on a similar substrate, but perturbed by the presence of a second, long, slow, interfacial layer, of slowly varying thickness, is studied. By projecting the elastic-wave equations onto a basis of local eigenmodes, an infinite system of coupled-mode equations describing the evolution of the amplitudes of each mode is obtained. Within the equations, the coupling is manifested by the presence of coupling coefficients that depend critically on the difference between the wavenumbers of adjacent modes. This system is truncated and solved by noting under what conditions the modes can be satisfactorily described by a WKBJ approximation (propagating without coupling) and under what conditions they are coupled by adjacent wavenumbers being brought into proximity by the slowly changing propagation environment. The criteria for coupling is that the difference in neighboring wavenumbers be of the same order as the slope of the inclusion. The case of the first three modes is worked out in detail: it is shown how initially modes one and two couple followed by the coupling of modes two and three.

10:30

4aUWb10. A sub-mesoscale hydrodynamic/acoustic simulation model for continental shelf-break regions. Steven Finette (Acoust. Div., Naval Res. Lab., Washington, DC 20375), Colin Y. Shen, and Thomas E. Evans (Naval Res. Lab., Washington, DC 20375)

A nonhydrostatic, hydrodynamic model of the sound speed field in a continental shelf-break environment has been developed and implemented. The model is based on a vorticity formulation of the equations of motion for an incompressible fluid with a free ocean surface, and it is capable of

simulating the generation and propagation of internal tides and solibores under tidal forcing. The model has been benchmarked with an exact numerical solution for a soliton. A set of space and time evolving sound speed distributions is integrated with a parabolic equation code to compute time and frequency dependent pressure fields. Two-dimensional examples of broad-band signal gain degradation on vertical arrays in this environment are presented, as well as range-frequency maps that illustrate the structure of the waveguide invariant in a shelf-break environment that is changing in time. Implications for source localization are considered. [Work supported by ONR.]

10:45

4aUWb11. Optimal temperature sampling with SPOTS to improve acoustic predictions. Erik R. Rike, Donald R. DelBalzo (Neptune Sci., Inc., 40201 Hwy. 190 E., Slidell, LA 70461), and Brian C. Samuels (Appl. Hydro-Acoust. Res., Inc., Centerville, VA 20120)

The Modular Ocean Data Assimilation System (MODAS) uses optimal interpolation to assimilate data (e.g., XBTs), and to create temperature nowcasts and associated uncertainties. When XBTs are dropped in a uniform grid (during surveys) or in random patterns and spaced according to resources available their assimilation can lead to nowcast errors in complex, littoral regions, especially when only a few measurements are available. To mitigate, Sensor Placement for Optimal Temperature Sampling (SPOTS) [Rike and DelBalzo, Proc. IEEE Oceans (2003)] was developed to rapidly optimize placement of a few XBTs and to maximize MODAS accuracy. This work involves high-density, *in situ* data assimilation into MODAS to create a ground-truth temperature field from which a ground-truth transmission loss field was computed. Optimal XBT location sets were chosen by SPOTS, based on original MODAS uncertainties, and additional sets were chosen, based on subjective choices by an oceanographer. For each XBT set, a MODAS temperature nowcast and associated transmission losses were computed. This work discusses the relationship between temperature uncertainty, temperature error, and acoustic error for the objective SPOTS approach and the subjective oceanographer approach. The SPOTS approach allowed significantly more accurate acoustic calculations, especially when few XBTS were used. [Work sponsored by NAVAIR.]

11:00

4aUWb12. Modeling mid-to-high frequency acoustic signal fluctuations induced by fetch limited sea surface roughness in shallow water. Robert Heitsenrether and Mohsen Badiey (College of Marine Studies, Univ. of Delaware, Newark, DE 19711, rheits@udel.edu)

Surface waves are among several environmental parameters that significantly influence mid-to-high frequency (1–18 kHz) acoustic wave propagation. In coastal regions, surface waves are fetch limited with reduced spectral level at lower frequencies. In order to assess the detail of an acoustic signal interaction with the sea surface in such regions, a combined approach based on experimental observation and modeling of both surface waves and acoustic waves has been adopted. Data from two broadband shallow water acoustic experiments are presented. These data include simultaneous wind speed and acoustic propagation measurements. The experimental design allowed an examination of received signals corresponding to single surface bounced ray paths. Measured data analysis shows a high correlation between time–angle–intensity fluctuations of received signals and varying sea surface conditions. An empirical fetch-limited ocean wave spectrum has been combined with an acoustic ray-based model to study acoustic wave propagation. Rough sea surface realizations are generated and used as sea surface boundaries with the acoustic model. This combined sea surface/acoustic model predicts the variability of acoustic signal fluctuations as a function of varying sea surface. Modeled time–angle–intensity signal fluctuations compare excellently with field data at lower wind speeds.

11:15

4aUWb13. Narrowband signals propagation in randomly inhomogeneous shallow water waveguide. Boris Katsnelson, Sergey Pereselkov (Voronezh Univ., 1, Universitetskaya sq., Voronezh 394006, Russia), and Valery Petnikov (General Phys. Inst., Moscow 113000, Russia)

In the presented work results of experiment and theoretical modeling are considered for the narrowband (ratio of frequency band to frequency ~ 0.1) sound propagation at the long range (up to 200 km) acoustic track in Barents sea. Modes selection for these conditions can be used for acoustical tomography of large-scale perturbations. Characteristics of the sound signals passing through this area (and in turn feasibility of tomographic methods) are determined by mutually competing mechanisms: waveguide dispersion, providing modes filtering, modes attenuation and modes coupling due to random inhomogeneities, masking separation of modes. Experimental results presented in this work show significant influence of noise (or modes coupling). For interpretation of experimental results (frequency and time dependencies of arriving pulses) background internal waves are taken as a main reason of the modes coupling. It is shown that for the modes separation it is necessary to use vertical array along with the measuring difference in arrival time, and correlation processing of received signals. [Work supported by RFBR, grants Nos. 03-05-64568 and 02-02-16509.]

11:30

4aUWb14. Recent developments in underwater acoustic modeling. Paul C. Etter (Northrop Grumman Corp., Baltimore, MD 21203)

This is the fourth paper in an ongoing series of research reviews presented at eight-year intervals [P. C. Etter and R. S. Flum, J. Acoust. Soc. Am. Suppl. 1 **65**, S42 (1979); P. C. Etter, *ibid.* **82**, S102 (1987); J. Acoust. Soc. Am. **97**, 3312 (1995)]. This review of international developments in underwater acoustic modeling reveals an inventory containing 114 propagation models, 17 noise models, 17 reverberation models and 26 sonar performance models representing a 38% increase over the 1995 inventory. As in previous reviews, older models have been retained to provide an historical perspective that enables researchers to pose well-informed questions regarding the need for future developments. When executed in higher-level simulations, these models generate predictive and diagnostic outputs that are useful to sonar technologists or acoustical oceanographers in the analysis of complex systems operating in the undersea environment. Recent applications of underwater acoustic models in naval operations, offshore industries and oceanographic research will be discussed. This review coincides with the appearance of the third edition of *Underwater Acoustic Modeling and Simulation* (Spon Press, London, 2003), which provides a detailed account of research conducted over the past four decades.

Session 4pAA

Architectural Acoustics and Noise: Acoustic Design of Government Buildings

David E. Marsh, Chair
Pelton Marsh Kinsella, 1420 West Mockingbird Lane, Suite 400, Dallas, Texas 75247-4932

Chair's Introduction—2:00

Invited Papers

2:05

4pAA1. Acoustical considerations for secondary uses of government facilities. Jack B. Evans (JEAcoust., 5806 Mesa Dr., Ste. 380, Austin, TX 78731, Evans@JEAcoust.com)

Government buildings are by their nature, public and multi-functional. Whether in meetings, presentations, documentation processing, work instructions or dispatch, speech communications are critical. Full-time occupancy facilities may require sleep or rest areas adjacent to active spaces. Rooms designed for some other primary use may be used for public assembly, receptions or meetings. In addition, environmental noise impacts to the building or from the building should be considered, especially where adjacent to hospitals, hotels, apartments or other urban sensitive land uses. Acoustical criteria and design parameters for reverberation, background noise and sound isolation should enhance speech intelligibility and privacy. This presentation looks at unusual spaces and unexpected uses of spaces with regard to room acoustics and noise control. Examples of various spaces will be discussed, including an atrium used for reception and assembly, multi-jurisdictional (911) emergency control center, frequent or long-duration use of emergency generators, renovations of historically significant buildings, and the juxtaposition of acoustically incompatible functions. Brief case histories of acoustical requirements, constraints and design solutions will be presented, including acoustical measurements, plan illustrations and photographs. Acoustical criteria for secondary functional uses of spaces will be proposed.

2:30

4pAA2. Acoustical and electronic media design aspects of the Texas Capitol restoration. Kenneth Dickensheets (Dickensheets Design Assoc., 12335 Hymeadow Dr., Ste. 200, Austin, TX 78750, ken@dickensheets.com)

A presentation of the studies involved in preparation for the restoration of the Texas State Capitol building and the implementation of acoustical, noise and vibration control and electro-acoustical recommendations to ensure that legislative bodies' and committees' deliberations would be heard while allowing for faithful historical restoration of the facility. Specific attention to and examples in the Senate Chamber will be presented. Solutions to difficult noise and acoustical issues in the context of historical preservation will be discussed. Historical and present photos will be shown.

2:55

4pAA3. The noise quandary of "green" light. Dorie A. Najolia (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138, dnajolia@acentech.com)

In recent years, the government has striven to build more sustainable and efficient facilities by introducing the LEED (Leadership in Energy and Environmental Design) rating system and establishing acoustical privacy standards. However, there are some LEED design credits that conflict with the acoustical goals. This paper reviews the issues that arise when architects attempt to accommodate daylighting and acoustical privacy in their designs for government buildings. In achieving LEED certification, points are earned by designs that introduce daylight throughout occupied spaces in a building. Creating a "line of sight" from exterior glazing to interior spaces generally involves using transom, clerestory, side lite windows, or entire glass walls. At the same time, acoustical goals require construction around these spaces to achieve moderate to high levels of sound isolation. Another conflict is that although sound data is available for glass, very little research is available to demonstrate the effect lightweight aluminum framing has on the entire wall system. Incorporating these standards so that they provide the desired lighting and acoustical effects in government buildings needs to be better understood, since achieving both can add cost and complexity to the building design. Alternatively, compromises can be made, if they are understood and planned.

3:20–3:35 Break

3:35

4pAA4. Acoustical design issues in justice facilities. Howard K. Pelton and Ted N. Carnes (Pelton Marsh Kinsella, 1420 W. Mockingbird Ln., Ste. 400, Dallas, TX, peltonhk@c-b.com)

This paper presents various acoustical design challenges experienced by the authors with regard to courtrooms and jail facilities. The challenges in courtrooms include avoiding the sound focusing tendencies of concave curved surfaces and the general preference by courtroom designers for hard (i.e., sound reflecting) surface material. Jail facilities have attempted in recent years to incorporate sound absorbing surfaces to reduce noise, but this is difficult to achieve with durable materials that also must meet the security requirements of such facilities. Another interesting challenge is mitigating the intrusive noise of industrial grade toilets flushing when they are located in holding tanks adjacent to courtrooms.

4:00

4pAA5. From the bank, through the cafeteria line, and on to council chambers—Do the acoustics right! R. Bob Adams (Hoover & Keith, Inc., 11391 Meadowglen, Ste. D, Houston, TX 77082)

A common characteristic of municipal governments is the re-use of existing facilities. When a building's original use and acoustical design are dissimilar to the new acoustical needs, the project can be a challenge. The City of San Antonio Council Chamber was built as a bank in the 1930's and is considered an architectural landmark. This case study of the Council Chamber discusses the acoustical problems observed with intelligibility, focusing, and general ambiance from the highly ornate ceiling, internal arches, and an overlooking mezzanine. Primarily used for open City Council meetings, the room is also used as a "studio" for broadcast of other city related issues. The presentation will include photographs identifying the historical aspects of the architecture as well as portions of the EASE model created for the project.

4:25

4pAA6. A new seamless, smooth, interior, absorptive finishing system. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774)

Government architecture typically employs classic forms of vaults, domes and other focusing or reflective shapes, usually created with hard materials like concrete and plaster. The use of conventional porous absorption is typically rejected as an acoustical surface material for aesthetic reasons. Hence, many of these new and existing facilities have compromised speech intelligibility and music quality. Acousticians have sought a field-applied, absorptive finishing system that resembles a smooth plaster or painted drywall surface, since the dawn of architectural acoustics. Some success has been achieved using sprayed cellulose or cementitious materials, but surface smoothness has been a challenge. A new approach utilizing a thin microporous layer of mineral particles applied over a mineral wool panel will be described. This material can be applied to almost any shape surface, internally pigmented to match almost any color and renovated. Because of these unique characteristics the new seamless, absorptive, finishing system is being specified for many new and renovated spaces. Application examples will be presented.

4:50

4pAA7. Multipurpose council chambers "in the round" poses acoustical challenges. Edward L. Logsdon (D. L. Adams Assoc., 1701 Boulder St., Denver, CO 80211)

The City of Aurora Council Chambers is used for both municipal and public meetings. The room is configured to provide close-in seating with good sightlines from each of the 300 stadium-style seats. Presentations can be made from the central podium location to either the audience or council dais requiring multiple loudspeaker zoning and control. The cylindrical ceiling, shaped to accommodate video projection and lighting equipment, is acoustically treated to eliminate late reflections. The City Council meetings are broadcast to public TV on a regular basis from this room requiring good room acoustics and sound isolation to reduce echo and achieve acceptably low background noise levels while satisfying the aesthetic palette of the interior designers. A case history will be presented along with photographs showing how specialty wood materials, both absorptive and diffusive, were incorporated along with absorptive plaster and cloth-covered fiberglass panels into the design of the building.

Session 4pABa

Animal Bioacoustics and Signal Processing in Acoustics: Medical Imaging Techniques to Understand Auditory Processing

Darlene R. Ketten, Chair

Biology Department, Marine Sensory Systems Group, Woods Hole Oceanographic Institution, Woods Hole, Massachusetts 02543

Chair's Introduction—1:30

Invited Papers

1:40

4pABa1. Hair cells in motion: Imaging the organ of Corti. David C. Mountain and K. Domenica Karavitaki (Hearing Res. Ctr., Boston Univ., 44 Cummington St., Boston, MA 02215)

The mammalian cochlea contains two types of sensory cells, inner hair cells (IHCs) and outer hair cells (OHCs). The IHCs provide the vast majority of the synaptic input to the auditory nerve while the OHCs express a unique motor protein, prestin, and appear to participate in an electromechanical feedback loop that amplifies the motion of the organ of Corti (OC). To study this amplification process we have employed stroboscopic video microscopy to quantify the motion of various elements of the OC. Extracellular electrical stimulation was used to excite OHC motility and a computer-controlled high-intensity light-emitting diode (LED) is used to illuminate the organ OC in an excised cochlear preparation. Motion is measured by extracting small regions of interest (ROIs) from the images and cross-correlating the ROIs taken during electrical stimulation with a reference image from the same ROIs taken with no stimulation. The observed motion is quite complex with several vibration modes observed. One of the major findings is that there appears to be oscillatory fluid flow within the tunnel of Corti suggesting that the OHC contractions are pumping fluid longitudinally within the organ. [Work funded by NIDCD.]

2:05

4pABa2. Multimodal imaging of the human temporal bone: A comparison of CT and optical scanning techniques. Arne H. Voie (Spencer Technologies, 701 16th Ave., Seattle, WA 98122, voie@spencertechnologies.com), Bruce Whiting, Margaret Skinner, J. Gail Neely, Kenneth Lee, Tim Holden, and Barry Brunsden (Washington Univ. School of Medicine, St. Louis, MO 63110)

A collaborative effort between Washington University in St. Louis and Spencer Technologies in Seattle, WA has been undertaken to create a multimodal 3D reconstruction of the human cochlea and vestibular system. The goal of this project is to improve the accuracy of *in vivo* CT reconstructions of implanted cochleae, and to expand the knowledge of high-resolution anatomical detail provided by orthogonal-plane optical sectioning (OPFOS). At WUSL, computed tomography (CT) images of the cochlea are used to determine the position of cochlear implant electrodes relative to target auditory neurons. The cochlear implant position is determined using pre- and post-operative CT scans. The CT volumes are cross-registered to align the semicircular canals and internal auditory canal, which have a unique configuration in 3-D space. The head of a human body donor was scanned with a clinical CT device, after which the temporal bones were removed, fixed in formalin and trimmed prior to scanning with a laboratory Micro CT scanner. Following CT, the temporal bones were sent to the OPFOS Imaging Lab at Spencer Technologies for a further analysis. 3-D reconstructions of CT and OPFOS imaging modalities were compared, and results are presented. [Work supported by NIDCD Grants R44-03623-5 and R01-00581-13.]

2:30

4pABa3. Functional brain imaging and bioacoustics in the Bottlenose dolphins, *Tursiops truncatus*. Sam Ridgway, James Finneran, Donald Carder, William Van Bonn, Cynthia Smith (U.S. Navy Marine Mammal Prog., Space and Naval Warfare Systems Ctr., San Diego, 53560 Hull St., San Diego, CA 92152-5001), Dorian Houser (Biomimetica, San Diego, CA), Robert Mattrey, and Carl Hoh (Univ. of California, San Diego, CA 92093)

The dolphin brain is the central processing computer for a complex and effective underwater echolocation and communication system. Until now, it has not been possible to study or diagnose disorders of the dolphin brain employing modern functional imaging methods like those used in human medicine. Our most recent studies employ established methods such as behavioral tasks, physiological observations, and computed tomography (CT) and, for the first time, single photon emission computed tomography (SPECT), and positron emission tomography (PET). Trained dolphins slide out of their enclosure on to a mat and are transported by trainers and veterinarians to the laboratory for injection of a ligand. Following ligand injection, brief experiments include trained vocal responses to acoustic, visual, or tactile stimuli. We have used the ligand technetium (Tc-99m) biscisate (Neurolite) to image circulatory flow by SPECT. Fluro-deoxy-d-glucose (18-F-FDG) has been employed to image brain metabolism with PET. Veterinarians carefully monitored dolphins during and after the procedure. Through these methods, we have demonstrated that functional imaging can be employed safely and productively with dolphins to obtain valuable information on brain structure and function for medical and research purposes. Hemispheric differences and variations in flow and metabolism in different brain areas will be shown.

2:55

4pABa4. Extreme variations in skull density of toadfish, *Opsanus ta*. Peggy Edds-Walton (Neurosci. Inst., Marine Biological Lab., Woods Hole, MA 02543) and Darlene Ketten (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Gross observations of the skull in sexually mature toadfish revealed variations in bone density that appear to be most extreme in the otic capsule. Four male and one female toadfish (17–30.5 cm SL) were scanned to obtain images formatted at 100 μm and 1 mm. Consecutive measurements were made from caudal of the otic capsule to the rostral edge of the otic capsule. Attenuation values were recorded from three sites on the skull (left parietal bone, parietal suture, ventral surface of the otic capsule) and along two otoliths (calcareous, associated with the sensory epithelia of the ear) within the otic capsule. In all five fish, the parietal suture had the highest attenuation. Attenuation values for the parietal bone varied with size, indicating increasing density with growth. Among all five fish, the lowest attenuations were obtained for the ventral wall of the otic capsule, with values similar to those of cartilage. In addition, the minimum values were found ventral to the saccular otolith. Given that the saccule is the primary auditory endorgan in this species [Edds-Walton *et al.*, J. Comp. Neurol. **411**, 212–238 (1999)], the co-occurrence of bone thinning in this area of the skull may have functional significance related to audition.

Contributed Paper

3:20

4pABa5. Structural (CT) and functional imaging (PET/SPECT) for the investigation of dolphin bioacoustics. Dorian S. Houser (Biomimetica, 5750 Amaya Dr., Ste. 24, La Mesa, CA 91942), James J. Finneran (U.S. Navy Marine Mammal Prog., Space and Naval Warfare Systems Ctr., San Diego, CA), Robert Mattrey, Carl Hoh (School of Medicine, Univ. of California, San Diego, CA), and Sam Ridgway (U.S. Navy Marine Mammal Prog., Space and Naval Warfare Systems Ctr., Univ. of California, San Diego, CA)

A combination of imaging modalities was used to address physiological and anatomical questions relevant to dolphin bioacoustics. Three dolphins (*Tursiops truncatus*) were scanned with CT to investigate *in vivo* dolphin cranial anatomy. One dolphin underwent SPECT and PET scanning to investigate blood flow and metabolic activity of the cranial tissues. Air spaces were mostly contiguous and covered the periotic bone and auditory bulla dorsally and medially. Cranial air was compartmentalized by the nasal plug and constriction of the palatopharyngeus muscle. Blood flow, determined from SPECT imaging of ^{99}Tc-bicisate distribution, was greatest in the brain, melon, and posterior fats of the lower jaw. Metabolic activity of tissues, assessed by monitoring the uptake of ^{18}F-deoxyglucose via PET, indicated that melon and jaw fats were metabolically inert compared to the brain. Nasal cavity and sinus air volume that is reduced during diving may be replenished with lung air via the palatopharyngeus and Eustachian tube. Air covering the bulla may protect the ears from outgoing echolocation pulses and contribute to spectral and time of arrival cues. Blood flow to the melon and lower jaw fats may serve to either regulate the temperature of acoustic lipids or act as a site of counter-current heat exchange.

THURSDAY AFTERNOON, 13 NOVEMBER 2003 GUADALUPE ROOM, 3:45 TO 5:05 P.M.

Session 4pABb

Animal Bioacoustics: Topics in Animal Bioacoustics

Darlene R. Ketten, Chair
Biology Department, Marine Sensory Systems Group, Woods Hole Oceanographic Institution, Woods Hole, Massachusetts 02543

Chair's Introduction—3:45

Contributed Papers

3:50

4pABb1. Pure-tone audiograms and hearing loss in the white whale (*Delphinapterus leucas*). James J. Finneran, Donald A. Carder (U.S. Navy Marine Mammal Prog., Space and Naval Warfare Systems Ctr., San Diego, Code 2351, 53560 Hull St., San Diego, CA 92152-5001), Randall Dear (Sci. Applications Intl. Corp., San Diego, CA 92110), Traci Belting (Point Defiance Zoo and Aquarium, Tacoma, WA 98466), and Sam H. Ridgway (Space and Naval Warfare Systems Ctr., San Diego, CA 92152-5001)

A behavioral response paradigm was used to measure pure-tone audiograms for two white whales (*Delphinapterus leucas*). Tests were conducted over a 20 month period at the Point Defiance Zoo and Aquarium, in Tacoma, Washington. Subjects consisted of two males, aged 8–10 and 9–11 during the course of the study. Subjects were born in an oceanarium and had been housed together for all of their lives. Hearing thresholds were measured using a modified up/down staircase procedure and acoustic response paradigm where subjects were trained to whistle in response to hearing test tones and to remain quiet otherwise. Test frequencies ranged from approximately 2 to 130 kHz. Best sensitivities ranged from 40 to 50 dB *re*: 1 Pa. Both subjects had traditional U-shaped mammalian audiograms; however, one subject exhibited significant high-frequency hearing loss, above approximately 37 kHz. The experimental setup and procedure will be presented and the measured hearing thresholds compared to those previously measured in white whales. The potential role of ototoxic antibiotics in the observed hearing loss will be discussed. [Work supported by ONR Marine Mammal S&T Program and the U.S. Navy CNO(N45).]

4:05

4pABb2. Detection and classification of right whales in the Bay of Fundy using independent component analysis. Michael Linford and Brian La Cour (Appl. Res. Labs., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78758)

A novel method of the detection and classification for marine mammals is presented which uses techniques from independent component analysis to solve the blind source separation problem for right whales in the Bay of Fundy. Using the fundamentally non-Gaussian nature of marine mammal vocalizations and data collected on multiple hydrophones, we are able to separate right whale source spectra, up to an unknown scale, from ambient noise. This technique assumes that the array data is a linear combination of the source signals but does not require specific knowledge of the array geometry. A detector/classifier algorithm is demonstrated which compares the estimated source spectra against known right whale vocalizations.

4:20

4pABb3. Dolphin echolocation strategies studied with the Biosonar Measurement Tool. Dorian S. Houser (Biomimetica, 5750 Amaya Dr., Ste. 24, La Mesa, CA 91942), Steve W. Martin, Michael Phillips, Eric Bauer, and Patrick W. Moore (U.S. Navy Marine Mammal Prog., Space and Naval Warfare Systems Ctr., San Diego, CA)

Two free-swimming dolphins (Tt722 and Tt673) were trained to carry the Biosonar Measurement Tool (BMT) during open water, proud target searches in order to explore echolocation behavior without the constraints of traditional experimental designs. The BMT recorded the angular motion, depth, and velocity of the dolphin as well as echolocation clicks and echoes returning from insonified targets. Mean search time for Tt722 was 24.6 ± 7.3 s and 6.5 ± 3.0 s for Tt673 on target present trials, the former strategy resulting in the lower false alarm rate. The majority of clicks exceeded 195 dB *re*: 1 μPa throughout all trials for both animals but each demonstrated preferences for particular frequency bands of echolocation. Considering all trials, only 3.6% of all clicks produced by Tt722 contained peak frequencies greater than 60 kHz whereas Tt673 produced clicks with peak frequencies above 60 kHz 20.4% of the time. Distinctive frequency bands in the distribution of clicks were notable: bands for Tt673 occurred at 38, 54, and 69 kHz with less defined higher order bands; bands for Tt722 occurred at 25, 35, and 40 kHz. Distinctive frequency bands suggest a preferential use or mechanical constraint on harmonically related click frequencies.

4:35

4pABb4. Frameless processing of bird songs using set membership identification. Rafi Mohammad, Mark M. Wilde, and Dale Joachim (Dept. of Elec. Eng. and Computer Sci., Tulane Univ., 211 Stanley Thomas Hall, New Orleans, LA 70118, joachimd@eecs.tulane.edu)

This paper describes a novel approach to feature extraction from bird songs using a set-membership identification (SMI) algorithm. The low computational complexity of the SMI algorithm allows frameless pointwise feature estimation and real-time processing. Both energy-based endpoint detection and set-to-point classification methods are incorporated in this SMI processing for enhanced labeling performance. The described algorithm serves as a front end to a fully automated bird identification system in which training data collection is automated by scheduled computer generated phone calls to a cellular monitoring station. RASTA processing of feature vectors compensates for the telephone channel effects.

4:50

4pABb5. Identification and assessment of constrictions in a branched tube system. Youhua Du and Ahmed M. Al-Jumaily (Diagnostics and Control Res. Ctr., Auckland Univ. of Technol., Auckland, New Zealand)

This paper focuses on developing an analytical methodology, based on the frequency spectrum of the input acoustic impedance, to identify and assess the location and severity of constrictions in a branched-treelike tube system. The method is applied to two models, one is for the trachea with the first-generation asymmetric bronchi and the other is for the trachea with two-generation bronchi. To develop the methodology, a different size constriction is introduced at various locations along the branches and resonant frequencies are analyzed. Correlation is performed between the resonant frequencies and the location and size of these constrictions. For the two models, charts are generated which can be used to determine the location and size of a constriction just by observing the trend of various resonances. For the first model it is shown that in the frequency range of 1–4000 Hz the third and sixth resonances reflect constriction occurrence in the larger branch and the fourth and seventh reflect constriction in the smaller branch. However, for the second model it is indicated that constriction is reflected: in first branch by the sixth resonance, in the second branch by the seventh resonance, and in the third branch by the ninth resonance.

THURSDAY AFTERNOON, 13 NOVEMBER 2003 TRINITY A ROOM, 2:30 TO 4:35 P.M.

Session 4pBB

Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Tissue Harmonic Imaging

Robin O. Cleveland, Chair

Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215

Invited Papers

2:30

4pBB1. Nonlinear acoustics in tissue harmonic imaging. Michalakis Averkiou (Philips Medical Systems, P.O. Box 3003, Bothell, WA 98041-3003)

In recent years the interest in nonlinear acoustics has dramatically increased in diagnostic ultrasound. There are two main areas where nonlinear acoustics is used in medical imaging: tissue harmonic imaging (THI) and imaging of ultrasound contrast agents. Although similar approaches are used in both of these areas, they are very different in that THI is based on nonlinear propagation of sound in tissue, whereas contrast imaging is based on the nonlinear scattering from resonant microbubbles. The clinical benefits of THI are reduced phase aberration artifacts and overall clutter, improved border delineation, and increased contrast resolution. The basic principles of nonlinear propagation of sound beams in tissue are discussed here. Theoretical and experimental results are used to demonstrate some of the properties of nonlinear propagation and their relation to imaging. To a large extent the clinical benefits of THI are explained with the nonlinear propagation properties. Imaging considerations and techniques like pulse inversion and power modulation are discussed. These techniques are also used in imaging contrast agents (microbubbles) and an effort is made to separate the various issues in these two imaging areas and explain their differences.

2:55

4pBB2. Physics of tissue harmonic imaging. Thomas L. Szabo (Boston Univ., 110 Cummington St., Boston, MA 02215, tlszabo@bu.edu)

Tissue harmonic imaging is used on almost all modern diagnostic ultrasound equipment. The clinical success of this modality involves a combination of fortuitous factors. The improvement in lateral resolution obtained by transmitting on a fundamental frequency and receiving on a harmonic is well known. Other advantages include a deeper focal range with less absorption than would have been obtained by transmitting and receiving at twice the fundamental frequency. Natural apodization considerably reduces the 3-D volume of interaction with tissue. Simulations indicate that harmonic imaging is more robust in the presence of aberration. Analysis shows that the improvements over what would have been achieved by transmitting at twice the frequency are for shallower focal depths. Reasons for the noticeably enhanced contrast of harmonic imaging are discussed.

Contributed Papers

3:20

4pBB3. Phase conjugation of the second harmonic of a focused ultrasound beam as a method for improving C-scan acoustical imaging in nonlinear inhomogeneous media. Leonid M. Krutyansky, Andrew P. Brysev, Roman V. Klopotov (Wave Res. Ctr., General Phys. Inst., Russian Acad. of Sci., 38 Vavilov Str., Moscow 119991, Russia, krut@orc.ru), Philippe J. Pernod, Vladimir L. Preobrazhensky (Institut d'Electronique, de Microelectronique et de Nanotechnologies, IEMN-DOAE, UMR CNRS 8520, Villeneuve d'Ascq 59651, France), Xiang Yan, and Mark F. Hamilton (Univ. of Texas, Austin, TX 78712-1063)

Acoustical imaging in complex media (e.g., biological tissue) can be affected by phase aberrations introduced in a wave during propagation. Wave phase conjugation (WPC) of ultrasound is known for its ability to compensate for phase distortions due to inhomogeneity of the propagation medium, and it can be used for improvement of acoustical imaging under these conditions. In a nonlinear medium harmonics are generated during propagation of an intense beam of ultrasound, and this principle is used in tissue harmonic imaging. The parametric method of WPC permits phase conjugation of a selected frequency component of the probe beam. In this way the peculiarities of WPC can be combined with advantages of harmonic imaging. Automated WPC-focusing of the conjugated second-harmonic component of a focused nonlinear probe beam is studied experimentally and theoretically for the case of a homogeneous medium, and experimentally for a medium with pseudo-random inhomogeneities. The generated conjugate wave can also be sufficiently intense to generate higher-order harmonics, which display enhanced focusing. Improvement of a C-scan harmonic imaging system operating in an inhomogeneous medium is provided as an example.

3:35

4pBB4. Effect of aberration on the acoustic field in tissue harmonic imaging (THI). Yuan Jing and Robin Cleveland (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215, yuanjing@bu.edu)

A numerical simulation was used to study the impact of an aberrating layer on the generation of the fundamental and second-harmonic (SH) field in a tissue harmonic imaging scenario. The simulation used a three-dimensional time-domain code for solving the KZK equation and accounted for arbitrary spatial variations in all acoustic properties. The aberration effect was modeled by assuming that the tissue consisted of two layers where the interface has a spatial variation C that acted like an effective phase screen. Initial experiments were carried out with sinusoidal-shaped interfaces. The sinusoidal interface produced grating lobes which were at least 6 dB larger for the fundamental signal than the SH. The energy outside of the main lobe was found to increase linearly as the amplitude of the interface variation increased. The location of the grating lobes was affected by the spatial period on the interface variation. The inhomogeneous nature of tissue was modeled with an interface with a random spatial variation. With the random interface the average sidelobe level for the fundamental was -30 dB whereas the SH had an average sidelobe level of -36 dB. [Work supported by the NSF through the Center for Subsurface Sensing and Imaging Systems.]

3:50

4pBB5. New formulation of the elastic energy density for soft tissue. Mark F. Hamilton, Yurii A. Ilinskii, and Evgenia A. Zabolotskaya (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

Measurements of the elastic constants in Landau's expansion of the strain energy density reported by Catheline *et al.* for a soft tissue phantom reveal that they differ by five orders of magnitude [J. Acoust. Soc. Am. **112**, 2404 (2002)]. A more appropriate expansion would possess constants of the same order. Landau's third-order expansion of the elastic energy density is extended to fourth order. The corresponding elastic constants are related to those for liquids. The principal contribution is an alternative formulation of the energy density that permits separation of effects due to compressibility and shear deformation. To fourth order its expansion is $\mathcal{E} \simeq \mathcal{E}_0(\rho) + \mu I_2 + \frac{1}{3}AI_3 + DI_2^2$, where ρ is density, I_2 and I_3 are the second- and third-order Lagrangian strain invariants used by Landau, μ is the shear modulus, A is one of Landau's third-order elastic constants, and D is a new fourth-order elastic constant. For processes involving mainly compressibility $\mathcal{E} \simeq \mathcal{E}_0(\rho)$, and for processes involving mainly shear deformation $\mathcal{E} \simeq \mu I_2 + \frac{1}{3}AI_3 + DI_2^2$. Comparison with the parameters in Mooney's potential function for incompressible elastic media shows the three constants to be of the same order. [Work supported by the IR&D Program at ARL:UT.]

4:05

4pBB6. Numerical modeling of Harmonic Imaging and Pulse Inversion fields. Victor F. Humphrey, Tracy M. Duncan (Dept. of Phys., Univ. of Bath, Bath BA2 7AY, UK), and Francis Duck (Royal United Hospital, Bath BA1 3NG, UK)

Tissue Harmonic Imaging (THI) and Pulse Inversion (PI) Harmonic Imaging exploit the harmonics generated as a result of nonlinear propagation through tissue to improve the performance of imaging systems. A 3D finite difference model, that solves the KZK equation in the frequency domain, is used to investigate the finite amplitude fields produced by rectangular transducers driven with short pulses and their inverses, in water and homogeneous tissue. This enables the characteristic of the fields and the effective PI field to be calculated. The suppression of the fundamental field in PI is monitored, and the suppression of side lobes and a reduction in the effective beamwidth for each field are calculated. In addition, the differences between the pulse and inverse pulse spectra resulting from the use of very short pulses are noted, and the differences in the location of the fundamental and second harmonic spectral peaks observed.

4:20

4pBB7. Statistical investigation of beam distortion by tissue inhomogeneity in tissue harmonic imaging. Xiang Yan and Mark F. Hamilton (Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712-1063)

For many patients, tissue harmonic imaging improves resolution by reducing phase distortion due to acoustic propagation through inhomogeneities in the body wall, reverberation in this layer, and artifacts due to sidelobes. Our investigation focuses on the first of these phenomena. A statistical analysis was conducted to quantify the improvement provided by tissue harmonic imaging in the presence of an inhomogeneous layer in close proximity to the source. The inhomogeneity is modeled as a phase screen of zero thickness located directly in front of the source. The fundamental beam and resulting second-harmonic generation are described in the parabolic approximation. The phase aberration is assumed to have zero mean and to be Gaussian correlated in space. For a source with Gaussian amplitude distribution, a solution requiring a double numerical integration was derived for the mean intensity of the second-harmonic beam profile in the target plane. The solution reveals the separate contributions due to the undistorted second-harmonic and the scattered component due to the inhomogeneity. The statistical solution is validated by comparison with ensemble averages of direct numerical simulations. Experiments were also conducted, the results of which confirm the advantages of tissue harmonic imaging. [Work supported by ONR.]

THURSDAY AFTERNOON, 13 NOVEMBER 2003 BRAZOS AUDITORIUM, 2:00 TO 5:00 P.M.

Session 4pMU

Musical Acoustics: Honoring the Contributions of Gabriel Weinreich II: Violin Acoustics

Uwe J. Hansen, Chair

Department of Physics, Indiana State University, Terre Haute, Indiana 47809

Invited Papers

2:00

4pMU1. The role of vibrato in the perception of violin quality. Colin E. Gough (School of Phys. and Astron., Univ. of Birmingham, Birmingham B15 2TT, UK)

The role of vibrato in the characterization of violin tone will be considered from both a physical acoustics and perceptual viewpoint. Musical examples of individual waveforms from violins of widely different qualities will be used to demonstrate the importance of vibrato and other temporal fluctuations in the recognition of the violin as a specific musical instrument and, by inference, of its perceived tonal quality also. Dynamic physical models will be introduced to describe the waveforms of sounds produced by the bowed string when vibrato is used. The sound produced by an individual instrument will be shown to be critically dependent on both the positions and the Q values of the acoustically important resonances excited, in addition to random noise generated by the bowing action. It will be argued that the temporal fluctuations in the sound of a violin played with vibrato are just as important as the temporal fluctuations associated with the very strong frequency-dependence of the directivity of the violin. Weinreich has emphasized the importance of the latter effect in any realistic reproduction of the sound of a violin.

2:30

4pMU2. Physics of the violin. Erik V. Jansson (Dept. of Speech, Music and Hearing, KTH, Dr. Kristinas vag 31, SE-Stockholm, Sweden)

A method to measure acoustical properties of the violin has been developed. The bridge is excited by an impulse force hammer (see www.speech.kth.se/music/acviguit4). Bridge vibrations are recorded by a small magnet and an electrical coil. Measurements can be made in an ordinary room and give a record of properties built into the violin body. An old good Polish violin, B Dankwart, Vilnius *ca*. 1600 shows typical results with peaks P1, P2 and the BH-hill. Our goal is as suggested by Gabriel Weinreich "to understand, not to copy,"—Stradivarius. By shifting the soundpost position the peaks P1, P2 and BH can be somewhat monitored. It can be shown that the BH is not confined to the bridge only. The feet distance of the bridge is important but also the top plate. Marcin Groblicz, the great Polish violin maker, was court instrument maker in Krakow *ca*. 1600 to Sigismund, King of Poland and Sweden. I believe Gabriel Weinreich has some connection to Vilnius in Poland but unfortunately not to Sweden. We like Gabi.

4p THU. PM

3:00

4pMU3. Bow response to the string it is bowing. Robert T. Schumacher (Dept. of Phys., Carnegie Mellon Univ., Pittsburgh, PA 15213)

The force that a bowed string exerts on the bow can be reconstructed from the forces on the string's termination. By using one or more accelerometers attached to the bow, the response of the bow to that reconstructed force can be recorded. Bow responses will be shown that were obtained using a bowing machine. At some harmonic frequencies of the string the signal from the longitudinal standing wave excited on the bow hair allows determination of the velocity of propagation of that wave. The consequence of the variable excitation of various harmonics of the standing wave as the bow stroke progresses is a rapidly changing excitation of the bows various normal modes. Three data channels, requiring at least a 128 kHz sampling rate for E-strings, are required for bow force reconstruction and motion detection of the bow. The validity of substituting the force of the string on the bridge instead of the reconstructed bow force is explored. That allows a reduction of the minimum number of data channels from three to two, a reduction in the necessary sampling rate, and use of a bow stroke by a player instead of a bowing machine.

3:30

4pMU4. Modal analysis of violin bodies viewed as three-dimensional structures. Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115), Nils-Erik Molin, and Anna Runnemalm (Luleå Univ. of Technol., Luleå, Sweden)

Modal analyses of violins show several strong modes in the low frequency range. Holographic interferograms suggest that four strong modes can be interpreted as doublets having two and three nodal planes that intersect a cylinder with a roughly elliptical cross section at the bridge [A. Runnemalm, N.-E. Molin, and E. Jansson, J. Acoust. Soc. Am. **107**, 3452–3459 (2000); M. Roberts and T. D. Rossing, Catgut Acoust. Soc. J. **3**, 9–15 (1998)]. This is especially clear when the instrument is viewed simultaneously from three sides using mirrors, and the holographic system is made sensitive to in-plane motion as well. These doublets are not unlike those observed in cylindrical vibrators such as bells, and they remind us that a violin is a 3-dimensional object.

4:00

4pMU5. Unusual motions of a vibrating string. Roger J. Hanson (Dept. of Phys., Univ. of Northern Iowa, Cedar Falls, IA 50614, roger.hanson@cfu.net)

The actual motions of a sinusoidally driven vibrating string can be very complex due to nonlinear effects resulting from varying tension and longitudinal motion not included in simple linear theory. Commonly observed effects are: generation of motion perpendicular to the driving force, sudden jumps in amplitude, hysteresis, and generation of higher harmonics. In addition, these effects are profoundly influenced by wire asymmetries which in a brass harpsichord wire can cause a small splitting of each natural frequency of free vibration into two closely spaced frequencies (relative separation ~0.2% to 2%), each associated with transverse motion along two orthogonal characteristic wire axes. Some unusual resulting patterns of complex motions of a point on the wire are exhibited on videotape. Examples include: sudden changes of harmonic content, generation of subharmonics, and motion which appears nearly chaotic but which has a pattern period of over 10 s. Another unusual phenomenon due to entirely different causes can occur when a violin string is bowed with a higher than normal force resulting in sounds ranging from about a musical third to a twelfth lower than the sound produced when the string is plucked.

4:30

4pMU6. Sound radiation by violins radically modal. George Bissinger (Phys. Dept., East Carolina Univ., Greenville, NC 27858)

Following a radically modal path pioneered by Gabi Weinreich in the early 1980s, normal-mode vibration and radiation analysis was applied to quality-rated violins. Being able to describe the violin in terms of normal-mode behaviors leads directly to a violin equation, allows extraction of a critical frequency for the violin, provides estimates of the fraction of vibrational energy radiated and extends to modal-average trendline modeling of the acoustic output, all of which can be linked to violin quality classifications if desired. Some quality-related results mirror Weinreich's conclusions from inverse radiativity measurements on violins, i.e., there is little difference in the radiativity of good and bad violins below 1 kHz. The first corpus bending modes (the baseball modes), first identified by Gabi and Eric Arnold, are a major contributor to violin sound and highlight one of the important aspects of Gabi's research, the coupling between wood and air. The first near-field acoustic holography results indicate that a significant part of the radiation from these structural modes is from the f-holes of the violin. These normal-mode investigations reflect some of the importance of Gabi's work for modern violin acoustics. [Research supported by NSF.]

THURSDAY AFTERNOON, 13 NOVEMBER 2003 BOSQUE ROOM, 1:30 TO 2:45 P.M.

Session 4pNSa

Noise: Topics in Urban and Community Noise

Bennett M. Brooks, Chair

Brooks Acoustics Corporation, 96 Main Street, Talcottville, Connecticut 06066

Contributed Papers

1:30

4pNSa1. Noise impact caused by electrical energy substations in Curitiba, Brazil. Fabiano B. Diniz and Paulo T. Zannin (Environ. Acoust. Lab., Dept. of Mech. Eng., UFPR, Curitiba, PR, Brazil)

This survey is intended to characterize the noise impact due to electrical energy substations in the city of Curitiba over the population living in their vicinity. This impact has been studied with the aid of a computational tool capable of mapping the acoustical field of substations and their vicinity. Several factors have been considered in this survey: sound power of the transformers; vehicle flow on the surrounding roads; positioning of the firewalls, of the buildings and of the walls; terrain topography. Four substations have been analyzed, and an acoustical map has been traced for each of them. With these maps it was possible to visualize what was the incident noise level on the building facades. The predicted noise levels have been compared to the environmental legislation of the noise emissions in effect in the city.

1:45

4pNSa2. CTA "L" train outdoor noise propagation in Wrigley-ville. Matthew McDuffee (Columbia College Chicago, 1232 S. 12th St., St. Charles, IL 60174, mcduffula@hotmail.com)

It is well known to those who reside in Chicago that the elevated mass transit trains, known as the L, are the largest culprit of outdoor noise pollution in most neighborhoods throughout the city. Many potential residents and buyers will make their choice of where to live in the city simply based upon the building's relationship to the L. For those who live directly

next to the tracks vibration becomes a major issue. The study focused on the environmental noise impact of the CTA red line train traveling through Chicago's north side in the Wrigley-ville neighborhood. Approximately 1.5 miles of elevated track stretch between the Addison station (near Wrigley field) and the bridge that crosses Montrose Avenue. Eight locations were chosen throughout an area covering 17 city blocks surrounding the red line tracks to perform field measurements. The field measurements, taken in Leq, were correlated against a computer model designed using Cadna/A outdoor noise propagation software. The models polygons (representing buildings) were created using ESRI Arc View mapping software to view digitized aerial photographs of the north side of Chicago. The models predictions correlated to within plus or minus 1 dB on 6 out of 8 field measurements.

2:00

4pNSa3. Evaluation and prediction of noise pollution levels in urban areas of Cdiz (Spain). Silvia Rivas, Ricardo Hernandez, and Jose Luis Cueto (Acoust. Vib. Lab. C.A.S.E.M, Campus Ro San Pedro 11510, Puerto Real, Cdiz, Spain, silvia.rivas@uca.es)

In the European policy, one of the most important objectives is to achieve a high level of health and environmental protection. The latest studies have shown that more than 20% of the world population lives under unacceptable noise levels and near 60% of the European population is exposed to worrying noise levels during the day. So, nowadays one of the objectives to be pursued is the protection against noise, one of the main environmental problems in Europe. During the last 10 years different studies have been carried out in urban areas of Cdiz, in order to evaluate the noise pollution level and its management. Those studies exposed how the continuous development of legal device, more capable each time, might moderate the upper emission levels. Instead of it, the share of the medium levels of the population hold up and could be detrimental for its normal development, increasing progressively. It will be useful to develop more studies in this area, as we are doing in El Campo de Gibraltar to study those damaging levels and to predict their tendency, in order to establish and improve new methods of environmental noise protection.

2:15

4pNSa4. Analysis over the new data treatment method developed by the community directive 2002/49/ce about environmental noise management. Jose Luis Cueto, Silvia Rivas, and Ricardo Hernandez (Acoust. and Vib. Lab. C.A.S.E.M., Campus Ro San Pedro s/n 11510, Puerto Real, Cdiz, Spain)

Nowadays a European Directive develops useful procedures in order to evaluate, by means of index, the acoustic pollution affecting urban areas. To apply properly those methods, it is necessary to use a long data sequence. Is usual to find in this kind of series some peculiar measures or events that could change indicatively our acoustic pollution index data. So, we suggest a simple data treatment method as a first step in the way of developing a better process of acoustic index construction. This methodology would permit us to introduce the acoustic climate concept.

2:30

4pNSa5. Evaluation of noise pollution in urban parks of Curitiba, Brazil. Andressa M. Ferreira (Environ. Acoust. Lab., Dept. of Mech. Eng., UFPR, 81531-990, Curitiba, PR, Brazil), Fabiano B. Diniz, and Paulo T. Zannin (Environ. Acoust. Lab., UFPR)

A study about the noise pollution found in six urban parks of Curitiba, Paran, Brazil. The equivalent noise levels (Leq) have been measured in points spread throughout the park, and interviews have been conducted with some park visitors. It has been found out that 17.83% out of the measurement sites have presented Leq levels over 65 dB(A), considered by the Preventive Medicine as the maximum level one can be exposed to without being exposed to health impairment risks, and 52.48% out of the meas sites do not satisfy the municipal law 10 625, which states the noise emission level of 55 dB(A) as the limit value for green areas (AV). The results of the questionnaires applied to the local visitors have showed that 39% out of the interviewed people users to visit the park every 75% of them seek for the realization of a physical activity during the realization of their activities in the parks, 22% out of interviewed people point the noise pollution as the source of annoyance and 29% of them point the local security.

THURSDAY AFTERNOON, 13 NOVEMBER 2003 BOSQUE ROOM, 3:00 TO 3:45 P.M.

Session 4pNSb

Noise: Measurement and Modeling of Noise Related Environments

Brandon D. Tinianov, Chair

Acoustical Laboratory, Johns Manville, 10100 West Ute Avenue, P.O. Box 625005, Littleton, Colorado 80162

Contributed Papers

3:00

4pNSb1. A method to separate contribution from coherent and interfering inputs. Valeri V. Lenchine, Hoon Wee, Jae-Man Joo, and Sang-Kyong Oh (Samsung Electr. DA R&D Ctr., 416, Maetan-3Dong, Paldal-gu, Suwon City, Gyeonggi-do 442-742, Korea)

Acoustical development tasks frequently involve the identification of contribution from different sources. A conventional way to detect principal exciters is to separate signals that are completely incoherent between each other and interfering inputs. However the method lacks physical meaning as the principle contributors are coherent. Moreover a conventional technique involves assumptions about independent inputs and the kind of dependence between interfering signals. It is suggested to extract noninterfering but partly coherent virtual signals from measured inputs that are considered as physical inputs. The procedure is based on the determination of a virtual input spectra matrix. As for many practical tasks measured inputs have approximately the same magnitudes of auto, cross spectra and high mutual coherence detection of principal contributors encounters ambiguity. A minimal principle value deviation is employed as criteria to choosing an appropriate virtual input spectra matrix. An example involving acoustical development of a multi indoor units air conditioner that is equipped by compressor generating pressure pulsation into discharge and suction pipeline is considered. These effects cause excessive sound emission of indoor units. The proposed technique is used for the separation of interfering coherent inputs (discharge and suction pressure). It enabled us to develop effective ways for the air conditioner noise reduction.

3:15

4pNSb2. Using the ideal function concept for machine noise control. Brian Landsberger (Caterpillar, Inc., Tech. Ctr., Bldg. E, P.O. Box 1875, Peoria, IL 61656)

An engineered system reaches its ideal function when all its input energy is transformed efficiently into creating the output energy. This leaves less energy available for creating unwanted noise and vibration. A large hydraulic pump evaluation is used to demonstrate the relation between energy efficiency and unwanted noise. Measurements were taken while conducting speed sweeps at various load conditions. Sound in the hemisphere surrounding the pump, acceleration at various locations on and around the pump, and shaft rotation instantaneous phase measurements were used to determine the magnitude, character, and source of the sound. A particularly loud noise level associated with a certain operating region was determined to result from driven vibration of the pump case. The disturbance appeared to originate from pulsations in the pump flow and pressure. This operating region also had a several percent drop in pump efficiency. In this example, more efficient energy transformation would have the dual benefit of higher efficiency and lower noise.

3:30

4pNSb3. Design features for free-field qualification of a new semi-anechoic room, and qualification performance. Kenneth A. Cunefare, Van Biesel, Mark Holdhusen, and Austin Shoemaker (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

Precision qualification of a semi-anechoic room requires careful attention to the sound source and traversing method. Prior work, with test sources mounted above the reflecting floor of such a room, has indicated the potential for image source problems in the resulting field. To address such shortcomings, the new Georgia Tech semi-anechoic room was constructed with a recessed enclosure in the center of the floor. This enclosure permits the implementation of test sources coincident with the reflecting plane of the floor. In addition, prior work in an anechoic room has indicated the inadequacy of qualification traverses implemented at large spacings. To address this issue, hard-points were designed and implemented within the room to permit installation of traverse cables extending radially from the in-floor source enclosure out to the walls and corners. These traverse cables are an integral component of a custom continuous traverse system. The design features of the chamber which facilitate chamber qualification will be presented, along with the broadband and pure tone results of the qualification performed on the chamber.

THURSDAY AFTERNOON, 13 NOVEMBER 2003 TRINITY B ROOM, 1:30 TO 5:00 P.M.

Session 4pPA

Physical Acoustics: Outdoor Propagation

D. Keith Wilson, Chair

Engineering Research and Development Center, US Army Cold Regions Research Laboratory, 72 Lyme Road, Hanover, New Hampshire 03755-1290

Contributed Papers

1:30

4pPA1. Exact formula for the sound scattering cross section per unit volume in a turbulent atmosphere. Vladimir E. Ostashev (NOAA/ Environ. Technol. Lab., 325 Broadway, Boulder, CO 80305 and Dept. of Phys., New Mexico State Univ., Las Cruces, NM 88003) and George H. Goedecke (New Mexico State Univ., Las Cruces, NM 88003)

The sound scattering cross section per unit volume is one of the most important statistical characteristics of a sound wave propagating in a turbulent atmosphere. In the literature, a formula for the sound scattering cross section is derived from an equation for a sound wave propagating in an atmosphere with temperature and velocity fluctuations. Such an equation is obtained from a complete set of linearized equations of fluid dynamics using some approximations. Thus, the classical formula for the sound scattering cross section is intrinsically approximate. In the present paper, sound scattering in a turbulent atmosphere is studied starting from the complete set of linearized equations of fluid dynamics. This approach results in an exact formula for the sound scattering cross section. The exact formula accounts for sound scattering by pressure fluctuations and by the divergence of velocity fluctuations while the classical formula does not. It follows from the exact formula that, in a dry air, a sound wave is scattered at 90 degrees only by pressure fluctuations. This result can be used as a basis for a new remote sensing technique for measuring pressure fluctuations in the atmosphere. [Work supported by the U.S. Army Research Office Grant No. DAAG19-01-1-0640.]

1:45

4pPA2. 3D finite-difference simulation of acoustic waves in turbulent moving media. Neill Symons, David Aldridge (Sandia Natl. Labs., Albuquerque, NM 87185-0750), D. Keith Wilson (U.S. Army Cold Regions Res. and Eng. Lab., Hanover, NH), David Marlin (U.S. Army Res. Lab., White Sands Missile Range, NM), and Vladimir Ostashev (NOAA Environ. Technol. Lab., Boulder, CO)

A finite-difference algorithm appropriate for modeling acoustic waves in a fully heterogeneous moving 3D media has been developed. The model is characterized by: acoustic velocity, density, and the three components of the background media velocity. The approach solves a set of coupled 1st order velocity-pressure differential equations appropriate for an adiabatic divergence-free background velocity. The equations are staggered in time and space and the algorithm uses second order temporal and fourth order spatial finite-differences. Since approximations are not adopted in the solution of the equations all arrivals are modeled with fidelity providing the spatial and temporal grids are chosen appropriately. The algorithm can include either a pressure or velocity free surface on the bottom boundary and absorbing boundaries on other model flanks. Designed to run on large scale parallel computational platforms, the algorithm has been validated for four machine architectures. Comparisons are presented to an analytic solution for a constant wind model and fast-field program results for a vertically stratified wind model. Data resulting from simulations through a kinematic turbulence wind profile developed with the quasi-wavelet method are also presented. Sandia National Laboratories is operated by Sandia Corporation, a Lockheed Martin Company, for the USDOE under Contract No. 94-AL85000.

2:00

4pPA3. Eddy-size decomposition of the scattering cross section using quasi-wavelets. D. Keith Wilson (U.S. Army Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755), Vladimir E. Ostashev (NOAA/Environ. Technol. Lab., 325 Broadway, Boulder, CO 80305), George H. Goedecke (New Mexico State Univ., Las Cruces, NM 88003), and Harry J. Auvermann (Dallas, TX 75228)

Unlike traditional Fourier spectral representations, quasi-wavelets (QWs) describe the structure of turbulence with spatially localized functions. As a result, QWs are particularly well suited to examining the dependence of a turbulent scattering process on the size and spatial location of the eddies. In this paper, the scattering cross section for QWs of an individual size class is derived. It is shown that previous results for the von Kármán spectrum can be reproduced when the scattered energies from a continuous distribution of QW sizes are combined. A Bragg resonance condition is derived for the eddy size that scatters most strongly for a given acoustic wavenumber and scattering angle. An example application of QWs to scattering over noise barriers shows that, for a typical barrier geometry, most of the scattered energy originates from eddies in the size range of approximately one-half to twice the size of the eddies responsible for maximum scattering.

2:15

4pPA4. Time-domain boundary conditions for outdoor ground surfaces. Sandra L. Collier (U.S. Army Res. Lab., ATTN: AMSRL-CI-EE, 2800 Powder Mill Rd., Adelphi, MD 20783-1197, scollier@arl.army.mil), Vladimir E. Ostashev (NOAA/Environ. Technol. Lab., Boulder, CO 80305-3328), D. Keith Wilson (Eng. Res. and Development Ctr., U.S. Army Cold Regions Res. and Eng. Lab., Hanover, NH 03755-1290), and David H. Marlin (U.S. Army Res. Lab, White Sands Missile Range, NM 88002-5501)

Finite-difference time-domain techniques are promising for detailed dynamic simulations of sound propagation in complex atmospheric environments. Success of such simulations requires the development of new techniques to accurately handle the reflective and absorptive properties of a porous ground. One method of treating the ground boundary condition in the time domain [Salomons *et al.*, Acta Acust. **88**, 483–492 (2002)] is to use modified fluid dynamic equations, where the ground is considered as a porous medium described by its physical properties. However, this approach significantly increases computation time, as the domain must be extended into the ground and a large number of grid points are needed. Standard impedance models for the ground boundary condition are frequency-domain models, which generally are non-causal [Y. H. Berthelot, J. Acoust. Soc. Am. **109**, 1736–1739 (2001)]. The development of a time-domain boundary condition from these models requires removing the singularity from the impedance equation when transforming from the frequency domain to the time domain. Alternatively, as the impedance boundary condition is a flux equation, a time-domain boundary condition can be derived from first principles, using the physical properties of the ground. We report on our development of a time-domain ground boundary condition.

2:30

4pPA5. Discussion of errors in the Green's function parabolic equation for outdoor sound propagation. Jennifer Cooper and David C. Swanson (Grad. Prog. in Acoust., Penn State Univ., 202 ASB, University Park, PA 16803, jlc375@psu.edu)

As is to be expected of any numerical method, use of the Green's function parabolic equation (GFPE) does result in small errors that can propagate and increase. The main causes of errors in using the GFPE for outdoor sound propagation with constant sound speeds appear to be inadequate spatial sampling, approximated starting fields, and errors in the numerical integration used to compute the surface wave. The relative significance of these errors as well as a set of appropriate computational parameters for avoiding or minimizing them will be discussed. With the proper parameter selection (for step sizes, attenuation layer depth, etc.), the GFPE produces reliably accurate results for outdoor sound propagation even over finite impedance grounds and beyond barriers or changes in ground height.

2:45

4pPA6. Modeling of secondary sonic booms: Influence of the variability of the atmosphere. Laurent Dallois, Philippe Blanc-Benon, and Julian Scott (Ctr. Acoustique—Ecole Centrale de Lyon, 69134 Ecully, France)

The shock waves generated by a supersonic aircraft are reflected in the upper part of the atmosphere. Back to the ground, they are indirect sonic booms called secondary sonic booms. The recorded signals of secondary sonic booms show a low amplitude and a low frequency. They sound like rumbling noises due to amplitude bursts. These signals strongly depend on the atmospheric conditions, in particular to the amplitude and to the direction of the wind in the stratopause. The propagation of the secondary sonic boom is studied using atmospheric models up to the thermosphere. By solving temporal ray equations, the secondary carpet position is investigated. An amplitude equation including nonlinearity, absorption, and relaxation by various chemical species is coupled to the ray solver in order to get information on the amplitude and on the frequency of the sonic boom at the ground level. Using this propagation model and the atmospheric model, the seasonal dependencies of the secondary sonic boom are investigated. Multipath arrivals are directly linked to wind field or 3-D inhomogeneities. They have been of special concern as a way to explain the rumble noise as a summation over different ray contributions.

3:00–3:15 Break

3:15

4pPA7. Attenuation of blast sound by a mixed stand of pine and hardwood. Michael J. White, Larry L. Pater, Ryan J. Lee, and George W. Swenson, Jr. (US Army ERDC/CERL, P.O. Box 9005, Champaign, IL 61826)

We performed an experiment to determine the attenuation of impulsive sound by a forest in northeastern Texas in July 2002. In the measurement, microphones were placed along a line that extended at one end into approximately 300 m of mostly 20-cm-diam pine trees, and at the other end into an open field of roughly the same extent. Explosive charges of 0.57- and 2.27-kg Composition C-4 were detonated at four locations along the line, at either edge of the open field and at either edge of the trees, in order to compare sound attenuation rates within the woods to those in the open field. Additional microphones were placed 2 and 4 km away to compare propagation from sources in either wooded or open positions to microphones in either wooded or open positions, with all paths substantially forested. Charge size, height-of-burst, and microphone height were varied in order to excite and probe a variety of acoustic propagation modes within the tree layer. We discuss preliminary results of 1/3-octave band analysis of this data set, consider ways to separately identify the observed power law and exponential decay rates, and speculate on their controlling mechanisms.

3:30

4pPA8. Forest physical sampling strategies for investigating noise mitigation benefits. Patrick J. Guertin and Michael J. White (US Army ERDC/CERL, P.O. Box 9005, Champaign, IL 61826)

Acoustic attenuation rates in tree stands have been seen to vary considerably between measurements in various settings, with little hint as to the controlling mechanisms. Usual descriptors for a forest include tree species, stand age, and basal diameter per land surface area, properties that may not correlate with acoustic attenuation rates. As part of a recent forest attenuation measurement, we developed a forest sampling regimen aimed at indicating and distinguishing between three candidate effects: (1) scattering by resilient, wood material, (2) scattering by more compliant, leaf material, (3) refraction by canopy microclimate influence. Estimates of the wood material were made by an established point sampling process, in which significant tree trunks subtend a minimum angle at breast height. Canopy microclimate effect was characterized using percent optical closure of the upper canopy, canopy thickness, and ground vegetation height. Digital photographs aimed outward in four cardinal directions from each sample point were processed to segregate percentages of wood material, leaf material, and sky. We are currently exploring the connection between these parameters and measured attenuation rates in a single stand of woods. We propose that these parameters could offer improved correlation with measured attenuation rates between forests.

3:45

4pPA9. Sound propagation in an urban environment. I: Preliminary analysis of measurements. Donald G. Albert (US Army ERDC Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755, dalbert@crrel.usace.army.mil) and Lanbo Liu (US Army ERDC Cold Regions Res. and Eng. Lab., Hanover, NH 03755)

Experimental measurements were conducted in a full-scale artificial village to determine the effect of buildings on sound propagation outdoors. The village consisted of 15 concrete block buildings arranged in a 150 $\times 150$ m area. Explosive charges were detonated to produce acoustic pulses that were digitally recorded by sensors scattered throughout the village. The measurements confirm that diffraction acts as a low pass filter on acoustic waveforms, greatly reducing the peak pressure received by a sensor in the shadow zone. The measured data present clear examples of the effect of reflections on propagation down a street canyon and diffraction around building corners. Complex signatures formed by multiple reflections and diffractions were recorded in situations where the acoustic waves passed multiple buildings in propagating from the source to the receiver. [Work funded by the U.S. Army.]

4:00

4pPA10. Sound propagation in an urban environment. II: Preliminary FDTD model. Lanbo Liu (US Army ERDC Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755, Lanbo.liu@erdc.usace.army.mil) and Donald G. Albert (US Army ERDC Cold Regions Res. and Eng. Lab., Hanover, NH 03755)

A 2-dimensional finite difference time domain (FDTD) method was used to simulate a data set recorded by a microphone array in an urban terrain. The FDTD model uses a coupled system of first-order differential equations for excess pressure and particle velocity. The perfectly matched layer method was used to truncate the model domain. This technique is much more efficient in diminishing unwanted reflections from the domain boundaries than any other absorptive boundary technique. The numerical integration was carried out with a staggered grid system. This algorithm has been validated with comparison with analytical solutions for cases with uniform physical parameters and simple geometries. For the predicted sound levels, although they are in general agreement with the measured values at 13 sensor locations, there is a significant discrepancy occurred at one particular sensor location (microphone location 9) behind one building (Building A). The observed sound level is about an order of magnitude greater than the predicted level, suggesting that much sound energy has propagated through this building. In contrast, a general consistency, or comparable level, can be reached between the measured and the modeled sound levels within 2-fold for the remaining 13 stations. [Work funded by the US Army.]

4:15

4pPA11. Wave propagation in viscoelastics with chemical relaxation. Timothy Margulies (Dept. of Mathematics, Community College of Philadelphia, 1700 Spring Garden St., Philadelphia, PA 19130)

Acoustic wave properties of fluids have provided a fruitful qualitative and quantitative analysis of fast chemical reactions. The purpose of this paper is to present an extension of the linear acoustic theory of Verdier and Piau [J. Acoust. Soc. Am. **101**, 4 (1997)] to include simultaneous chemical kinetic relaxation. The energy equation, equation of state with thermodynamic relations are formulated to account for chemical reactions. A general wave equation for a compressible viscoelastic with thermal transfer and chemical progress variable perturbations are explicitly modeled. Longitudinal wave propagation is described by a biquadratic equation for the sound absorption and dispersion. Approximations of small thermal transfer and viscous effects yield linearly additive contributions of viscous, thermal, and reaction contributions. Here, linearly viscous or visco-elastic effects can be examined, in addition to the chemical kinetic effects. Sample calculations illustrating the physio-chemical dynamics will be presented.

4:30

4pPA12. Wave propagation in viscoelastic horns using a fractional calculus rheology model. Timothy Margulies (Dept. of Mathematics, Community College of Philadelphia, 1700 Spring Garden St., Philadelphia, PA 19130)

The complex mechanical behavior of materials are characterized by fluid and solid models with fractional calculus differentials to relate stress and strain fields. Fractional derivatives have been shown to describe the viscoelastic stress from polymer chain theory for molecular solutions [Rouse and Sittel, J. Appl. Phys. **24**, 690 (1953)]. Here the propagation of infinitesimal waves in one dimensional horns with a small cross-sectional area change along the longitudinal axis are examined. In particular, the linear, conical, exponential, and catenoidal shapes are studied. The wave amplitudes versus frequency are solved analytically and predicted with mathematical computation. Fractional rheology data from Bagley [J. Rheol. **27**, 201 (1983); Bagley and Torvik, J. Rheol. **30**, 133 (1986)] are incorporated in the simulations. Classical elastic and fluid "Webster equations" are recovered in the appropriate limits. Horns with real materials that employ fractional calculus representations can be modeled to examine design trade-offs for engineering or for scientific application.

4:45

4pPA13. TWA Flight 800, explosion airblast unexplained. Jack W. Reed (JWR, Inc., 1128 Monroe SE, Albuquerque, NM 87108, jwreed@nmia.com)

TWA Flight 800 disintegrated off Long Island, NY, on 16 July 1996. Immediate reports from other flyers described what appeared as attacking missiles. Search for terrorists began quickly, with over 1000 FBI agents to collect debris, interview eyewitnesses, and analyze sightings to give a missile launch point. They found no evidence of criminal attack, and turned investigations over to the NTSB to find some accidental cause. The "empty" central fuel tank was determined to be the likely explosion source. On the other hand, early witnesses reported a "loud" bang after seeing a great fireball fall from the sky, but at 15 km or greater range, they saw and heard two different events. This acoustic discrepancy has not been adequately investigated. When finally released, FBI reports from more than 200 "ear-witnesses" give similar observations. Their loudness reports confirm that at least a ton of TNT equivalent explosion had occurred. NASA acousticians engaged by NTSB, however, through spectral analysis techniques for sonic booms, concluded that a 10-kg TNT explosion, from detonating fuel tank vapors, could be heard on Long Island. Evidence for a much larger yield is presented here, but its form remains a mystery.

Session 4pSA

Structural Acoustics and Vibration: Sound Field Reconstructions

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

Contributed Papers

2:00

4pSA1. Application of multisource discrimination to the HELS method. Nassif Rayess (Dept. of Mech. Eng., Univ. of Detroit–Mercy, Detroit, MI 48219), Manmohan Moondra, and Sean Wu (Wayne State Univ., Detroit, MI 48202)

In most engineering applications, the acoustic fields are usually generated by multiple incoherent sources. When such acoustic fields are sampled and taken as input data to the Helmholtz equation least-squares (HELS) formulations directly, the reconstruction might not be accurate. Hence there is a need to discriminate the contributions from individual sources and separate the composite sound field into a set of spatially coherent subfields that are also mutually incoherent. In this paper, we apply the principal component decomposition technique to the HELS method to reconstruct acoustic radiation from multiple incoherent sources in a free field. The key ingredients of this technique include a diagonalization by singular value decomposition (SVD) of the cross-spectral matrices generated by a number of reference microphones. Each diagonal term corresponds to a subfield that results from a so-called virtual sound source. Even though such virtual sources are not always representative of the actual sources, they are nonetheless incoherent and thus can be used as input to HELS. Experimental validations of this technique on reconstructing acoustic pressure and time-averaged normal acoustic intensity on the surfaces of multiple incoherent sources are presented. [Work supported by NSF.]

2:15

4pSA2. Analysis and comparison of three energy density probe designs. Lance L. Locey and Scott D. Sommerfeldt (N283 Eyring Sci. Ctr. (ESC), Provo, UT 84602, Lance@Locey.com)

Previous research has demonstrated the utility of acoustic energy density measurements as a means to gain a greater understanding of acoustic fields. Three energy density probes are under development. The first probe has three orthogonal pairs of microphones embedded on the surface of a sphere. The second design is a similarly sized sphere with four surface mounted microphones, equidistant from the center of the sphere. The four microphones are arranged to correspond with the origin and unit vectors of a Cartesian coordinate system, where the origin and the tips of the three unit vectors are on the surface of the sphere. As a result, all four microphones lie on the surface of the top hemisphere. The third design consists of a similarly sized sphere with four surface microphones arranged in a tetrahedron configuration. The author will discuss some of the errors and limitations associated with the four microphone designs as compared to the six microphone design. [Work supported by NASA.]

2:30

4pSA3. Helmholtz equation least squares method for transient nearfield acoustic holography. Huancai Lu and Sean Wu (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI 48202)

The generalized Helmholtz equation least squares (HELS) formulations for reconstructing transient acoustic radiation from an arbitrary object subject to an arbitrary time-dependent excitation are derived. To facilitate the derivations, the Laplace transform is employed and the vibro-acoustic quantities on the source surface are solved explicitly in terms of the acoustic pressures measured on a conformal surface around the source at close range multiplied by transfer functions in the Laplace domain first. The vibro-acoustic responses in the time domain can then be expressed as convolution integrals of the measured acoustic pressure signals over temporal kernels. Replacing the spherical Hankel functions in the transfer functions with polynomial expressions, we can recast the infinite integrals in the inverse Laplace transform as contour integrals and evaluate the temporal kernels by using residue theorem. Once the temporal kernels are determined, the vibro-acoustic quantities anywhere in the field, including those on the source surface can be reconstructed directly. Numerical examples of reconstruction of transient acoustic radiation from a baffled disk subject to impulsive and arbitrarily time-dependent excitations are demonstrated. [Work supported by NSF.]

2:45

4pSA4. Reconstruction of acoustic radiation from vibrating objects in a half space using hybrid nearfield acoustical holography. Xiang Zhao and Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, xiangzhao@wayne.edu)

Hybrid nearfield acoustical holography (NAH) [Wu, J. Acoust. Soc. Am. **113**, 2252 (2003)] is utilized to reconstruct the vibro-acoustic fields generated by vibrating structures in half space. These hybrid formulations are derived from a modified Helmholtz equation least-squares method, where the field acoustic pressure is expanded in terms of both outgoing and incoming spherical waves, and the Helmholtz integral formulations implemented by the boundary element method (BEM). To overcome the ill-conditioning difficulties in the resultant matrix equations, regularization techniques such as the method of Tikhonov and generalized cross validation are employed. The main advantage of this hybrid approach is that satisfactory reconstruction can be obtained with relatively few acoustic pressure measurements taken over a conformal surface around the source. This is because a large portion of the input data is regenerated by the modified HELS, but not actually measured. Hence, the efficiency of reconstruction is enhanced. If the same number of measurements is used in an inverse BEM code, the reconstructed vibro-acoustic quantities on the source surface may be distorted. Numerical examples of a dilating and an oscillating sphere and a cylinder with spherical endcaps in half space are demonstrated. [Work supported by NSF.]

3:00

4pSA5. On the reconstruction of sound field using the equivalent source modeling. Jeong-Guon Ih and In-Youl Jeon (Dept. of Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Science Town, Taejon 305-701, Korea)

The sound field radiated from a vibrating structure is reconstructed by using the equivalent source modeling. In this method, the equivalent multipole sources which are similar to the spherical harmonics in the Helmholtz equation least-squares (HELS) method are distributed over or inside a vibrating surface. To improve the accuracy and usefulness, the effective independence (EfI) method is used to determine the optimal position of equivalent sources and the regularization scheme is adopted to determine the optimal expansion number of spherical harmonics and to suppress the high order poles. After regenerating field pressures using those optimized equivalent source modeling, the vibrating source is effectively recon-

structed with minimal measurement combining the proposed method with the near-field acoustic holography (NAH) using the inverse boundary element method (BEM). In this study, an application example of a commercial vacuum cleaner using the equivalent source modeling will be demonstrated. [Work supported by the BK21 project and NRL.]

3:15

4pSA6. Reconstruction of the surface normal acoustic intensity from noisy pressure data by regularized Helmholtz equation least squares method. Tatiana Semenova and Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI 48202)

Reconstructing the normal acoustic intensity on a source surface from the field pressure measurements is critical in identifying the noise sources and their transmission paths. In this paper, we consider the reconstruction of time-averaged normal acoustic intensities from noisy pressure data using the Helmholtz equation least squares (HELS) method with regularization by truncated singular value decomposition or the method of Tikhonov. The regularization parameter is calculated by generalized cross validation, L-curve, and quasioptimality criterion and the best solution is chosen. Numerical results show that the normal velocity reconstruction is more sensitive to noise in the input data than the pressure reconstruction, and depends critically on the choice of the regularization parameter. It is found that the HELS-based Dirichlet-to-Neumann (DtN) map has a similar accuracy as compared to the Green's formula-based DtN map with Tikhonov regularization for 1% noise or larger in the input data, provided that HELS converges sufficiently fast. Moreover, determining the surface normal velocity from the reconstructed surface pressure is more stable than reconstructing it directly from the field pressure data. It is concluded that a regularized HELS, when it converges fast enough, can be an easy-to-use and cost-effective tool for intensity reconstruction [Work supported by NSF.]

THURSDAY AFTERNOON, 13 NOVEMBER 2003 SAN ANTONIO ROOM, 1:00 TO 5:10 P.M.

Session 4pSC

Speech Communication: Statistical Patterns in Speech

Andrew J. Lotto, Cochair
Department of Psychology, University of Texas–Austin, 1 University Station A8000, Austin, Texas 78712-0187

Lori L. Holt, Cochair
Department of Psychology, Carnegie Mellon University, 5000 Forbes Avenue, Pittsburgh, Pennsylvania 15213

Randy L. Diehl, Cochair
Department of Psychology, University of Texas−Austin, 1 University Station A8000, Austin, Texas 78712-0187

Chair's Introduction—1:00

Invited Papers

1:05

4pSC1. What are the statistics in statistical learning? Lori L. Holt (Dept. of Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, lholt@andrew.cmu.edu) and Andrew J. Lotto (Boys Town Natl. Res. Hospital, Omaha, NE 68131)

The idea that speech perception is shaped by the statistical structure of the input is gaining wide enthusiasm and growing empirical support. Nonetheless, statistics and statistical learning are broad terms with many possible interpretations and, perhaps, many potential underlying mechanisms. In order to define the role of statistics in speech perception mechanistically, we will need to more precisely define the statistics of statistical learning and examine similarities and differences across subgroups. In this talk, we examine learning of four types of information: (1) acoustic variance that is defining for contrastive categories, (2) the correlation between acoustic attributes or linguistic features, (3) the probability or frequency of events or a series of events, (4) the shape of input distributions. We present representative data from online speech perception and speech development and discuss inter-relationships among the subgroups. [Work supported by NSF, NIH and the James S. McDonnell Foundation.]

1:30

4pSC2. Bayesian approaches to perception. Randy L. Diehl and Sarah C. Sullivan (Dept. of Psych. and Ctr. for Perceptual Systems, Univ. of Texas, 1 University Station A8000, Austin, TX 78712-0187)

The concepts of Bayesian statistical decision theory have recently transformed research in perception by providing a rigorous mathematical framework for representing the statistical properties of the environment, for describing the tasks that perceivers perform, and for deriving computational theories of optimal performance on those tasks. Such computational theories (ideal observers) offer an appropriate benchmark for evaluating human performance, and, moreover, they can be modified to yield an excellent starting point for developing testable models. Unlike theories that focus on the role of invariant information for perceived events, the Bayesian approach treats information as probabilistic. The approach is illustrated for cases in which listeners learn to categorize artificial stimulus sets whose statistical properties (prior probabilities and stimulus likelihoods) are controlled by the experimenter. Prospects for extending the Bayesian framework to the analysis of speech perception are discussed in light of the recent progress in developing ideal observers for natural environmental stimuli. A necessary but daunting task will be the measurement of probability distributions that characterize natural speech categories. [Work supported by NIDCD.]

1:55

4pSC3. Explicit pattern recognition models for speech perception. Terrance M. Nearey (Linguist., Univ. of Alberta, Edmonton, AB T6G 0A2, Canada)

Optimal statistical classification of arbitrary input signals can be obtained, in principle, via a Bayesian classifier, given (perfect) knowledge of the distributions of signal properties for the set of target categories. At least for certain constrained problems, such as the perception of isolated vowels, simple (imperfect) statistical pattern recognition techniques can accurately predict human listeners' performance. This paper sketches several relatively successful case studies of the application of static pattern recognition techniques to speech perception. (Static techniques require inputs of a fixed length, e.g., $F1$ and $F2$ for isolated vowels.) Real speech clearly requires dynamic pattern recognition, allowing inputs of arbitrary length. Certain such methods, such as dynamic programming and hidden Markov models, have been widely exploited in automatic speech recognition. The present paper will describe initial attempts to apply variants of such methods to the data from a perception experiment [T. Nearey and R. Smits, J. Acoust. Soc. Am. **111** (2002)] involving the perception of three (VCV) or four (VCCV) segment strings. Practical and conceptual problems in the application of such techniques to human perception will be discussed. [Work supported by SSHRC.]

2:20

4pSC4. Early language acquisition: Statistical learning and social learning. Patricia K. Kuhl (Ctr. for Mind, Brain & Learning, Univ. of Washington, Seattle, WA 98195)

Infants are sensitive to the statistical patterns in language input, and exposure to them alters phonetic perception. Our recent data indicate that first-time exposure to a foreign language at 9 months of age results in learning after only 5 h, suggesting a process that is fairly automatic, given natural language input. At the same time, it appears that early phonetic learning from natural language may be constrained by the need for social interaction. Our work demonstrates that infants learn phonetically when exposed to a live, but not a pre-recorded, speaker. This talk will focus on statistical learning in a social context and develop the thesis that this combination provides an ideal situation for the acquisition of a natural language.

2:45–3:00 Break

3:00

4pSC5. Distributional and statistical bases of allophonic groupings. James L. Morgan, Katherine S. White, Cecilia Kirk (Cognit. & Linguist. Sci., Brown Univ., Providence, RI 02912, James_Morgan@Brown.Edu), Sharon Peperkamp, and Emmanuel Dupoux (Ecole Normale Superieure, 75005 Paris, France)

Segments of a given language do not all have equal status: some are phonetic variants of others. These variants occur only in certain contexts. In addition to occurring in more restricted contexts, however, variant allophones typically occur with lesser frequency. Which (if either) of these asymmetries is important for infants learning of allophonic groupings? Infants were exposed to artificial languages with allophones. Either stops or fricatives were intervocalicly voiced; these voiced segments did not otherwise occur. Infants were then tested on preference for passages like "rot pazo na bazo rot pazo na bazo" or "na zine rot sine na zine rot sine . . . " All items were grammatical for both groups. For the stop allophone group, /pazo/ and /bazo/ should be one word, whereas /zine/ and /sine/ should be two words; *vice versa* for the fricative allophone group. In previous work using a similar technique, infants listened longer to passages containing what was perceived as two variants. Across a series of experiments, the number and frequency of contexts in which allophonic variants occurred were manipulated. Cross-experiment analyses reveal the relative contributions of distributional and statistical asymmetries. We interpret and discuss our findings in light of current statistical learning approaches to language acquisition.

3:25

4pSC6. Using pronunciation data to study word recognition. Mark A. Pitt (Dept. of Psych., Ohio State Univ., Columbus, OH) and Keith Johnson (Ohio State Univ., Columbus, OH)

Many of the mysteries of spoken word recognition have evolved out of the observation that pronunciation is highly variable yet perception is amazingly stable. Extreme forms of variation can even result in different phonetic percepts, such as consonant assimilation (e.g., "green ball:" "greem ball") and deletion (e.g., "and:" "an"), yet listeners still perceive the intended word. The results from two lines of work will be presented in which we are studying the regularity of these production phenomena to evaluate models

of how phonological variants are recognized. In one project we examined variation itself. Phonological and acoustic analyses of phonological variation in the Buckeye corpus of conversational speech were carried out in which we asked questions such as the following: How predictable and consistent is regressive assimilation? and How acoustically similar is the assimilated segment to an intended production of that same segment? In a related project, we examined the listener's sensitivity to the variation found in conversational speech. Results reveal the complexities of these production phenomena, the challenges models must overcome to account for how variants are recognized, and listeners' sensitivity to stochastic properties of pronunication variation. [Work supported by NIDCD.]

3:50

4pSC7. Phonotactic probability and neighborhood density effects in the perception and production of speech in adults. Michael S. Vitevitch, Jonna Armbruster, and Julia Fitzer (Dept. of Psych., 1415 Jayhawk Blvd., Univ. of Kansas, Lawrence, KS 66045, mvitevit@ku.edu)

Phonotactic probability refers to the frequency with which phonological segments and sequences of segments occur in a language. Neighborhood density refers to the number of words that sound similar to a given word. Awareness to phonotactic probability and neighborhood density occurs early in life and seems to influence lexical development. Recent research in our lab suggests that these two variables also influence the on-line processing of spoken words in adults. In the case of spoken word recognition, when sub-lexical representations dominate processing, high probability stimuli are responded to more quickly than low probability stimuli. However, when lexical representations dominate processing, stimuli with sparse neighborhoods are responded to more quickly than stimuli with dense neighborhoods. Similarly, in speech production, pictures of high probability words are named more quickly than pictures of low probability words. However, in contrast to speech perception, pictures of words with dense neighborhoods are produced more quickly than pictures of words with sparse neighborhoods, suggesting that statistical regularities in the language may be used differently depending on the demands imposed by a given process or task.

4:15

4pSC8. Statistical learning and the challenge of syntax: Beyond finite state automata. Jeff Elman (Dept. of Cognit. Sci., Univ. of California–San Diego, La Jolla, CA 92093)

Over the past decade, it has been clear that even very young infants are sensitive to the statistical structure of language input presented to them, and use the distributional regularities to induce simple grammars. But can such statistically-driven learning also explain the acquisition of more complex grammar, particularly when the grammar includes recursion? Recent claims (e.g., Hauser, Chomsky, and Fitch, 2002) have suggested that the answer is no, and that at least recursion must be an innate capacity of the human language acquisition device. In this talk evidence will be presented that indicates that, in fact, statistically-driven learning (embodied in recurrent neural networks) can indeed enable the learning of complex grammatical patterns, including those that involve recursion. When the results are generalized to idealized machines, it is found that the networks are at least equivalent to Push Down Automata. Perhaps more interestingly, with limited and finite resources (such as are presumed to exist in the human brain) these systems demonstrate patterns of performance that resemble those in humans.

Contributed Papers

4:40

4pSC9. The effects of distributional information on the formation of nonspeech sound categories by infants. Jessica F. Hay (Dept. of Psych., Univ. of Texas, Austin, TX 78712, hay@psy.utexas.edu)

In order to learn native sound categories, infants may be attending to the distributional information available in their native language environment. The present study examined eight-month-old infants ability to form categories with small amounts of exposure to distributional information. Infants were exposed to a bimodal distribution of nonspeech sounds (narrow-band noise bursts that vary in center frequency). Their ability to discriminate within (same mode) and between category (different mode) differences was then assessed through a habituation-dishabituation procedure. Data were interpreted in terms of dishabituation to change stimuli. The change stimuli were equal in physical step size and either came from within the same mode of the bimodal distribution or from the other mode. The findings may have implications regarding auditory category formation in general and could lend support to the hypothesis that infants use distributional information available in their ambient language environment to learn their native speech categories. [Work supported by NIDCD.]

4:55

4pSC10. Statistical modeling of infant-directed versus adult-directed speech: Insights from speech recognition. Katrin Kirchhoff and Steven Schimmel (Dept. of Elec. Eng., Univ. of Washington, Seattle, WA 98115)

Studies on infant speech perception have shown that infant-directed speech (*motherese*) exhibits exaggerated acoustic properties, which are assumed to guide infants in the acquisition of phonemic categories. Training an automatic speech recognizer on such data might similarly lead to improved performance since classes can be expected to be more clearly separated in the training material. This claim was tested by training automatic speech recognizers on adult-directed (AD) versus infant-directed (ID) speech and testing them under identical versus mismatched conditions. 32 mother–infant conversations and 32 mother–adult conversations were used as training and test data. Both sets of conversations included a set of cue words containing unreduced vowels (e.g., sheep, boot, top, etc.), which mothers were encouraged to use repeatedly. Experiments on continuous speech recognition of the entire data set showed that recognizers

trained on infant-directed speech did perform significantly better than those trained on adult-directed speech. However, isolated word recognition experiments focusing on the above-mentioned cue words showed that the drop in performance of the ID-trained speech recognizer on AD test speech was significantly smaller than vice versa, suggesting that speech with over-emphasized phonetic contrasts may indeed constitute better training material for speech recognition. [Work supported by CMBL, University of Washington.]

THURSDAY AFTERNOON, 13 NOVEMBER 2003 SAN MARCOS ROOM, 2:15 TO 5:00 P.M.

Session 4pSP

Signal Processing in Acoustics, Underwater Acoustics, Speech Communication, Animal Bioacoustics, Noise and Engineering Acoustics: Detection and Classification in Acoustics IV

Patrick J. Loughlin, Chair

Department of Electrical Engineering, University of Pittsburgh, 348 Benedum Hall, Pittsburgh, Pennsylvania 15261

Invited Papers

2:15

4pSP1. Detection and classification of underwater targets by echolocating dolphins. Whitlow Au (Marine Mammal Res. Prog., Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734)

Many experiments have been performed with echolocating dolphins to determine their target detection and discrimination capabilities. Target detection experiments have been performed in a naturally noisy environment, with masking noise and with both phantom echoes and masking noise, and in reverberation. The echo energy to rms noise spectral density for the Atlantic bottlenose dolphin (*Tursiops truncatus*) at the 75% correct response threshold is approximately 7.5 dB whereas for the beluga whale (*Delphinapterus leucas*) the threshold is approximately 1 dB. The dolphin's detection threshold in reverberation is approximately 2.5 dB vs 2 dB for the beluga. The difference in performance between species can probably be ascribed to differences in how both species perceived the task. The bottlenose dolphin may be performing a combination detection/discrimination task whereas the beluga may be performing a simple detection task. Echolocating dolphins also have the capability to make fine discriminate of target properties such as wall thickness difference of water-filled cylinders and material differences in metallic plates. The high resolution property of the animal's echolocation signals and the high dynamic range of its auditory system are important factors in their outstanding discrimination capabilities.

3:00

4pSP2. Speech enhancement for better hearing aid design using chirped wavelets. Adele Doser (Sandia Natl. Labs., P.O. Box 5800, M.S. 1188, Albuquerque, NM 87185-1188)

Wavelet techniques are utilized in the enhancement of speech signals for hearing-impaired persons. The interest is in schemes that isolate and enhance short duration, broadband speech elements (such as unvoiced, fricative consonants) while still capturing narrowband speech components (such as vowels). Discerning unvoiced consonants presents an obstacle for hearing-impaired persons (i.e., "bark" and "park" may sound identical). A scheme is proposed, using chirped wavelets, to seize the characteristics of broadband speech elements. The method is tested on speech data, in noise-free and noisy conditions, and compared with sinusoidal-based wavelets. Results demonstrate that chirped wavelets are useful in the analysis and enhancement of speech for the hearing impaired. It is hoped these techniques might be incorporated in future hearing aid design.

3:30

4pSP3. Local characteristics of dispersive pulse propagation. Patrick J. Loughlin (Dept. of Elec. Eng., Univ. of Pittsburgh, 348 Benedum Hall, Pittsburgh, PA 15214)

As a wave propagates, its shape and other physical characteristics may change, depending on the initial wave and the propagation environment. In this talk, the effects of structural dispersion on the spatial and temporal spreading of a propagating pulse wave, and on the spectral characteristics of the pulse over time, are considered. These effects are quantified by obtaining the local spatial, temporal, and spectral moments of the pulse. General results for any dispersion relation, as well as the particular case of a two-plate waveguide, are presented. [Work supported by ONR.]

4:00–4:15 Break

4:15

4pSP4. Material discrimination using bispectral signatures. Paul A. Nyffenegger and Melvin J. Hinich (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

A method is presented for material discrimination and characterization using bispectral signatures acquired from an object actively probed with acoustic pulses. Although bispectral techniques have proven useful in a diverse array of fields including passive acoustic ranging, bispectral processing in active acoustic applications has not been widely explored. The mechanisms responsible for the bispectral signatures revealed using active acoustics have not been well studied and little is known about the relative contributions to the bispectrum originating in the physical properties of the target material itself rather than from target structural acoustics and the propagation media. In a pilot experiment, we determine bispectral signatures for three targets of differing composition but similar dimensions using a submerged ultrasonic apparatus. The experiment is designed to isolate effects due to target properties from those attributable to propagation path, source, or receiver. The source wavelet is a broad-spectrum linear frequency modulated pulse. The normalized bispectrum is calculated using conventional nonparametic methods, and is averaged across many frames. Results indicate that at ultrasonic frequencies this technique provides signatures with the potential of discriminating between classes of materials such as plastic, metal, and rock. [Work supported by Applied Research Laboratories.]

4:30

4pSP5. Spatio-temporal gradient signal processing for reconstructing scattered wave fields. Kenbu Teramoto and Kohsuke Tsuruta (Dept. of Mech. Eng., SAGA Univ., Saga-shi 8408502, Japan, tera@me.saga-u.ac.jp)

This paper presents a subspace signal processing method which detects the scattered sound waves based on the spatio-temporal gradient analysis over the Lamb wave field. The proposed method has an ability to classify the surface acoustic wave field through the rank of the covariance matrix defined over the four-dimensional vector space which is spanned by the following components: a vertical displacement, its vertical velocity, and a pair of shear strains of the plate surface. The covariance matrix provides the information about cracks. When a unique sound source exists on the surface without any cracks or scatterers, the rank of the covariance matrix is reduced to two. Increasing the rank up to three, however, one or more scatterers exist on the surface. Therefore, by the singular value decomposition (SVD) of the covariance matrix, optimal, orthonormal bases are obtained. In formulating the scattered wave field by the unknown object, its best reconstruction, in the LMS sense, can be obtained by the sum of orthogonal vector spaces. The computational process in a wave field near the cracks is discussed and their physical meanings are investigated through DTD-simulations and acoustic experiments.

4:45

4pSP6. Constrained maximum consistency multi-path mitigation. George B. Smith (Naval Res. Lab., Code 7185, Stennis Space Center, MS 39529-5004)

Blind deconvolution algorithms can be useful as pre-processors for signal classification algorithms in shallow water. These algorithms remove the distortion of the signal caused by multipath propagation when no knowledge of the environment is available. A framework in which filters that produce signal estimates from each data channel that are as consistent with each other as possible in a least-squares sense has been presented [Smith, J. Acoust. Soc. Am. **107** (2000)]. This framework provides a solution to the blind deconvolution problem. One implementation of this framework yields the cross-relation on which EVAM [Gurelli and Nikias, IEEE Trans. Signal Process. **43** (1995)] and Rietsch [Rietsch, Geophysics **62**(6) (1997)] processing are based. In this presentation, partially blind implementations that have good noise stability properties are compared using Classification Operating Characteristics (CLOC) analysis. [Work supported by ONR under Program Element 62747N and NRL, Stennis Space Center, MS.]

THURSDAY AFTERNOON, 13 NOVEMBER 2003 PECOS ROOM, 2:15 TO 5:00 P.M.

Session 4pUWa

Underwater Acoustics: Seabed Acoustics

Marcia J. Isakson, Chair

Applied Research Laboratory, University of Texas–Austin, 10000 Burnet Road, Austin, Texas 78758

Contributed Papers

2:15

4pUW1. Volume scattering from random porous media. Keiichi Ohkawa (Appl. Marine Phys. Div. RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149)

The new volume scattering model, which enables us to treat the scattering of Biot's slow compressional wave from random porous media, is derived applying the Born approximation to Biot's equations of motion. Within the framework of the Biot theory it is assumed that the fluid-saturated unconsolidated sediment has low values of frame bulk and shear moduli relative to the other moduli of the medium and the shear wave is negligible. This enables us to treat the Biot theory easier. The equations of motion in inhomogeneous media are then simplified and coupled Helmholtz equations for compressional waves are derived applying the mode's decoupling method to the simplified equations of motion. The Born approximation is applied to the coupled Helmholtz equations in random media. The derived equations can be treated as extended forms of fluid bottom models. In this model, the 3-D power spectrum is inferred from porosity and permeability fluctuations instead of velocity and density fluctuations generally used. The permeability fluctuation can be estimated from grain-size distribution. In the limit of low frequency or high porosity, the results coincide with the fluid bottom model [T. Yamamoto, J. Acoust. Soc. Am. **99**, 866–879 (1996)]. The differences of this new model and others will be discussed.

2:30

4pUWa2. Field measurements of attenuation and permeability of shallow water sediments. Tokuo Yamamoto (Geoacoustics Lab., AMP, RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149) and Junichi Sakakibara (JFE Civil Corp.)

The velocity and attenuation of sands and clays in shallow oceans were measured by cross-hole tomography experiments using the pseudo-random binary sequence (PRBS) as source signals with the PRBS frequency of 1 and 3 kHz. Using measured velocity images, 2D velocity and density fluctuation spectra were calculated. The acoustic attenuation due to scattering by velocity and density fluctuations in the sediment volume was calculated using the measured velocity and density fluctuation spectra. The attenuation due to scattering occupies 30 to 98% of the total attenuation measured during the field experiments. The intrinsic attenuations of sands and clays were 0.3 to 1.5 dB/m/kHz and 0.01 to 0.02 dB/m/kHz, respectively. The permeability of the sands and the clays inverted by the Biot theory is 2.0 to 14.0 darcy and 0.1 to 120 md, respectively. The acoustically inverted permeability values agreed very well with the direct measurements of permeability of the sediments. The experimental results indicate that the acoustic attenuation measured in the fields is mainly due to volume scattering and that the intrinsic attenuation is due to the Biot mechanism of pore-fluid within the sediments. [Work supported by ONR Code 321OA and Kansai Power and Light Co.]

2:45

4pUWa3. Reflection and transmission coefficients in terms of pressure for Biot's compressional waves and the importance of the shear modulus for Biot's slow wave. Keiichi Ohkawa (Appl. Marine Phys. Div. RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149)

The method to determine reflection and transmission coefficients in terms of pressure at the fluid–sediment interface is presented. In general, reflection and transmission coefficients derived from the Biot theory have been computed using displacement potentials. Within the framework of the Biot theory it is assumed that the fluid-saturated unconsolidated sediment has low values of frame bulk and shear moduli relative to the other moduli of the medium and the shear wave is negligible. This enables to treat the Biot theory easier. By introducing effective densities, reflection and transmission coefficients can be treated as if the medium is a fluid. This approach using the models decoupling method is fairly different from the effective density fluid model [K. Williams, J. Acoust. Soc. Am. **110**, 2276–2281 (2001)]. It is shown that pressure coefficients give the same results as computed using displacement potentials. This method gives the advantage of directly computing the pressure for problems of reflection, transmission, and scattering. However, boundary conditions should be carefully treated. The pressure coefficient for Biot's slow wave cannot be accurately calculated when the equilibrium of solid pressure at the interface is constrained instead of the normal traction. This disagreement demonstrates that the shear modulus plays an important role in the generation of the slow wave.

3:00

4pUWa4. Spherical wave effects in sediment reflection coefficients. H. John CaminIII and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78758, misakson@arlut.utexas.edu)

Several research institutions are investigating models to properly describe high frequency, sandy-sediment, acoustic properties because of their importance in shallow water operations. In this talk we will focus on the error contribution of spherical wave effects on each of four common models: the visco-elastic model, the Effective Density Fluid Model (EDFM), Buckinghams micro-sliding model and the Biot/Stoll poro-elastic model. This will be accomplished by comparing the known models of reflection coefficients to laboratory measurements. Theoretical data will be calculated for each of the models using OASES, which utilizes the depth-dependent Green's function and a direct global matrix approach to calculate the full field. Parameters for the visco-elastic model have been measured and verified at the test location. Parameters for the EDFM, Buckingham and Biot/Stoll models were determined from measured values and previous inversions of plane wave reflection data taken at the same location. Air/water interface data will be measured and computed for each model as a test case. Linearly Frequency Modulated (LFM) chirps over the range of 30–160 kHz will be used to establish ranges in frequency. [Work supported by ONR, Ocean Acoustics.]

3:15

4pUWa5. A comparison of high-frequency plane-wave reflection coefficient data taken from a smooth sand/water interface with current models. Andrew Worley and Marcia Isakson (Appl. Res. Labs., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78758-4423, adworley@arlut.utexas.edu)

Dispersion and attenuation data have shown that a fluid model or visco-elastic model is not adequate to describe acoustic interactions in littoral environments. This study will compare laboratory reflection data to five models: the visco-elastic model, the Biot–Stoll poro-elastic model, the effective density fluid model (EDFM), the Buckingham micro-sliding model, and the Biot–Stoll squirt flow + shear model (BISQS). The BISQS model allows for frequency dependence in the frame bulk modulus and the frame shear modulus by considering the squirt flow effects and grain shearing effects at the sand grain interfaces. This model has been shown to produce good agreement with *in situ* dispersion and attenuation data; however, it has not yet been compared to reflection data. A simulated annealing inversion algorithm with rotated coordinates will be used. Special emphasis has been placed on quantifying the effects of the statistical nature of the data on the inversion outcome. [Work supported by ONR, Undersea Signal Processing, John Teague.]

3:30–3:45 Break

3:45

4pUWa6. Experimental study of the poroelastic behavior of a sandy sediment. Michael W. Yargus (Naval Surface Warfare Ctr., Carderock Div., Detachment Bremerton, 530 Farragut Ave., Bremerton, WA 98314-5215) and Darrell R. Jackson (Univ. of Washington, Seattle, WA 98105)

Reflection measurements were conducted on water-saturated sand over the frequency range 200–300 kHz. Three reflection boundaries were used: water-sand, pressure-release-sand, and acrylic-sand. A reflection ratio technique was used to remove unknown calibration and geometric factors, permitting measurement of parameters relating to the Biot fast wave and bounding of parameters of the slow wave. The interpretation of these results was facilitated by use of a mechanical model. Measured (or bounded) parameters include acoustic impedances, effective densities, wave speeds (phase velocities), effective pressures, fluid-frame displacement ratios, pressure reflection coefficients, and material moduli. The acoustic impedance divided by the phase velocity provides the "effective density" [K. L. Williams, J. Acoust. Soc. Am. **110**, 2956–2963 (2001)] for the fast wave. As expected, the effective density was less than the total density of the sediment (effective density = 89% of total). The fluid-frame displacement ratio was 2.2 for the fast wave. These results provide strong evidence for the importance of Biot effects in water-saturated sand.

4:00

4pUWa7. Linear coupled electrokinetic-acoustic behavior of consolidated and unconsolidated granular materials. Gareth Block, John G. Harris, and Nicholas Chotiros (Appl. Res. Labs., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78758, gblock@arlut.utexas.edu)

Two electrokinetic (EK) techniques are being developed to study wave propagation in consolidated and unconsolidated granular materials. In contrast to the more common acoustical techniques, EK transduction utilizes the electrokinetic coupling between sediment grain surface chemistry and pore fluid motion. When Biot theory is applicable, coupled electrokinetic-Biot theory predicts that a macroscopic electric potential is generated by fluid motion in electrolyte-saturated porous media. (Our previous research on the reciprocal case—of creating a pressure wave by applying a high voltage impulse to electrolyte-saturated sediments—has been presented at recent ASA meetings.) To assess the validity of Biot theory as a model of ocean sediments, various EK measurements were taken in the laboratory. Preliminary data using silica sand, as well as unconsolidated and consolidated (sintered) glass microspheres, will be presented. We will also describe the coupled electrokinetic-Biot theory, the underlying volume-averaged poroelastic model upon which it is based, and related issues in homogenization theory. [Work supported by ONR, Ocean Acoustics.]

4:15

4pUWa8. The effect of marine sediments on the underwater penetration of a sonic boom. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA/Environ. Technol. Lab., Boulder, CO 80305-3328, Oleg.Godin@noaa.gov) and David M. F. Chapman (Defence Res. and Development Canada Atlantic, Dartmouth, NS B2Y 3Z7, Canada)

Understanding and quantifying sonic boom penetration into the ocean is of substantial practical interest because of the potential impact of supersonic aircraft on marine mammals. Observations by Desharnais and Chapman [F. Desharnais and D. M. F. Chapman, J. Acoust. Soc. Am. **111**, 544–553 (2002)] indicate, in qualitative agreement with numerical simulations, that sonic boom penetration in coastal regions may be significantly enhanced compared to the deep water case by the resonant excitation of seismo-acoustic waves in marine sediments. In this paper, an analytic theory originally developed to model interface waves in soft sediments with power-law shear speed profiles [O. A. Godin and D. M. F. Chapman, J. Acoust. Soc. Am. **110**, 1890–1907 (2001)] is extended to simulate the shallow-water sound field induced by a shock wave incident from the air side. The theory explains salient features of the observations, including the nonmonotonic depth dependence of the sound exposure spectral density (SESD). Our analysis demonstrates that SESD in shallow water is sensitive to the geoacoustic parameters of the seabed, particularly the shear modulus profile. It may be possible to extract the shear wave attenuation coefficient in the sediment at low frequencies by analyzing sonic boom records from a vertical line array of hydrophones.

4:30

4pUWa9. Estimation of seabed reflection properties from direct blast pulse shape. Mark K. Prior and Christopher H. Harrison (SACLANTCEN, Viale San Bartolomeo 400, 19038 La Spezia, Italy, prior@saclantc.nato.int)

A method is described by which the reflection properties of the seabed can be estimated from the fall-off with time of intensity received on a single hydrophone, tens of water depths away from a broadband source in shallow water. The method is simple and directly applicable to most active sonars. The theoretical basis of the method is described and results presented using experimental data measured in a shallow water region of the Mediterranean Sea. Seabed properties are deduced directly by the method in the form of the rate of change of reflection loss with respect to an angle close to grazing. An extension to the method is proposed whereby seabed sound speed, density and attenuation can be estimated. Good agreement is shown between the results of the method and the results of matched field inversions carried out in the same area using a large-aperture vertical line array. A similar method using target echoes, instead of direct arrivals, is also demonstrated. The features of the method that make it attractive as a candidate for through the sensor probing of the environment are discussed.

4:45

4pUWa10. Geoacoustic inversion of a shallow fresh-water environment. Steven A. Stotts, David P. Knobles, Robert A. Koch, James N. Piper (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029), and Jason A. Keller (Univ. of Texas, Austin, TX 78713-8029)

A recent experiment was conducted at The University of Texas/Applied Research Laboratories test station located at Lake Travis, Austin, TX. Implosive (light bulb), explosive (firecracker), and tonal sources were recorded on a dual receiver system located on the bottom next to a range-independent underwater river channel. Inversion results of the broadband time series obtained over ranges less than 1.5 km were used to predict measured transmission loss at several tonal frequencies in the band from 250–1000 Hz. The average water depth was approximately 38 m along the channel during the experiment. Sound speed profiles were calculated from recorded temperature readings measured as a function of depth. Implosive source spectrums were measured and used to evaluate a model/data correlation cost function in a simulated annealing algorithm. Comparisons of inversion results using both a normal mode and a ray-based plane wave reflection coefficient forward model [Stotts *et al.*, J. Acoust. Soc. Am. (submitted)] are discussed. Predicted transmission loss based on the inversion results are compared to the measured transmission loss. Differences between fluid and elastic layer bottom models will also be presented.

THURSDAY AFTERNOON, 13 NOVEMBER 2003 — SABINE ROOM, 2:30 TO 4:10 P.M.

Session 4pUWb

Underwater Acoustics, Engineering Acoustics and Signal Processing in Acoustics: Gradient Array Acoustics II

Paul C. Hines, Cochair
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Daniel L. Hutt, Cochair
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Chair's Introduction—2:30

Invited Papers

2:35

4pUWb1. Overview of vector sensors for undersea applications. James F. McEachern (Office of Naval Res., 800 N. Quincy St., Arlington, VA 22217-5660, mceachj@onr.navy.mil)

An overview of vector sensors for sonar applications is presented. The most prolific use of vector or pressure gradient sensors has been in directional sonobuoys to accomplish effective directional measurements with a sensor that is much smaller than the signal wavelength. Common characteristics, implementation issues, and self noise sources of directional hydrophones are reviewed. Arrays

of vector sensors began to emerge in the 1980s and have the attractive property of being able to provide substantially higher directivity for a smaller equivalent aperture than that required for a scalar (omnidirectional) sensor array. Dyadic sensors bring additional directivity but development is required to realize them in a compact configuration. Future applications of vector sensors are discussed. The limited amount of information relative to the ocean acoustic vector fields is noted as a corollary technological requirement for the widespread adoption of vector sensors in sonar systems.

3:00

4pUWb2. Multimicrophone-technology in hearing systems. Benefits and trends. Torsten Niederdraenk (Siemens Audiologische Technik GmbH, D-91058 Erlangen, Germany, torsten.niederdraenk@siemens.com)

Hearing impaired people often suffer from reduced communication skills in real-life situations. In order to increase the signal-to-noise ratio, directional microphones are applied in hearing instruments. Repelling noise from backward directions and focusing on sounds coming to the face of the listener, they provide the hearing impaired with better communication abilities. This contribution reports on higher-order directional microphones in hearing instruments. While the hearing aid user demands an improvement of the signal-to-noise ratio, the directional microphones of hearing instruments should be very small or rather invisible. Some higher-order directional microphone solutions that are positioned on top of a spectacle frame or body-worn devices are already known on the market. Of course, these arrangements are not very practicable in use and cosmetically unappealing. The only relevant approach for the customer seems to be the integration of higher-order directional microphones in the hearing instrument. In terms of the distorted sound field around the head and small microphone distances a special approach has been implemented, that adaptively combines directive microphones of different order.

Contributed Papers

3:25

4pUWb3. Sound source localization with a gradient array using a coherence test. Satish Mohan, Michael E. Lockwood, Douglas L. Jones (Beckman Inst., Univ. of Illinois at Urbana-Champaign, 405 N. Matthews Ave., Urbana, IL 61801, smohan@uiuc.edu), Quang Su, and Ronald N. Miles (State Univ. of New York, Binghamton, NY 13902-6000)

The localization of sound sources is difficult when the number of sources is greater than the number of sensors. An algorithm is described that can localize in such acoustic scenes by utilizing a statistical coherence test to identify desirable time–frequency bins where the MUSIC and minimum-variance spectral estimators are applied. Localization results are then obtained by integrating across the selected time–frequency bins. The algorithm was used to localize signals in azimuth and elevation from a co-located microphone array consisting of one omnidirectional microphone and three gradient microphones. With 2.5 seconds of five speech sources, the azimuth and elevation estimates produced by the algorithm, over a hundred realizations of additive white Gaussian noise, had biases under four degrees and standard deviations less than two degrees. The algorithm was also used to compare the localization performance, in azimuth, of an array of two co-located gradient microphones with an array of two 15-cm-spaced omnidirectional microphones in the free-field.

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4pUWb4. Distortion of interfering speech in the aggregate beamformer. David I. Havelock (Natl. Res. Council, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada)

The aggregate beamformer is an alternative to conventional beamformers. It samples an array of sensors randomly, reducing the undesired off-beam signals to noise. Important advantages of this technique over conventional beamforming are the reduced front-end hardware requirements, such as anti-alias filters, the ability to operate at a lower total sampling rate, particularly for arrays with many elements, and improved beamforming time-delay resolution without the need for interpolation. The aggregate beamformer output contains residual noise that is proportional to the level of interfering off-beam signals. On-beam signals do not cause residual noise. The residual noise level is controlled by adjusting the total sampling rate. In speech applications, the residual noise is perceived as a distortion of the off-beam signal. This distortion may help to discriminate desired (on-beam) and undesired (off-beam) speech. The principles of operation of the aggregate beamformer are described and a demonstration of the residual noise in a speech pick-up application is presented.

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4pUWb5. Beamforming with collocated microphone arrays. Michael E. Lockwood, Douglas L. Jones (Beckman Inst., Univ. of Illinois at Urbana-Champaign, 405 N. Matthews Ave., Urbana, IL 61801, melockwo@uiuc.edu), Quang Su, and Ronald N. Miles (State Univ. of New York, Binghamton, NY 13902-6000)

A collocated microphone array, including three gradient microphones with different orientations and one omnidirectional microphone, was used to acquire data in a sound-treated room and in an outdoor environment. This arrangement of gradient microphones represents an acoustic vector sensor used in air. Beamforming techniques traditionally associated with much larger uniformly spaced arrays of omnidirectional sensors are extended to this compact array (1 cm^3) with encouraging results. A frequency–domain minimum-variance beamformer was developed to work with this array. After a calibration of the array, the recovery of sources from any direction is achieved with high fidelity, even in the presence of multiple interferers. SNR gains of 5–12 dB with up to four speech sources were obtained with both indoor and outdoor recordings. This algorithm has been developed for new MEMS-type microphones that further reduce the size of the sensor array.