Architectural Acoustics: Acoustics of Virtual Environments

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Chair's Introduction—8:30

Invited Papers

8:35

4aAA1. Application-driven design of auralization systems. Durand R. Begault (NASA Ames Res. Ctr., MS 262-2, Moffett Field, CA 94035-1000, Durand.R.Begault@nasa.gov)

Acoustical environmental simulation (auralization) involves rendering a binaural signal that articulates simultaneously the positional information of a sound source and the source's acoustic and vibratory interaction with its environmental context. This places significant challenges on both the modeling and rendering components of such a system. Overcoming real-time limitations of processors can be accomplished by implementing rendering limits based on auditory threshold data for both early reflections and late reverberant energy. However, final assessment of system quality is dependent on the specific task or goal of the simulation. The use of auralization in accomplishing a telerobotic task requires head tracking and low latency, while environmental cues such as obstacles with acoustic reflections can be simplified and exaggerated beyond veridical representation. Contrasting this is the task of assessing speech intelligibility of an emergency public address system, or the quality of a space for musical performance, requiring a more accurate level of simulation of acoustical materials and sound-source characteristics. Modeling accuracy is complicated by such factors as the level of variance in the absorptive and diffusive properties of materials in terms of their real-world application. Assessment of auralization quality must therefore involve a best estimate of significant factors for a particular application.

8:55

4aAA2. The importance of perceived self-motion in experiencing convincing virtual acoustic rendering. William L. Martens (Faculty of Music, McGill Univ., Montreal, QC H3A 1E3, Canada, wlm@music.mcgill.ca)

One factor that makes a virtual acoustic environment more convincing is the capacity to present information that enables a listener to answer questions about the geometry of the acoustical space in which they are immersed. This paper reviews the results of several studies in which virtual sound sources were presented to listeners via a loudspeaker array located in an anechoic chamber, and asked about details such as the location of a missing wall in a simulated rectangular room, and whether or not a reduction in the loudness of a virtual source was due to the interposition of a virtual wall between source and listener. When a listener was unaware of changes in the virtual listening position, such questions were difficult to answer; however, when listeners were informed of their path through the environment via computer graphic animation, performance improved significantly. The results of these perceptual experiments suggest that perceived self-motion may be an important determinant of how convincing virtual acoustic rendering may be. The identity of the sound source itself was also found to have a substantial impact, since listeners reported self-motion more often when presented with sound sources that were judged less likely to be in motion.

Contributed Papers

9:15

4aAA3. Active field control for piano room. Atsuko Ito and Yasushi Shimizu (Adv. System Development Ctr., Yamaha Corp., 10-1 Nakazawa-cho, Hamamatsu 430-8650, Japan, aito@beat.yamaha.co.jp)

Active field control (AFC) is an innovative system that employs electroacoustic technologies enabling "natural changes" to major auditory impressions, such as reverberance, loudness, and spaciousness. This talk presents typical design concept of AFC application for a piano room with a low ceiling in architecture and with a miniconcert in program. The aim is to accommodate a condition appropriate for both a solo piano performance and a miniconcert, regardless of the low ceiling height and significant increase of absorption by the audience in a miniconcert. This can be realized by using an AFC system, which converts the room into different acoustical conditions as follows. One is that AFC alters acoustics of the designed space, and makes it sound as if it has a higher ceiling, which leads to a longer reverberation. The other is negative absorption to compensate added absorption with the audience for a miniconcert. In the presentation, these technologies will be presented, along with the acoustical data.

9:30


When based on geometrical acoustics, computational models used for auralization of auditorium sound fields are physically inaccurate at low frequencies. To increase accuracy while keeping computation tractable, hybrid methods using computational wave acoustics at low frequencies have been proposed and implemented in small enclosures such as simplified models of car cabins [Granier et al., J. Audio Eng. Soc. 44, 835–849 (1996)].


4AA5. A binaural Web-based tour of the acoustics of Troy Music Hall. Rendell R. Torres, James Cooney, and Yasushi Shimizu (Prog. in Architectural Acoust., Rensselaer Polytech. Inst., Greene Bldg., 110 8th St., Troy, NY 12180-3590, rrtorres@rpi.edu)

For classical music to become more widely enjoyed, it must sound exciting. We hypothesize that if people could hear examples of truly exciting acoustics, classical music would be perceived less as a rarefied delicacy and more as a viscerally engaging listening experience. The Troy Savings Bank Music Hall in Troy, New York, is a legendary 1200-seat concert hall famous for its acoustics. Such landmarks are commonly documented architecturally but with few attempts to document their acoustics in a way that it is listenable. Thus, the goal is to capture and sonically disseminate the hall’s acoustics through a Web-based acoustical tour, where one can click on various seats to hear binaural auralizations of different instruments and see corresponding views of the stage. The hope is that these auralizations will not only sonically document the acoustics of the hall but also tantalize even geographically distant listeners with binaural samples of how exciting music can be in excellent acoustics. The fun and challenges of devising (let alone standardizing) such an auralization-based system of documentation will be discussed, and a demonstration given. This process can be applied to other historically and acoustically significant spaces. [Work supported by the National Endowment for the Arts.]

10:00–10:15 Break

10:15


Digital waveguide mesh (WGM) models have been shown to be a viable method of obtaining accurate room impulse responses (RIRs) for a virtual space. However, the large memory and long processing requirements of these models have restricted their use for large rooms and for those with nontrivial geometric features. The development and inclusion of a KW-pipe interface allows the interconnection of finite difference and wave-based mesh implementations of a WGM model. The former is efficient for the main body of the mesh, with the latter, more processor intensive method allowing more accurate simulation at the boundaries of the modeled space. The resultant hybrid model removes some of the above computational constraints, allowing larger rooms and more complex geometries to be modeled. Model visualizations and RIR data are presented, demonstrating the correct operation of the KW-pipe interface. RIR data from two differing mesh topologies show that nontrivial geometries can be successfully modeled using this technique, with considerable computational savings on purely wave-based WGMs. Comparisons with RIRs obtained through more traditional ray-tracing and image-source techniques show favorable results and demonstrate the complex wave phenomena that are inherent in WGM models. [Work supported by EPSRC.]

10:30


Because of the spontaneity and high level of call and response, many charismatic churches have verbal and musical communication problems that stem from highly reverberant sound fields, poor speech intelligibility, and muddy music. This research looks at the subjective dimensions of room acoustics perception that affect a charismatic worship space, which is summarized using the acronym RISCS (reverberation, intimacy, strength, coloration, and spaciousness). The method of research is to obtain acoustical measurements for three worship spaces in order to analyze the objective parameters associated with the RISCS subjective dimensions. For the same spaces, binaural room impulse response (BRIR) measurements are done for different receiver positions in order to create an auralization for each position. The subjective descriptors of RISCS are analyzed through the use of listening tests of the three auralized spaces. The results from the measurements and listening tests are analyzed to determine if listeners’ perceptions correlate with the objective parameter results, the appropriateness of the subjective parameters for the use of the space, and which parameters seem to take precedence. A comparison of the multi-source auralization to a conventional single-source auralization was done with the mixed down version of the synchronized multi-track anechoic signals.

10:45

4AA8. Sensitivity of room acoustic parameters to changes in scattering coefficients. Jonathan Rathsam and Lily M. Wang (Architectural Eng. Prog., Univ. of Nebraska–Lincoln, Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0681, jrrathsam@mail.unomaha.edu)

This project uses the room acoustics computer modeling program, ODEON, to investigate the sensitivity of room acoustic parameters to changes in scattering coefficients. Particularly, the study is interested in determining if the results from certain room models are more sensitive to scattering coefficients than from other models, due to their geometry or absorption characteristics. If so, how can one quantify a model’s susceptibility to being sensitive to scattering? Various models of three real spaces in Omaha, Nebraska are tested. The predicted reverberation, clarity, and spaciousness parameters are compared at various receiver locations, while the scattering coefficient of all surfaces is varied from 0 to 0.1, 0.3, 0.5, and 0.8. The resulting data are analyzed by frequency according to the (1) average absorption of the room; (2) magnitude variation of absorption within the room; (3) spatial distribution of absorption within the room; and (4) level of model detail. Initial results indicate that parameters studied may show more sensitivity to scattering coefficients in models that have a wider range of absorption values, more disparate distribution of absorption, and lower detail level. Various schemes that include these aspects are proposed for computing a model’s sensitivity to changes in scattering.

11:00

4AA9. Investigation of the optimum acoustical conditions for speech using auralization. Wonyoung Yang and Murray Hodgson (School of Occupational and Environ. Hygiene, Univ. of BC, 2206 East Mall, Vancouver, BC V6T 1Z3, Canada, wyang@interchange.ubc.ca)

Speech intelligibility is mainly affected by reverberation and by signal-to-noise level difference, the difference between the speech-signal and background-noise levels at a receiver. An important question for the design of rooms for speech (e.g., classrooms) is, what are the optimal values of these factors? This question has been studied experimentally and theoretically. Experimental studies found zero optimal reverberation time, but theoretical predictions found nonzero reverberation times. These contradictory results are partly caused by the different ways of accounting for background noise. Background noise sources and their locations inside the room are the most detrimental factors in speech intelligibility. However,

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were played in the virtual room and auralized for listeners. The Modi
a virtual room with known reverberation. Speech intelligibility test signals
using speech and noise sources of known relative output levels located in
major room-acoustical factors for speech intelligibility were controlled
noise levels also interact with reverberation in rooms. In this project, two
major room-acoustical factors for speech intelligibility were controlled
using speech and noise sources of known relative output levels located in
a virtual room with known reverberation. Speech intelligibility test signals
were played in the virtual room and auralized for listeners. The Modified
Rhyme Test (MRT) and babble noise were used to measure subjective
speech intelligibility quality. Optimal reverberation times, and the optimal
values of other speech intelligibility metrics, for normal-hearing people
and for hard-of-hearing people, were identified and compared.

11:15
4aAA10. An evaluation of differences due to changing source
directivity in room acoustic computer modeling. Michelle C. Vigeant
and Lily M. Wang (Architectural Eng. Prog., Univ. of Nebraska–Lincoln,
Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0681,
mvigeant@unlnotes.unl.edu)

This project examines the effects of changing source directivity in
room acoustic computer models on objective parameters and subjective
perception. Acoustic parameters anduralizations calculated from omni-
directional versus directional sources were compared. Three realistic di-
rectional sources were used, measured in a limited number of octave bands
from a piano, singing voice, and violin. A highly directional source that
beams only within a sixteenth-tant of a sphere was also tested. Objec-
tively, there were differences of 5% or more in reverberation time (RT)
between the realistic directional and omnidirectional sources. Between the
beamming directional and omnidirectional sources, differences in clarity
were close to the just-noticeable-difference (jnd) criterion of 1 dB. Sub-
jectively, participants had great difficulty distinguishing between the real-
istic and omnidirectional sources; very few could discern the differences
in RTs. However, a larger percentage (32% vs 20%) could differentiate
between the beaming and omnidirectional sources, as well as the respec-
tive differences in clarity. Further studies of the objective results from
different beaming sources have been pursued. The direction of the beam-
ing source in the room is changed, as well as the beamwidth. The objective
results are analyzed to determine if differences fall within the jnd of
sound-pressure level, RT, and clarity.

THURSDAY MORNING, 27 MAY 2004
LIBERTY 3, 8:00 A.M. TO 12:05 P.M.

Session 4aAB

Animal Bioacoustics: Natural Acoustic Behavior of Animals: Session in Memory of Donald R. Griffin III

Andrea M. Simmons, Chair
Department of Psychology, Brown University, Providence, Rhode Island 02912

Invited Paper

8:00

4aAB1. Acoustic behavior of echolocating bats in complex environments. Cynthia Moss, Kaushik Ghose, Marianne Jensen (Dept. of Psych., Inst. for Systems Res., Univ. of Maryland, College Park, MD 20742, cmoss@psyc.umd.edu), and Annemarie Surlykke (Univ. of Southern Denmark, Odense, Denmark)

The echolocating bat controls the direction of its sonar beam, just as visually dominant animals control the movement of their eyes
to foveate targets of interest. The sonar beam aim of the echolocating bat can therefore serve as an index of the animal’s attention to
objects in the environment. Until recently, spatial attention has not been studied in the context of echolocation, perhaps due to the
difficulty in obtaining an objective measure. Here, we describe measurements of the bat’s sonar beam aim, serving as an index of
acoustic gaze and attention to objects, in tasks that require localization of obstacles and insect prey. Measurements of the bat’s sonar beam aim are taken from microphone array recordings of vocal signals produced by a free-flying bat under experimentally controlled conditions. In some situations, the animal relies on spatial memory over reflected sounds, perhaps because its perceptual system cannot easily organize cascades of echoes from obstacles and prey. This highlights the complexity of the bat’s orientation behavior, which can alternate between active sensing and spatial memory systems. The bat’s use of spatial memory for orientation also will be addressed in this talk. [Work supported by NSF-IBN-0111973 and the Danish Research Council.]

Contributed Papers

8:20
4aAB2. Dolphin and bat sonar: Convergence, divergence, or parallelism. Darlene R. Ketten (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543) and Harvard Med. School, Boston, MA 02114, diketten@whoi.edu), James Simmons (Brown Univ., Providence, RI 02912), Allyn E. Hubbard, and David A. Mountain (Boston Univ., Boston, MA)

During the explosive period of mammalian radiation, two groups emerged with highly effective biosonar systems, bats and toothed whales. In the intervening 50 million years, these groups evolutionarily honed their hearing for operation in radically different media. This paper addresses what functional aspects the media influenced in the biosonar receptors of bats versus dolphins by comparing the auditory peripheries of these groups. Data were obtained using thin-section microscopy, CT imaging, and inner-ear models. Inner-ear anatomy is fundamentally similar in these animals, although differences exist in both neural density and distribution in each group. Specialist ears are present in both groups, suggesting at least one odontocete species has cochlear specializations consistent with CF-FM bats, including specialized basilar-membrane regions and high-frequency neural foveal areas. Cochlear specializations in both groups are primarily linked to peak spectra of sonar signals, may expand frequency representation, and may enhance tuning in adjacent ear segments by generating standing wave phenomena. Most differences, such as the soft-tissue external ear analogs in odontocetes, are clearly media driven. Other differences among species within each group are correlated with signal type or habitat complexity. [Work supported by Mellon Foundation; Seaver Institute; ONR.]

8:35
4aAB3. Biosonar signal processing of bats during flight observed by a telemetry microphone on the head. Hiroshi Riquimaroux, Yoshiaki Watanabe (Fac. Eng., Doshisha Univ., Kyotanabe 610-0321, Japan, hrikimar@mail.doshisha.ac.jp), and Liang-Kong Lin (Tunghai Univ., Taichung, Taiwan)

The purpose of the present study was to examine the daily consistency of constant frequency component (CF) of CF-FM bats and to examine biosonar sound characteristics of leaf-nosed bats (Hipposideros terasen-sis) during flight by a telemetry microphone system (telemike) mounted on the bat’s head. Experiments were done within a steel-walled chamber. When at rest, the fundamental frequency of the bat was about 35 kHz. The second harmonic (70 kHz) was the most intense. The second harmonic of CF component (CF2) at rest and during flight varied significantly over several months. A systematic change in CF2 frequency during flight was observed through the telemike. Doppler-shift compensation in CF2 frequency, amplitude compensation, and pulse emission rate just before landing were confirmed. We will discuss time-sharing dual signal processing found during flight. [Research supported by Special Coordination Funds, and a grant to RCAST at Doshisha University from the Ministry of Education, Culture, Sports, Science and Technology of Japan.]

8:50
4aAB4. A portable system for marine mammal auditory-evoked potential measurements. James J. Finneran (U.S. Navy Marine Mammal Prog., Space and Naval Warfare Systems Ctr., San Diego, Code 2351, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil) and Dorian S. Houser (Biometrica, La Mesa, CA 91942)

Limitations to behavioral measures of hearing sensitivity in marine mammals include the time and expense typically required to train subjects. These limitations have resulted in limited subjects and lingering questions regarding intraspecifc variability. An alternative to behavioral methods is the electrophysiological method, where passive electrodes are used to measure auditory-evoked potentials (AEPs) generated by the brain in response to sound stimuli. Marine mammal AEP measurements have been limited by the complexity of the technique and the limited applicability of commercially available AEP systems. In this paper, a portable, laptop computer-based system for marine mammal AEP measurements will be presented. The system features commercial off-the-shelf components, including a data acquisition PC card, biopotential amplifier, and program-mable attenuator. The system is housed in a rugged, shock-resistant case. Custom software is used to present sound stimuli, record evoked responses, and analyze the resulting data. The system has been used to measure auditory brainstem responses to clicks and tone pips and envelope following responses to amplitude-modulated tones in bottlenose dolphins. Preliminary data obtained with the system will be presented and compared to behavioral hearing measures. [Work supported by the ILIR at SPAWAR SYS-SD and the ONR.]

9:05
4aAB5. Ecological echoes observed by moving biomimetic sonar characterize objects. Roman Kuc (Dept. of Elec. Eng., Yale Univ., New Haven, CT 06520-8267)

This paper examines echoes from in situ foliage, similar to those observed by flying bats. A moving sonar converts echoes into spike sequences and applies neural-computational methods to estimate ranging and travel speed. Retro-reflectors and favorably oriented planar patches generate strong echoes (SEs), identified by spike firing rate. Linear sonar trajectories cause SEs to form hyperbolic patterns, termed glints, specified by passing range and travel speed. Passing-range specific detectors compare successive SE times with values in a table and tally coincidences. A glint terminates after a sufficient number of coincidences are tallied and two consecutive mismatches occur in the maximum-count detector. SE arrival jitter necessitates a coincidence window. Short windows identify individual glints while long windows generalize extended objects. SEs from distant objects exhibit almost constant incremental delays used to estimate sonar travel speed, necessary for robust glint detection. Passing-range estimates may explain how bats can fly through small openings without collision.

9:20
4aAB6. How bats’ ears probe space: A numerical analysis of pinna shapes. Rolf Müller, John C. T. Hallam (Maersk Inst., Univ. of Southern Denmark, DK-5230 Odense M, Denmark, rolfm@mpi-sdu.dk), Herbert Peremans (Univ. of Antwerp, BE-2000 Antwerp, Belgium), Alexander Streicher, and Reinhard Lerch (Univ. of Erlangen-Nürnberg, D-91052 Erlangen, Germany)

Like any antenna, bat pinnae have directivity patterns which impose a frequency-dependent sensitivity weighting on possible locations of sonar targets. Here, numerical predictions of such directivities are presented. The predictions are based on finite-element meshes generated from computer-tomographic cross-section images of pinna samples obtained from several bat species. A time-domain finite-element model was combined with a near-field to far-field transformation to yield high-resolution directivity estimates. The data obtained enable the introduction of new visualization approaches to the study of bat biosonar. These techniques portray the spatial and frequency dimension of the patterns, facilitate quantitative analysis, and reveal functional properties relevant to specific sensing tasks, like scanning the surface of an extended target. The computer graphics representations of the pinna shapes (voxel arrays, volume and surface meshes) can be readily manipulated to shed light onto how the
observed properties arise. For example, appendages have been removed by “Boolean surgery” or altered in orientation or shape by standard computer graphics transformations. The results of these manipulations are compared with respect to wave-field amplitudes as well as with respect to directivity patterns. [Work supported by the European Union (CIRCE Project, IST-2001-35144).]

9:35
4AB7. Digital neuromorphic processing for a simplified algorithm of ultrasonic reception. Lin Qiang and Chris Clarke (Dept. of Elec. & Electron. Eng., Univ. of Bath, Bath BA2 7AY, UK, L.Qiang@bath.ac.uk)

Previously, most mammalian auditory systems research has concentrated on human sensory perception whose frequencies are lower than 20 kHz. The implementations almost always used analog VLSI design. Due to the complexity of the model, it is difficult to implement these algorithms using current digital technology. This paper introduces a simplified model of biosonic reception system in bats and its implementation in the “Chiroptera Inspired Robotic CEPhaloid” (CIRCE) project. This model consists of bandpass filters, a half-wave rectifier, low-pass filters, automatic gain control, and spike generation with thresholds. Due to the real-time requirements of the system, the system employs Butterworth filters and advanced field programmable gate array (FPGA) architectures to provide a viable solution. The ultrasonic signal processing is implemented on a Xilinx FPGA Virtex II device in real time. In the system, 12-bit input echo signals from receivers are sampled at 1 M samples per second for a signal frequency range from 20 to 200 kHz. The system performs a 704-channel per ear auditory pipeline operating in real time. The output of the system is a coded time series of threshold crossing points. Comparing hardware implementation with fixed-point software, the system shows significant performance gains with no loss of accuracy.

9:50
4AB8. Off-axis signal processing of cetacean biosonar. Walter M. X. Zimmer (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, I-19138 La Spezia, Italy, walter@saclantc.nato.int)

Echolocation or biosonar plays a fundamental role for odontocetes to probe their environment, and their characteristics have been studied extensively for over 40 years. In summary, cetacean biosonar can be modeled as broadband transient-like signals radiating from a finite piston or aperture. The resulting sonar beam is directional, with a directivity index exceeding 25 dB for some species. The estimation of relevant sonar parameters is usually obtained from animals that are kept within a controlled and well-instrumented environment. This paper shows how measurements of opportunity from free-ranging odontocetes may be used to obtain their relevant biosonar parameters, in particular directivity index and source level. Frequently these measurements are made with a single hydrophone that is sufficiently deep so that surface-reflected echoes separate from the direct arrival of the echolocation clicks. Also, the received signal often fades in and out as the sonar beam of the scanning animal crosses the hydrophone. The presented technique exploits this scattering and the observation that broadband signals from a finite aperture will appear distorted when recorded off the acoustic axis, as the transfer function of the aperture modifies the spectrum of the transmitted signal.

10:05-10:20 Break

10:20
4AB9. The whistles of bottlenose dolphins (Tursiops truncatus) from the Gulf of Mexico. Carmen Bazua-Duran (Lab. Acustica Aplicada y Vibraciones, CCADET, UNAM, Cd. Universitaria, 04510 Mexico DF, Mexico, bazua@servidor.unam.mx)

This work presents the description and geographic comparison of whistles from bottlenose dolphins recorded in three coastal areas in the northwestern Gulf of Mexico (Galveston and Corpus Christi Bays and Madre Lagoon), one oceanic area in the northern Gulf, and one coastal area in the southern Gulf (Terminos Lagoon). The 1499 whistle contours analyzed were categorized into 289 whistle types, of which 120 types were unique to a specific area. From the remaining 169 types, more types were common between Madre and Corpus Christi than between Galveston and either Corpus Christi or Madre, results in agreement with the dolphin mixing patterns between these three areas. Whistles from the oceanic area were more similar to those of Madre, suggesting that contact between coastal and oceanic dolphins in the northwestern part of the Gulf may be through Madre Lagoon. Terminos and Galveston whistles, the areas further apart, were very similar, indicating that contact between dolphin groups may not be the only parameter determining whistle repertoire similarities. Dolphin whistle similarities may also depend on comparable habitat use and population structure. The new signal type curve shows that more than 250 whistles are needed for each area in order to adequately describe the whistle repertoire.
whales is obtained independent of the separation distance between the two hydrophones. However, knowledge of the separation distance between the hydrophones provides the 3D coordinates of the whales within a left–right ambiguity. Whale tracks have been obtained using actual data collected off Ogasawara Islands in Japan. These results demonstrate the utility of this method for studying the bioacoustics and behavior of deep diving sperm whales.

II:05

4aAB12. The “gunshot” sound produced by male North Atlantic right whales and its potential function in reproductive advertisement. Susan E. Parks (Cornell Bioacoustics Res. Prog., 159 Sapsucker Woods Rd., Ithaca, NY 14850, sep6@cornell.edu), Philip Hamilton, Scott D. Kraus (New England Aquarium, Boston, MA 02110), and Peter L. Tyack (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

North Atlantic right whales (Eubalaena glacialis) commonly use sound to mediate social interactions between individuals. Surface active groups (SAGs) are the most commonly observed social interaction on the summer feeding grounds. These groups are typically composed of an adult female with two or more males engaged in social behavior at the surface. Several distinct types of sounds have been recorded from these groups. One sound commonly recorded from these groups is a brief broadband sound, referred to as a gunshot sound because it sounds like a rifle being fired. This sound has been recorded in the Bay of Fundy, Canada from both lone whales (N = 9) and social SAGs (N = 49). Those lone whales producing gunshot sounds whose sex could be determined (N = 9) were all mature males. In surface active groups, the rate of production of gunshot sounds was weakly correlated with the total number of males present in the group. Given the behavioral contexts of gunshot sound production by male whales, gunshots probably function in a reproductive context as an agonistic signal directed toward other males, an advertisement signal to attract females, or a combination of the two functions.

II:20

4aAB13. Influence of ambient noise on the use of sound by marine animals. Douglas H. Cato (Defence Sci. & Tech Org., and Univ. of Sydney Inst. of Marine Sci., P.O. Box 44, Pyrmont, NSW 2009, Australia, doug.cato@dsto.defence.gov.au)

Ambient noise provides the basic limitation on the use of sound by animals since signals of interest must be detected against the noise. In the ocean, ambient noise is very variable, both temporally and spatially. Temporal variation is more than 30 dB, with 20 dB being common and occurring over time scales of order 1 day, causing substantial variation in audible distances of sources (order a factor of 10). This paper examines how ambient noise affects the use of sound by animals and how it may have shaped aspects in development acoustic function. It draws on a large amount of data from the Australian region which is relatively free of anthropogenic noise, and contains a wide range of biological activity. Characteristics of ambient noise are compared with those of some marine animal vocalizations to assess whether these have any optimization against the noise. Although low acoustic absorption loss in water provides the potential for sound to be used over large distances, the variability of ambient noise and propagation suggests that high source levels of marine animals are needed to allow efficient use of sound over moderate distances most of the time, rather than large distances under exceptional conditions.

11:35

4aAB14. The impact of observer dynamics on sonar perception. Richard J. Rikoski (NSWC Panama City, Code R11—Robotic Technologies Branch, 110 Vernon Ave., Panama City, FL 32407, RikoskiRJ@ncsc.navy.mil)

The minimum information necessary for a moving observer to track objects depends on body dynamics, sensor design, sensor configuration, and environmental perturbations. This paper will present governing equations relating sonar measurements to 6 DOF dynamics, and eight nondimensional parameters describing the transitions between various regimes. Using nondimensional analysis, it will be shown that to minimize second-order effects, an observer should increase its ping rate as it approaches a target. Bats have been observed to increase their ping rate as they approach a target [Altrichamg, Bats: Biology and Behavior (1996)]. Alternatively, an observer might use a target’s angular rate to predict second-order range effects. This requires the angular rate to become observable prior to second-order range effects becoming observable. This yields a design constraint relating velocity, ping rate, and aperture, but not wavelength. Based on this constraint, it appears some marine mammals could track objects in the second-order regime [Au, The Sonar of Dolphins (1993)]. [Work supported by ONR.]

11:50


Data from the August 2003 experiment conducted by the Scripps Institution of Oceanography in the Southern California Offshore Range (SCORE) are used to localize and track marine mammals. SCORE is a naval training area near the island of San Clemente located in relatively shallow water. The water depth where the experiment was conducted is around 360 m. Data were recorded on a 100-m, eight-element vertical array deployed from the Floating Instrument Platform (FLIP) and four bottom-mounted seismometers deployed in an area covering approximately three square kilometers. During the course of the 7-day experiment continuous recording of the ocean environment was made. The recordings contain numerous blue and fin whale calls. Matched field processing was used on the vertical array data to localize and track singing whales. The differences between different animal calls (extended, low frequency calls in the case of blue whales and short, impulsive calls in the case of fin whales) are exploited to track different animal species. Animals were also independently tracked by comparing the predicted (computed using a propagation model) and measured difference in time of arrival recorded in each seismometer pair. The tracking results obtained from the two techniques are compared.
Session 4aAO

Acoustical Oceanography and Animal Bioacoustics: D. Van Holliday Special Session on Acoustical Measurements of Marine Organisms I

Orest I. Diachok, Chair
3272 Fox Mill Road, Oakton, Virginia 22124

Chair’s Introduction—8:30

Invited Papers

8:40

4aAO1. The beginning of Holliday’s underwater bioacoustics phase. Paul E. Smith (SIO, Univ. of California, San Diego, CA 92093)

When Van Holliday entered graduate school, research on and management of coastal schooling fishes were in a crisis of disarray. Lasker’s sardine physiological work had yielded the unlikely result that the adult sardine had, at one time in the 1930s, consumed most of the production of the California Current. In the analysis of possible errors we had to know: (1) the swimming speed; (2) the size distribution; and (3) the depth distribution of fish in the schools. There were also opportunities. The new DAVID STARR JORDAN, a 50-m research stern trawler, had a Norwegian polished array of WWII tube-type acoustics with steerable powerful sonars at 11 and 30 kHz. This author had equipped the vessel with borrowed trawls. UCSD/SIO/MPL Professors Vic Anderson and Fred Spiess recommended that we add a research program to study the frequency content of acoustic signals. They also nominated a graduate student, Holliday, to design and lead this fisheries research. Van worked at a furious pace to assemble the sources and processors to carry out this work and then doubled and tripled that pace in the 20 months of sea work. Implementation of Holliday’s surveys was trapped in the technological dead zone of NOAA.

9:00

4aAO2. Critical scales for understanding the structure, dynamics, and impacts of zooplankton patches. Percy Donaghay (Grad. School of Oceanogr., Univ. of Rhode Island, Narragansett, RI 02882)

The development and application of high resolution acoustic systems by Holliday and Greenlaw has played a key role in demonstrating that zooplankton could form highly concentrated layers ranging in thickness from a few decimeters to a few meters. These thin layers are remarkable in that they persist longer and have higher spatial coherence than multi-meter thick patches that can be sampled by nets and conventional acoustics. These results are inconsistent with the classical theory that thicker patches are more persistent and thus more important. Simultaneous measurements of temporal and spatial changes in the finescale physical, optical, chemical structure have provided important insights into the mechanisms controlling not only the dynamics of thin zooplankton layers but also their impacts on marine ecosystems and the transmission of sound in the ocean. [Work supported by grants from the ONR Biology and Chemistry program.]

9:20

4aAO3. A tribute to Van Holliday: Completing the circle from plankton to whales. William C. Cummings (5948 Eton Ct., San Diego, CA 92122-3203, oshundbs@san.rr.com)

After the Bachelor’s and Master’s degrees from the University of Texas and a Doctorate from the University of California, San Diego, Van Holliday hit the water with both feet moving as fast as possible, nary sinking a bit to this day. From the outset he has been a leader and innovator in methods of high-frequency sonar to identify and assess individual animals, populations, and biomasses from zooplankton to adult fishes. Although we’ll revisit some of those highlights, my 35-year association with this distinguished scientist also included an opportunity to share comparatively little known acoustics research at the highest marine trophic level, the pinnipeds and whales. Van’s scientific endeavors, together with extremely talented colleagues, remain tireless yet somehow he always finds somewhithal to unselfishly help others as testimony to the adage that whenever help is needed one had best seek out their busiest acquaintance. In recognition of his contributions to basic and applied marine science, the U.S. Navy recently bestowed upon Van its highest award for Science and Technology. It will be a pleasure and honor to tell you more of this valued friend and colleague, because you will never hear about it from the scientist, mentor, and gentleman himself.

9:40

4aAO4. Vertical array measurements of humpback whale songs. Whitlow W. L. Au, Marc O. Lammers (Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734, wau@hawaii.edu), Adam A. Pack, and Louis Herman (Kewalo Basin Marine Mammal Lab., Honolulu, HI 96814)

The songs of eight male humpback whales were recorded at ranges varying from 20 to 40 m with a vertical array of hydrophones that had a flat frequency response to 24 kHz. The songs consisted of bursts of sounds called units. Units were organized into phrases and phrases into themes. Most of the units had mean duration between 1 and 2 s and mean silent periods between units between 1 and
2 s. Many of the recorded songs contained units that had high-frequency harmonics that extended beyond 22 kHz. These harmonic results suggest that humpback whale songs have a broadband quality not previously reported and may provide some insights on the high-frequency limit of hearing in these whales. The source levels of the songs were also estimated by considering the root-mean-square sound-pressure level referenced to 1 m for the unit with the largest level for different phrase within a song. Source levels varied between 171 to 189 dB re: 1 μPa. Singing escorts have been regularly observed within two whale lengths of females and these observations and knowledge of source levels provide estimates of sound-pressure levels that male humpback whales expose female whales to.

10:00–10:15  Break

10:15


Van Holliday completed his Ph.D. dissertation, entitled “Resonance and Doppler Structure from Pelagic Fish Schools,” in 1972. He soon published two journal articles based on this research, marking the first steps in a long, distinguished and ongoing contribution to fisheries science. He has published extensively to document his acoustic research on marine life, covering the size spectrum from plankton to marine mammals. This entire body of work is of interest to fisheries biologists who strive to understand the dynamics of populations at all trophic levels to help them better understand the fish populations they study. From this perspective, his advances in acoustic methods, technologies, and instrumentation, and his extensive biological and ecological research should all be considered important contributions to fisheries science. Trained as a physicist, his unique ability to transcend the barriers between physical and biological scientists has been elemental to his success. We recognize Van Holliday for his groundbreaking acoustic research on fish and other forms of marine life, and his scientific and technical excellence. We also recognize Van for his leadership, encouragement, mentoring, and support of colleagues and young scientists, and the vision and focus that he continues to bring to the field of fisheries acoustics worldwide.

10:35

4aAO6. Light and lunar cycle as cues to diel migration of a sound-scattering layer. Kelly J. Benoit-Bird and Whitlow W. L. Au (Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734, benoit@hawaii.edu)

The Hawaiian mesopelagic boundary community is an island-associated midwater scattering layer comprised of small fishes, shrimps, and squids that undergoes diel vertical as well as horizontal migrations. It has been hypothesized that light levels are an important cue or trigger for vertical migration and presumably, horizontal migration. The migration pattern of the scattering layer was measured over complete lunar cycles while the incident light levels were recorded. Due to differences in the rise and set times of the moon and cloud cover, light and lunar cycle were not completely coupled, allowing separation of the light effects of moon phase and other cues associated with lunar cycle. Four calibrated echosounder moorings were deployed with approximately even spacing, perpendicular to the leeward coast of Oahu. Moorings were deployed for one complete lunar cycle at each of three locations, recording 10 echoes every 15 min. Light sensors measured the nocturnal light intensity at 30-s intervals. Statistical analysis revealed significant effects of both light and other lunar cycle cues. Overall, the effect size was very low considering the light transmission characteristics of the subtropical Pacific, making measurement from stationary acoustic platforms critical.

Contributed Papers

10:55

4aAO7. Classification of bioabsorption lines at low frequencies: The Van Holliday connection. Orest Diachok, Stephen Wales (Naval Res. Lab., Washington, DC 20375, OrestDia@aol.com), and Paul Smith (Southwest Fisheries Sci. Ctr., La Jolla, CA 92038-0271)

We will describe preliminary results of the second Bioacoustic Absorption Spectroscopy experiment in the Santa Barbara Channel (SBC), BAS II, which was designed to measure the effects of bioabsorptivity on transmission loss (TL) at frequencies between 0.25 and 6 kHz between an ultrabroadband source and a 16-element vertical receiving array. A fisheries echo sounder and trawls provided bioacoustic parameters. Colocated cores and chirp sonar provided geoacoustic parameters (Turgut, unpublished). Temporal changes in temperature structure were measured with three orthogonal thermistor strings. Highest losses were observed at night when fish were dispersed, and at the average depth of the absorption layer, 13 m, in accord with echo-sounder data and simulations. Absorption lines were evident at the resonance frequencies of 15-cm sardines (1.1 kHz) and 11-cm anchovies (1.8 kHz), in accord with trawls, and 20-cm jack mackerel (0.7 kHz), a species known to inhabit the SBC. This classification is consistent with Holliday’s (1972) backscattering measurements of the resonance frequency of jack mackerel (samples collected by Purse Seiner corroborated his classification). Frequencies of absorption lines changed in accord with depth changes at twilight. [Work supported by the ONR and the Southwest Fisheries Science Center.]

11:10


A number of Holliday’s seminal contributions to the acoustics of observing aquatic organisms are noted. Streaming pelagic fish have been sized from their echo resonance structure; fish swimming speeds have been measured through the Doppler effect; and snappers/groupers have been enumerated by high-resolution sidescan sonar. Zooplankton have been sized through multiple-frequency measurements of backscatter, and related to the underlying oceanography. Bowhead whales have been localized and tracked by means of a passive sonobuoy array. Certain themes have recurred, emphasizing the importance of both frequency diversity and spatial resolution in quantitative applications. Devices have been designed and built to solve specific problems, advancing the state of the art in these areas. For the student, the respective published work exemplifies the scientific method, introducing an important observational problem, describing its systematic approach, and interpreting results by means of simple but sound physical models. [Work supported by ONR.]
Session 4ABB

Biomedical Ultrasound/Bioreponse to Vibration: Ultrasonic Imaging and Therapy

Pierre Mourad, Chair
Applied Physics Laboratory, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105

Contributed Papers

8:00

4ABB1. 2D focal-field aberration dependence on time/phase screen position and correlation lengths. Sven Peter Näsholm (Medisinsk teknisk forskningssenter, Dept. of Circulation and Medical Imaging, N-7489 Trondheim, Norway, peter.nasholm@medisin.ntnu.no)

For high-frequency annular array transducers used in medical ultrasound imaging, aberrations due to tissue and body wall have a significant effect on energy transfer from the main lobe to the sidelobes of the acoustic field; that is, the aberrations make the total sidelobe level increase. This effect makes the ultrasound image poor when imaging heterogeneous organs. This study performs an analysis of the focal-field quality as a function of time/phase screen z position and time/phase screen correlation length. It establishes some rules of thumb which indicate when the focal-field sidelobe energy is at its highest. It also introduces a simple screened-scaling model which is useful as long as the screen position is not closer to the focus than a certain limit distance. The scaling model allows the real screen to be treated as a scaled screen at the position z = z_{screen} such that the 2D sound fields after 3D propagation from the annular arrays to the focal plane have been simulated using an angular spectrum method. The aberrators are represented by amplitude and phase/time screens.

8:15

4ABB2. Computational evidence for a discrete-scatterer aberration model in medical ultrasound. James C. Lacefield (Univ. of Western Ontario and Robarts Res. Inst., London, ON N6A 5B9, Canada, jlacefield@eng.uwo.ca)

Many techniques for correcting ultrasound focus distortion model the aberrating properties of tissue with a single time-shift screen, but simulations and phantom studies suggest single-screen models are ineffective for transmit focus compensation. Extension of the models to include multiple parallel screens is a logical increment in complexity, but the number of screens must be manageable and readily determined to yield practical aberration correction methods. To assess the feasibility of multi-screen strategies, simulations were performed to search for a general form for the aberration profile of breast tissue. Two-dimensional propagation of 3-MHz planar wavefronts through digitized breast specimens was computed using a k-space method [Tabei et al., J. Acoust. Soc. Am. 111, 53–63 (2002)] and waveforms were sampled at 1-mm intervals along the propagation direction. Arrival time, amplitude, and coherence fluctuations were correlated with scattering from distinct structures. This observation was most apparent when the first derivatives of those parameters with respect to the propagation direction were compared with the connective tissue architecture in the specimens. The assumption underlying time-shift screen models that aberration arises from smooth fluctuations in the acoustic properties of tissue merits reexamination. [Research supported by an NSERC Discovery Grant.]

8:30

4ABB3. Eigenfunction analysis of stochastic backscatter for aberration correction in medical ultrasound imaging. Trond Varslot, Eirik Mo, Harald Krogstad, and Bjørn Angelsen (Norwegian Univ. of Sci. and Technol., Trondheim, Norway)

A filter for aberration correction in medical ultrasound imaging is presented. The filter is optimal in the sense of maximizing the expected energy in a modified beamformer output of the received acoustic backscatter. The situation considered is frequently found in applications when imaging organs through a body wall: aberration is introduced in a layer close to the transducer, and acoustic backscatter from a scattering region behind the body wall is measured at the transducer surface. The scattering region consists of scatterers randomly distributed with very short correlation length compared to the acoustic wave length of the transmit pulse. The scatterer distribution is therefore assumed to be δ-correlated. Theoretical considerations imply that maximizing the expected energy in a modified beamformer output signal naturally leads to eigenfunctions of a Fredholm integral operator, where the associated kernel function is a spatial correlation function of the received stochastic signal. Aberration characterization and aberration correction have been studied for simulated data constructed to mimic aberration introduced by the abdominal wall. The results compare well with what is obtained using a diffraction limited time-reversal filter based on simulated point source data.

8:45

4ABB4. Iteration of ultrasound aberration correction methods. Svein-Erik Maasoey, Bjørn Angelsen (Dept. of Circulation and Imaging, Norwegian Univ. of Sci. and Technol., Olav Kyrres Gate 3, 7491 Trondheim, Norway, svein-erik.masoey@medisin.ntnu.no), and Trond Varslot (Norwegian Univ. of Sci. and Technol., 7491 Trondheim, Norway)

Aberration in ultrasound medical imaging is usually modeled by time-delay and amplitude variations concentrated on the transmitting/receiving array. This filter process is here denoted a TDA filter. The TDA filter is an approximation to the physical aberration process, which occurs over an extended part of the human body wall. Estimation of the TDA filter, and performing correction on transmit and receive, has proven difficult. It has yet to be shown that this method works adequately for severe aberration. Estimation of the TDA filter can be iterated by retransmitting a corrected signal and re-estimate until a convergence criterion is fulfilled (adaptive imaging). Two methods for estimating time-delay and amplitude variations in receive signals from random scatterers have been developed. One method correlates each element signal with a reference signal. The other method use eigenvalue decomposition of the receive cross-spectrum matrix, based upon a receive energy-maximizing criterion. Simulations of iterating aberration correction with a TDA filter have been investigated to study its convergence properties. A weak and strong human-body wall model generated aberration. Both emulated the human abdominal wall. Results after iteration improve aberration correction substantially, and both estimation methods converge, even for the case of strong aberration.
4aBB5. Enhanced detection of acousto-photonic imaging signals using a photorefractive crystal based system. Lei Sui, Todd Murray, Gopi Maguluri (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215, suilei@bu.edu), Alex Nieva, Florian Blonigen, Charles DiMarzio (Northeastern Univ., Boston, MA 02115), and Ronald A. Roy (Boston Univ., Boston, MA 02215)

Acousto-photonic imaging is a dual-wave sensing technique where diffuse coherent light wave interacts with a superimposed acoustic field. A phase-modulated light wave emanates from the interaction region and carries with it information about the local opto-mechanical properties of the insonated media. The coherent nature of the light produces a speckle field. The modulation of the speckle field is spatially incoherent, yielding a small modulation depth when collecting multiple speckles or extremely low light levels when detecting at the single speckle level. We report preliminary results from a new detection scheme where the scattered laser light is mixed with a reference beam in a photorefractive crystal. The crystal serves as a dynamic holographic medium and a photorefractive grating is formed from which the reference beam diffracts. The diffracted reference beam has the same spatial structure as the scattered light, and the two interfere at the photodetector where the phase modulation on the scattered beam is converted to an intensity modulation. Measurements of the signals are presented for gel phantoms seeded with suspended polystyrene beads. The results exhibit qualitative agreement with a simple theoretical model. [Work supported by the Center for Subsurface Sensing and Imaging Systems via NSF ERC Award No. EEC-9986821.]

9:15
4aBB6. Combination of B-mode imaging and acousto-photonic sensing using a commercial ultrasound scanner. Emmanuel Bossy, Lei Sui, Todd W. Murray, and Ronald A. Roy (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, ebossy@bu.edu)

The acousto-photonic imaging (API) of a turbid medium is based on the interaction of multiply scattered coherent laser light with an ultrasonic field. The two waves mix and the photons emanating from the interaction region are phase modulated at the ultrasound frequency. This technique yields information on the optical and acoustical properties of the medium, with the interaction region defined by the dimension of the ultrasonic beam. We investigated the feasibility of combining conventional B-mode ultrasound imaging and API in vitro, using a commercial medical scanner (Analogic AN2300). The AN2300 was used to both generate the B-mode images and excite API signals. Gel-based acousto-optic phantoms were fabricated; these contained imbedded targets possessing acoustical and/or optical contrast. Analogous to power Doppler measurements, B-mode images were first acquired and then used to select regions of interest within which API signals were generated. API information was then color-coded and superimposed on top of the frozen B-mode image. Preliminary results show that API signals can be excited using a commercial scanner, and serve to augment conventional B-mode images with information related to the opto-acoustic properties of the medium. [Work supported by the Center for Subsurface Sensing and Imaging Systems via NSF ERC Award Number EEC-9986821.]

9:30
4aBB7. Measurements of the stability of the effective apodization for the nonlinearly generated second harmonic as a function of propagation distance. Russell Fedewa, Kirk Wallar, Mark Holland (Washington Univ. in St. Louis, St. Louis, MO 63130), James Jago, Gary Ng, Matthew Rielly, Brent Robinson (Philips Medical Systems, Bothell, WA), and James Miller (Washington Univ. in St. Louis, St. Louis, MO)

The concept of an effective apodization was introduced to approximate the nonlinearly generated second harmonic field pattern based solely on the linear propagation of a field transmitted at the frequency of the harmonic. We have previously demonstrated that transmitting with an effective apodization determined from measurements made only in the focal plane yields an accurate description of the nonlinearly generated field over a very wide range of depths. The goal of this work was to determine the stability of the effective apodizations obtained from measurements of the ultrasonic field in a series of planes before and beyond the focal zone. Transverse 2D scans of the transmitted fields were performed with a 0.6-mm-diam. hydrophone for both vascular and cardiac arrays. Linear angular spectrum backpropagation of the measured fields determined the effective apodizations. These measurements were compared with simulations based on a nonlinear Burgers equation enhanced angular spectrum approach. The resulting effective apodizations of the second harmonic field were remarkably constant (differing by typically a few percent and never by more than 9.4% from the mean) for ten axial positions ranging from 0.1 to 1.6 times the focal distance. [Work supported by NIH HL072761 and Philips Medical Systems.]

9:45
4aBB8. Dual transmission model of the fetal heart tone. Donald A. Baker (Ste. 230, 3003 3rd Ave. NE, Camas, WA 98607) and Allan J. Zuckerwar (NASA Langley Res. Ctr., Hampton, VA 23681)

Detection of the fetal heart tone by auscultation is sometimes easy, other times very difficult. In the model proposed here, the level of difficulty depends upon the position of the fetus within the maternal abdomen. If the fetus lies in the classical left/right occiput anterior position (head down, back against the maternal abdominal wall), detection by a sensor or stethoscope on the maternal abdominal surface is easy. In this mode, named here the “direct contact” mode, the heartbeat pushes the fetus against the detecting sensor. The motion generates pressure by impact and does not involve acoustic propagation at all. If the fetus lies in a persistent occiput posterior position (spine-to-spine, fetus facing forward), detection is difficult. In this, the “fluid propagation” mode, sound generated by the fetal heart and propagating across the amniotic fluid produces extremely weak signals at the maternal surface, typically 30 dB lower than those of the direct contact mode. This reduction in tone level can be compensated by judicious selection of detection frequency band and by exploiting the difference between the background noise levels of the two modes. Experimental clinical results, demonstrating the tones associated with the two respective modes, will be presented.

10:00–10:15 Break

10:15
4aBB9. Flow measurements based on speckle decorrelation: Simulation and experiment. Oliver D. Kriptgans, Juan Zhu, Jonathan M. Rubin, J. Brian Fowlkes (Dept. of Radiol., Univ. of Michigan Health Systems, Ann Arbor, MI 48109-0553), and Anne L. Hall (GE Med. Systems, Milwaukee, WI)

Traditional Doppler-based flow measurements suffer from bad signal to noise for large angles between the wavevector and the flow direction. To overcome this limitation, speckle decorrelation might be used to measure lateral flow. Experiments were performed on a flow phantom (tube diameter 6.35 mm, flow 1.6 mL/min) with the tube axis positioned in the imaging plane of a GE Logiq 9 scanner. IQ data sets of ten frames with 16 firings per scanline were recorded and speckle decorrelation used to estimate flow speeds throughout the image. The decorrelation computations were performed over different kernel types and compared to simulations performed using Field II by J. Jensen with acoustical transmit as well as beamforming parameters set to match experiments. Speckle decorrelation rates in experiments scale correctly for the parabolic flow profile inside the tube. Simulations reproduced a similar but smoother profile. Flow velocities could be estimated using a scaling factor based on the spatial correlation of the beam. The combination of velocity estimates from Doppler and speckle decorrelation may provide a more uniform display of flow and lead to less angle dependence. [Research supported by U.S. Army Grant No. DAMD17-00-1-0344 and GE Medical Systems.]


10:30
4aBB10. Validation of Doppler ultrasound measurements using particle, image velocimetry in a flow phantom. John Cosgrove (School of Phys., Univ. of Edinburgh, Edinburgh EH9 3JZ, UK), Siobhan Meagher, Peter Hoskins, Clive Greathead (The Univ. of Edinburgh, UK), and Richard Black (The Univ. of Liverpool, UK)

Cardiovascular disease is responsible for over 50% of all deaths in the world and there is a substantial amount of evidence which suggests that abnormal vessel wall shear stress is correlated with the development of atherosclerosis. Wall shear stress is calculated from wall shear rates, the measurement of which is a technically challenging problem for ultrasound. In this study a flow phantom consisting of a meshed-gear pump and corresponding control electronics is used to generate a range of flow waveforms in a straight tube. These flows are measured using Doppler ultrasound and compared to corresponding particle image velocimetry (PIV) measurements and to analytical solutions of the flow equations for a range of Wormsley parameters. Although previous studies have been undertaken calibrating Doppler ultrasound in straight tubes, they have not used PIV. This study serves as a prelude to investigations using PIV to assess the accuracy of Doppler ultrasound in phantoms with anatomically realistic geometries for which there are no analytical solutions to the flow.

[Research funded by the Engineering and Physical Sciences Research Council UK.]

10:45
4aBB11. In vitro microscopic imaging of rt-PA thrombolysis with 120-KHz ultrasound in a human clot model. Jason Y. Cheng (Dept. of Emergency Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267-0769, Jason.Cheng@uc.edu), Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH 45267-0586), and George J. Shaw (Univ. of Cincinnati, Cincinnati, OH 45267-0769)

Substantial enhancement of recombinant tissue plasminogen activator (rt-PA) thrombolysis can be achieved with diagnostic frequency ultrasound. Microscopic visualization of human whole-blood clots treated with human fresh frozen plasma (HFFP), rt-PA, and 120-KHz pulsed ultrasound offers a useful method to study the possible mechanisms. Whole human-blood clots were formed inside of glass micropipette (1.7-mm diameter) and placed in a water tank at 37 °C. The clot–plasma interface was imaged using an inverted optical microscope. Clots were exposed to HFFP (control), HFFP and rt-PA (0.0945 mg/ml), (sham), or HFFP, rt-PA (0.0945 mg/ml), and 120-KHz pulsed ultrasound (ultrasound treated) for 30 min. Thrombolysis at the clot surface was recorded with a CCD camera and the lytic front was analyzed as a function of time. Images of the clots were analyzed to quantify the overall amount of lysis and compared to a dynamic chemical model of enzymatic degradation in the presence of ultrasound-induced flow. Sham and ultrasound-treated clots showed significant lysis with the lytic rate of (1.53±0.02 microns/min; mean ± s.d.) and (5.17±0.10 microns/min), respectively. Ultrasound enhanced thrombolysis by a factor of 4. In addition, the lytic rate profile may offer a suggestion to the mechanism of enhancement. No thrombolysis was observed in control clots. [The authors of this study gratefully acknowledge the support of the Whitaker Foundation Grant RG-0128-01.]

11:00

A high-power prototype dedicated to trans-skull therapy has been tested in vivo on 20 sheep. The array is made of 200 high-power transducers working at 1-MHz central and is able to reach 260 bars at focus in water. An echographic array connected to a Philips HDI 1000 system has been inserted in the therapeutic array in order to perform real-time monitoring of the treatment. A complete craniotherapy has been performed on half of the treated animal models in order to get a reference model. On the other animals, a minimally invasive surgery has been performed thanks to a time-reversal experiment: a hydrophone was inserted at the target inside the brain thanks to a 1-mm craniotherapy. A time-reversal experiment was then conducted through the skull bone with the therapeutic array to treat the targeted point. For all the animals a specified region around the target was treated thanks to electronic beam steering. Animals were finally divided into three groups and sacrificed, respectively, 0, 1, and 2 weeks after treatment. Finally, histological examination confirmed tissue damage. These in vivo experiments highlight the strong potential of high-power time-reversal technology.

11:15

Ultrasonic imaging systems assume a constant acoustic velocity in human tissues in the beamforming process. However, the case of brain imaging, strong skull aberrations induce a displacement of the focal stain, a spreading in the main lobe, and an increase in the side lobes level. This limits considerably ultrasonic brain imaging applications. It has been shown that a very accurate focusing could be achieved through a human skull by using an inverse filter technique. This method, based on a set of acoustic sensors located inside the brain, demonstrated invasively that it was possible to achieve high-resolution brain imaging. A noninvasive ultrasonic method is presented. It is based on two arrays located on each side of the head. Each array focuses on the other one to deduce the aberrations induced by the skull bone. Once the effect of the skull bone in front of each array is extracted, the wavefronts are corrected and emitted. This noninvasive technique restores the position of the focal stain and lowers the secondary lobe level up to 15 dB compared to a cylindrical focusing.

11:30
4aBB14. Ultrasound produced homogenization and liquefaction of brain tissue in vivo at pressures beyond the cavitation threshold. Natalia Vykhostseva, Nathan McDannold, and Kullervo Hynynen (Harvard Med. School/Brigham and Women’s Hospital, 221 Longwood Ave., Rm. 013, Boston, MA 02115)

Histological effects of short-pulsed focused ultrasound on tissue pressure amplitudes above the cavitation threshold were studied in the brain. Sixty locations in the brains of 13 rabbits were sonicated at a depth of 4–10 mm (1.5 MHz, burst lengths, 1–100 ms, PRF 1–100 Hz, acoustic power, up to 950 W). MRI was used to aim the beam, monitor the temperature, and evaluate the effects. It was found that high-intensity short-pulsed ultrasound could effectively destroy targeted tissues. The temperature elevation measured during the sonication was below the threshold for tissue damage. The tissue effects were different from that which has been described for the thermal or cavitation mechanisms. The lesions appeared as homogeneous regions with no distinguishable cell debris. The central core of the lesion region consisted of a puddling-like mass, arranged in a helically coiled, layered fashion. In some cases, the core seemed to be completely liquefied. These findings appeared to be associated with nonlinear ultrasound absorption and possibly cavitation-related effects. The spiral shape observed in some cases may indicate the existence of an axial radiation torque in the focal zone. The lesions’ appearance may point to new interactions between ultrasound and tissue, which could be exploited for new therapies.

11:45
4aBB15. Transcranial MRI-guided FUS-induced BBB opening in the rat brain. Lisa H. Treat, Nathan J. McDannold, and Kullervo Hynynen (Dept. of Radiol., Brigham and Women’s Hospital/Brigham Med. School, 221 Longwood Ave., LMRC Rm. 013, Boston, MA 02115, treat@mit.edu)

The blood-brain barrier (BBB) has been a major limitation in treating diseases of the brain because therapeutic agents are either unable to penetrate or have dose-limiting side effects in diffuse opening of the BBB. A previous study demonstrated that focused ultrasound (FUS) can locally
open the BBB in a rabbit model when a piece of skull is removed and that magnetic resonance imaging (MRI) can be used to guide and monitor the procedure. This study examined whether the same desired effect of local BBB disruption can be achieved by applying FUS through an intact skull in a rat model. Twenty-eight Sprague-Dawley rats were anesthetized, shaved, and sonicated at four focal locations in the brain, using a 1.5-MHz focused transducer. Contrast-enhanced MR images were obtained before and after sonication. The images indicated contrast agent penetration at the focal coordinates following Optison-enhanced sonication. This study demonstrated that the distortion of the ultrasound beam by the rat skull was not significant enough to inhibit focal BBB opening. Subsequent experiments using MRI-guided FUS to aid in targeted drug delivery to brain tumors in a rodent model could thus be performed more efficiently without cranial surgery. [Research funded by NIH Grant No. CA76550.]

THURSDAY MORNING, 27 MAY 2004

Liberty 5, 8:25 To 11:45 a.m.

Session 4aEA

Engineering Acoustics: Transducers, Arrays and Calibration

Kirk Jenne, Cochair

NAVSEA Division Newport, 1176 Howell Street, Newport, Rhode Island 02841-1708

Kenneth Walsh, Cochair

K M Engineering Ltd., 51 Bayberry Lane, Middletown, Rhode Island 02842

Chair’s Introduction—8:25

Contributed Papers

8:30

4aEA1. A new microsystem for near whispering. Sungjoon Choi, Wonkyu Moon (Dept. of Mech. Eng., POSTECH, San 31, Hyoja-Dong, Nam-gu, Pohang, Kyungbuk 790-784, South Korea), and Jeong Hyun Lee (iCurie Lab., Daechi-dong, Kangnam-gu, Seoul 135-846, South Korea)

A new microphone system was developed to monitor the human voice near the microphone in a noisy environment. The system is equipped with two special functions in addition to the usual microphone functions: reduction of air blow effects by the mouth and focused reception to a sound source. A wind filter was developed to reduce the air blow effects from the mouth during speaking. This filter is a plate perforated by an array of small holes; the method used to design the filter is also presented. To achieve focused reception, four microphones were used in conjunction with a new signal processing method. The proposed signal processing method effectively increases the directivity in the desired direction. Additionally, it provides the system with focusing on the source since the source is located adjacent to the system. A prototype of the proposed system was fabricated and subjected to performance tests. The results showed that air blow effects can be reduced by up to 20 dB and the directional gain is more than 4 dB. The proposed microphone system shows such good performance that it can be used in mobile phones for whispering communication.

8:45

4aEA2. On the potential performance of micro-fabricated subminiature microphones based on electron surface tunneling. Michael Pedersen (CNRI, 1895 Preston White Dr., Ste. 100, Reston, VA 20191)

The principle of electron tunneling across a vacuum potential energy barrier, as utilized in surface tunneling microscopy (STM), is a well known, highly sensitive, method for the detection of surface features and displacements in the subnanometer regime. In this paper, the basic properties of the surface tunneling principle and its possible utilization in acoustic sensors are discussed, with particular focus on the expected noise properties of a microphone based on tunneling in comparison to more conventional devices based on capacitive detection. It is found that the noise performance is largely limited by thermal noise due to the acoustical radiation impedance of the device in combination with electronic noise in the circuit used to establish and control the tunneling current. Finally, possible methods for sensor fabrication are discussed with special attention aimed at resolving the common problem of vibration sensitivity associated with STM. This problem stems from the relative large inertial mass and low flexural rigidity of the tunneling tip suspension system in most detector designs. Alternative approaches suitable for acoustic sensors in which the tunneling tip is fixed are described and it is shown that low vibration sensitivities comparable to capacitive devices can be achieved.

9:00

4aEA3. Numerical evaluation of the omnidirectional behavior of regular polyhedron loudspeakers. Heather Smith and Timothy Leishman (Dept. of Phys., Brigham Young Univ., N283 ESC, Provo, UT 84602, hm73@email.byu.edu)

A regular polyhedron loudspeaker (RPL) consists of a rigid enclosure in the shape of a regular polyhedron with loudspeaker drivers mounted within each face. Room acoustics measurements often incorporate dodcachecolate loudspeakers presumably as omnidirectional sources of sound. This research is intended to support experimental work to determine which of the five regular polyhedron loudspeaker configurations actually produces the most omnidirectional field over a wide frequency range. Loudspeaker models based on the boundary element method were created to assess frequency-dependent radiation patterns. These models assume that the loudspeaker drivers radiate as plane circular pistons. Far-field sound pressures were generated for several field points around the models (5-deg polar and azimuthal angle increments). Results are presented as directivity balloon plots and compared with experimental measurements. Other numerical results are shown as variations on the experimental setup. These involve dimensional scaling of an entire loudspeaker and dimensional scaling of driver diameters while keeping enclosure dimensions constant. They also involve generation of data at more closely spaced field points (1-deg polar and azimuthal angle increments). Results from a spherical enclosure with radiating pistons centered on the angular coordinates of dodecahedron facial centers are also compared to those produced by a dodecahedron loudspeaker model.
9:15
4aEA4. Optical mapping of the acoustic output of a focused transducer. Robert D. Huber, Diane J. Chinn, and David H. Chambers (Lawrence Livermore Natl. Lab., P.O. Box 808 L-333, Livermore, CA 94551)

A Michelson interferometer is used to map the ultrasonic displacement of the lens at the end of a delay rod of a 50-MHz immersion transducer. The purpose of mapping the displacement is to provide a source function to a model that predicts the ultrasonic propagation in, and interaction with, various materials. The output of the Michelson interferometer can be calibrated, and then used to determine the displacement of the transducer lens surface moving at ultrasonic frequencies. Using the interferometer, the displacement of the transducer lens is measured at discrete points along its surface. This displacement map then provides the ultrasound propagation model with the actual source function. Direct comparison between a model with a simulated source function and experimentally obtained data is presented. [Work performed under auspices of the U.S. Department of Energy by the Lawrence Livermore National Laboratory under Contract No. W-7405-ENG-48.]

9:30
4aEA5. Effects of coupled vibrations in cylindrical shell transducers. Boris Aronov, Sundar Regmi, and David A. Brown (BTech Acoust. and Univ. of Massachusetts Dartmouth, Acoust. Res. Lab., ATMC, 151 Martine St., Fall River, MA 02723, DBrown122@cox.net)

Coupled vibrations in cylindrical transducers can have unwanted effects on the resulting radiation or reception of sound. An investigation of vibrations of piezoelectric cylindrical shell transducers using analytical and experimental methods shows the presence of an additional flexural resonant mode that was not previously reported [see J. F. Haskins and J. L. Walsh, “Vibrations of ferroelectric cylindrical shells with transverse isotropy,” J. Acoust. Soc. Am. 29, 6 (1957)]. The additional flexural mode can be strongly coupled to radial and axial modes in cylindrical shells with finite thickness. The effect can be substantial, resulting in unwanted resonances that may be in the frequency band of interest for both projectors and receivers. An equivalent circuit is derived, which describes the cylindrical transducer operation taking into account the effect of coupled vibrations. The results of calculations in terms of resonant frequencies and frequency responses for both transmit and receive modes are in good agreement with experiments. The overall performance of the cylindrical shell transducer can be greatly enhanced by using cylinders of appropriate height to diameter aspect ratio. [Work supported in part by ONR 321SS.]

9:45

Development of local volume displacement sensors is presented. This development supports the implementation of noise control techniques that are based on minimization of local volume displacements, velocities, or accelerations of a vibrating structure. In this paper, a general methodology for the development of local volume displacement sensors for vibrating plates using Poly(vinylidene fluoride) (PVDF) is presented. This methodology was verified experimentally for a clamped plate. The local volume displacement measured using a PVDF sensor matched the local volume displacement found using multiple accelerometer measurements. The resulting sensor spans the entire length of the plate and covers the width of the area of interest. The sensor is composed of a number of strips whose width varies quadratically on the local area and linearly elsewhere. Design issues for a clamped plate, a simply supported plate, and a plate with arbitrary boundary conditions are discussed along with a presentation of some sample sensor shapes.

10:00–10:15 Break

10:15

The development of volume displacement sensors for vibrating beams is revisited with emphasis on numerical approach. This development supports the implementation of noise control techniques that are based on minimization of volume displacements, velocities, or accelerations of a vibrating structure. This paper first reviews some of the existing general methodologies for the development of volume displacement sensors for vibrating beams using Poly(vinylidene fluoride) (PVDF). The presentation includes the quadratic and modal development of volume displacement sensors for vibrating beams. These techniques are extended to numerical analysis by discretizing the beam and assuming constant sensor shape on each beam element. The result is a system of linear equations in which the assumed constant shapes are the unknowns. The size of the system of equations, which determines the accuracy of the sensor, is directly related to the highest frequency in the signal to be processed. The resulting sensors are numerically and experimentally verified for a simply supported beam. The results show for low-frequency application, relatively simpler sensor shapes can be utilized, whereas for higher frequencies the sensor shape converges to quadratic function. Finally, sensor accuracy and the range of the application frequency are discussed and some sample shapes presented.

10:30

The mutual resistance of transducer arrays is investigated in order to design arrays with improved performance for high intensity sounds at a given frequency. This work proposes the theory that the mutual resistance is related to the loading effects of pressure waves propagated from a piston driver on the surface of another driver. Using this interpretation, the important characteristics of the mutual resistance of two piston drivers are explained and the conditions for local maxima in the mutual resistance are easily determined. On the basis of analyses of the interactions between a driver and acoustic pressure waves, we propose a method to determine the driver radius and the distance between two drivers that give maximum mutual radiation resistance. To evaluate the proposed method, the total resistance of a transducer array is calculated using the formulas for mutual and self-resistance established by Pritchard. The results of the calculations of the total resistances of arrays with many drivers show that a transducer array with drivers arranged sparsely can achieve a larger value of the radiation power per unit area as well as better radiation efficiency than an array in which the drivers are in a closely packed arrangement at a given frequency.

10:45

We present our recent experimental results of the mutual radiation impedance of piezoelectric cylindrical shell transducers in arrays using the Z- and V-methods previously reported by the authors [J. Acoust. Soc. Am. 112, 2407 (2002)]. The mutual radiation impedance in an array of two coaxially aligned cylindrical transducers was determined as a function of separation distance. The results obtained by both methods are in good agreement with the calculations based on the analytical equations for a similar array developed by D. H. Robey [J. Acoust. Soc. Am. 27, 706–710 (1955)]. The mutual radiation impedances in other array geometries in-
Parametric arrays produce audible sound from the audio frequency modulation of high-intensity ultrasound as a result of nonlinear propagation. This paper presents observations of the audible sound resulting from modulated radiation pressure. Audio signals due to the parametric array effect depend strongly on the square of the modulation frequency. It is often shown that the low-frequency region shows a flat frequency response, although it has not been clarified whether this is related to the system noise floor, modulated radiation pressure, or microphone nonlinearity. In this investigation, microphone nonlinearity was found to be very strong at short distances unless acoustic filtering was used. A novel low-pass acoustic filter (passes less than 10 kHz) was constructed using four layers of polyethylene spaced at 4 mm. Using this scheme, the ultrasonic carrier was attenuated and microphone nonlinearity was reduced by six orders of magnitude. Modulated radiation pressure was successfully observed at distances of 1 to 3 m. The observed audio signal showed a flat frequency response and agreed with theoretical predictions for modulated radiation pressure. The reproduced audio signal from radiation pressure appears in a lower frequency region of parametric array audio systems, and the effect should be taken into the system design.

11:15

A hand-held, battery-powered pistonphone was designed to calibrate infrasonic microphones in the field. With a battery voltage of 12 V, a prototype pistonphone provides a sound pressure level of 110±0.5 dB re 20 μPa at a frequency of 13.8 Hz. The microphone is inserted into an acoustic coupler, in which a stretched diaphragm is excited to generate infrasound. The diaphragm motion is generated by an eccentric ball bearing, such that the outer race does not rotate but remains in contact with the diaphragm at the same point during a complete rotation cycle. Thus this mechanism avoids sliding friction and insures a positive diaphragm displacement, independent of acoustic load. The diaphragm tension is adjusted to a sufficient high level to prevent “floating” (separation from the eccentric) at the operating frequency. Further, the pistonphone is designed to have a high “thermal wave number” to insure adiabatic as opposed to isothermal wave propagation.

11:30
4aEA12. Method and system for calibrating acoustic receivers in borehole logging tools. Fernando Garcia Osuna and Toru Ikegami (Schlumberger, 110 Schlumberger Dr., Sugar Land, TX 77478, fgarcia@sugar-land.oilfield.slb.com)

Monopole and quadrupole contamination on dipole measurements is one of the main problems for sonic logging tools that employ arrays of acoustic receivers to measure a borehole’s dipole mode. The amplitude and phase mismatch of the array of sensors in acoustic logging tools plays an important role in the quality of dipole measurements. Acoustic receivers often have different sensitivities, and different sensitivities to the same wave field result in a greater possibility of no dipole contamination. Even similarly or identically manufactured receivers tend to report different amplitudes and time measurements (i.e., amplitude and phase mismatch). In practice, the no-dipole modes are removed by making the measurements at different azimuth and extracting the dipole mode from those that reject the other contaminating modes. Therefore, to improve rock formation slowness estimation and downhole modal computation, it is necessary to calibrate acoustic logging tools by detecting and correcting the amplitude and phase mismatch of the individual sensors mounted in the logging tools. A method and system for calibrating acoustic receiver arrays mounted in a downhole logging tool and the factors that affect their sensitivity are discussed in this paper.

THURSDAY MORNING, 27 MAY 2004 VERSAILLES BALLROOM, 10:00 A.M. TO 12:00 NOON

Session 4aED

Education in Acoustics: Hands on Demonstrations for High School Students

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

Chair’s Introduction—10:00

Approximately 20 acoustics demonstrations will be distributed in the room. All demonstrations will be available for high school students’ hands-on experimentation. Participation by other conference attenders is welcome as long as their activity does not interfere with student learning.
Musical Acoustics: General Topics in Musical Acoustics

Roger J. Hanson, Chair

Physics Department, University of Northern Iowa, Cedar Falls, Iowa 50614

Contributed Papers

8:30

4aMU1. Radiation control applied to sound synthesis: An attempt for “spatial additive synthesis.” Olivier Warusfel, Nicolas Misdariis, Terence Caulkins, and Etienne Corteel (IRCAM, I place Stravinsky, 75004 Paris, France)

Sound synthesis is generally focused on the reproduction of the spectral characteristics of the source or on the simulation of its physical behavior. Less attention is paid to the sound playback step which generally results in a simple diffusion on a conventional loudspeaker setup. Stating the perceptual importance of a faithful reproduction of the source radiation properties, the paper presents a method combining a synthesis engine, based on physical modeling, with a rendering system allowing an accurate control on the produced sound-field. Two sound-field synthesis models are considered. In the first one, a local 3D array of transducers is controlled by signal processing for creating elementary directivity patterns that can be further combined in order to shape a more complex radiation. The dual approach consists in surrounding the audience with transducer arrays driven by wave field synthesis in order to simulate the sound field associated to these elementary directivity patterns. In both cases, the different radiating modes of a given instrument are synthesized separately, in conjunction with their associated radiation pattern, and then superimposed in the spatial domain, i.e., during the propagation in air. This approach, referred as “spatial additive synthesis,” is illustrated, taking the example of different musical instruments.

8:45

4aMU2. Digital filters for accurate simulation of wave propagation losses in tubes. Reiner Wilhelms-Tricarico and Richard McGowan (CRESS LLC, 1 Seaborn Pl., Lexington, MA 02420, reiner@speech.mit.edu)

A boundary layer approximation for viscous damping in one-dimensional wave transmission in a tube results in an irrational frequency-dependent damping filter for sound propagation. This filter can be approximated with high accuracy by a rational filter function that can be obtained from Padé approximations or continued fraction expansion. Taking into account the viscous losses in a Kelly–Lochbaum structure that represents sound propagation in a tube with spatially varying cross section results in replacing the delay elements of the lattice filter for the loss-free case by special recursive filters. The design, implementation, and applications of the filter structures will be presented.

9:00

4aMU3. In-room sound reproduction using active control: Simulations in the frequency domain and comparison with wave field synthesis. Philippe-Aubert Gauthier, Alain Berry (GAUS, Mech. Eng. Dept., Université Sherbrooke, 2500 boul. Université, Sherbrooke, QC J1K 2R1, Canada, philippe_aubert_gauthier@hotmail.com), and Wieslaw Woszczuk (McGill Univ., Montréal, QC H3A 1E3, Canada)

Active sound control simulations were performed for progressive sound field reproduction over a “large” area using multiple monopole loudspeakers. The model is limited to the simulation of the acoustical output of the prescribed loudspeaker array in a simple room, and is based on achieving an optimal control in the frequency domain. This rather simple approach is chosen for this first feasibility study concerning a limited number of possible configurations of sensing microphones and loudspeakers. Other issues of interest concern the comparison with wave field synthesis, the control mechanisms and transducer configurations. As it is demonstrated, in-room reproduction of sound field using active control can be achieved with a residual normalized squared error below 2% while open-loop wave field synthesis gives more than 100% of error in the same situation. Usage of active control technique suggests the possibility to automatically overcome the room’s natural dynamics. A special surrounding configuration of sensors is introduced for a sensor-free listening area. [Work supported by NSERC, NATEQ, VRQ, and Université de Sherbrooke.]

9:15

4aMU4. Philosophical and cultural perspectives on acoustics in Vedic Hinduism. M. G. Prasad (Dept. of Mech. Eng., Stevens Inst. of Technol., Hoboken, NJ 07030, mprasad@stevens.edu)

Acoustics plays a very important multi-faceted role in Vedic Hinduism. Vedas, that is an infinitely large collection of chants (mantras) in ancient Sanskrit language, form the foundational literature of Vedic Hinduism. The Vedic chants have specific acoustical qualities and intonations. The Vedic literature describes the various aspects of acoustics, namely, philosophical, spiritual, and cultural. The use of sounds from conch-shell, bells, cymbal in addition to the Vedic chants in rituals shows the spiritual aspects. Vedic literature discusses the role of sound in the philosophical understanding of our world. Music, both vocal and instrumental, plays an important role in the cultural aspects of Vedic Hinduism. It can be seen that certain musical instruments such as “mrdangam,” a percussion drum, reflect scientific principles underlying in their design. This paper presents an overview of the various important and interesting roles of acoustics in Vedic Hinduism.

9:30

4aMU5. Measurement of effects on tone with lip-protecting music splints for wind instrument players. Chigusa Katada, Kazunori Nozaki, Mihiaru Imai (Div. of Community Dentistry and Informatics, Molecular Oral Biol. and Dentistry, Osaka Univ., 1-8 Yamadaoka, Suita, Osaka 565-0871, Japan, c_katada@dent.osaka-u.ac.jp), Masayuki Kawamoto, Yuko Shima, Hiroo Tamagawa, Yoshinobu Maeda (Osaka Univ. Dental Hospital, Suita, Osaka 565-0871, Japan), Naoki Ohboshi (Kyoto Univ. Medical Hospital, Sakyo-ku, Kyoto 606-8501, Japan), and Tadao Toda (Osaka College of Music, Osaka 561-8555, Japan)

To protect against lip trauma from wind instruments, music splints that cover the sharp edges of incisor teeth are often manufactured by dentists. Wind instrument players who have installed these custom-made music splints often express not only their lip comfort but also changes in their tone quality. In this study, we investigated the effect of the splints on the tone quality. We recorded three types of trumpet sounds such as long tones, arpeggios with perfect fifth, and tonguing tones with and without using a splint, respectively, by a professional trumpet player in an anechoic room. After fast Fourier transform, the higher harmonics was observed more in the splint group than in the nonsplint group, with sharp peaks from 5000 to 8000 Hz. We also examined the differences of these sound groups with recognition tests by two groups of listeners such as
professional musicians and nonprofessional persons. Though sound-pressure levels of higher harmonics in two sound groups were lower than those at 400 to 2000 Hz, the musically trained persons recognized the difference perfectly. These results suggest the target of measurement to evaluate the effect of music splints.

9:45
4aMU6. The Sagrada Familia Cathedral where Gaudi envisaged his bell music. Shigeru Yoshikawa and Takafumi Narita (Dept. of Acoust. Design, Grad. School of Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, shig@design.kyushu-u.ac.jp)

The Sagrada Familia Cathedral in Barcelona, Spain was constructed in 1882. According to Antoni Gaudi, who worked over its grand plan, the Cathedral was supposed to be a huge musical instrument as a whole in the event of completion. As as result, the music of bells was expected to echo through the air of Barcelona from the belfries. However, Gaudi’s true intention cannot be exactly known because the materials prepared by him were destroyed by war fire. If his idea of the Sagrada Familia as an architectural music instrument is true, an acoustical balance should be considered between the roles of the Cathedral: bell music from the belfries and quiet service in the chapel. Basic structure of the Sagrada Familia seems to be an ensemble of twin towers. Following such speculation, we made a simplified acrylic 1/25-scale model of the lower structure of a twin tower located at the left side of the Birth Gate. The higher structure of this twin tower corresponds to the pinnacle where the bells should be arranged. The lower structure (about 45 m in actual height) has five passages connecting two towers. One of two towers includes five or six tandem columns whose ends are both squeezed to about 1.5 m in diameter. These columns seem to function as a kind of muffler. The location and shape of the roof over the nave is indefinite and tentatively supposed at the top of the lower structure. Based on our scale model, acoustical characteristics of the lower twin-tower structure as a muffler and acoustical differences between the exterior field and nave field will be reported and discussed.

10:00
4aMU7. Synthesis of audio spectra using a novel Bessel expansion. V. Vijayakumar and C. Eswaran (Multimedia Univ., FOSEE, Melaka 75450, Malaysia)

It is shown in this paper that the intensity variations of an audio signal in the frequency domain can be expressed by using a novel mathematical function containing a series of weighted complex Bessel functions. By proper choice of values for two parameters, this function can transform an input spectrum of discrete frequencies of unit intensity into the known spectra of different musical instruments. Two specific examples of musical instruments are considered for evaluating the performance of this method. It is found that this function yields musical spectra with a good degree of accuracy. The proposed method is compared with known synthesis techniques such as FM, AFM, and DFM [J. M. Chowning, J. Audio Eng. Soc. 21, 526–534 (1973); Palamin et al., ibid. 36, 671 (1988); Tan, ibid. 42 (11), 918–926 (1994)]. A brief discussion on the physical basis for the derivation of the proposed function is also presented.

THURSDAY MORNING, 27 MAY 2004
ROYAL BALLROOM A, 8:35 A.M. TO 12:00 NOON

Session 4aNs

Noise and jointly sponsored by all ASA Technical Committees: On the Occasion of His 90th Birthday: Special Session to Honor the Contributions of Leo L. Beranek to Acoustics and Teaching

Tony F. W. Embleton, Chair

8:40
4aNsi. Leo Beranek’s contributions to architectural acoustics from development of criteria for design to practical applications in buildings of all types. William J. Cavanaugh (Cavanaugh Tocii Assoc., Inc., 327F Boston Post Rd., Sudbury, MA 01776, wcavanaugh@cavtoci.com)

This paper traces Leo Beranek’s continuing contributions in architectural acoustics since the author joined the consulting staff of Bolt Beranek and Newman Inc. in February 1954. They are legion, including room acoustics, sound and vibration transmission, control of ambient noise from building mechanical electrical and plumbing systems, electroacoustic applications and other issues through the evaluation of the finished building spaces themselves. While Beranek’s classic texts, Acoustical Measurements (1948) and Acoustics (1954) served as essential references for the early BBN staff, criteria in architectural acoustics were severely limited if not nonexistent. If he did not take up the challenge to fill a criterion gap, his inspirational leadership guided others to do so, as it did the author and colleagues trying to better understand speech privacy in buildings. Beranek’s real labor of love in architectural acoustics is in understanding music performance halls. Beranek’s dedicated research over a span of five decades has produced countless journal papers in concert hall acoustics as well as three important texts, the most recent in 2003. Leo Beranek’s contributions have earned him the Sabine and Gold Medals of the Acoustical Society and countless other honors including most recently the National Medal of Science.
4aNS2. Recent concert halls and opera house in Japan. Takayuki Hidaka (Takenaka R&D, 1-5-1 Otsuka Inzai, Chiba 270-1395, Japan, hidaka.takayuki@takenaka.co.jp)

Since we invited Dr. Beranek to Japan for the first time in 1989, we had been working together with him for a period of 13 years, until 2001, on seven hall projects as acoustic design consultants. All of these halls are of premium importance to Japan. Dr. Beranek always came up with innovative concepts and helped create halls endowed with high acoustic originality. These halls are now loved by music-related people and opera fans and regarded as the pride of Japan. The reviews and studies achieved through these projects were published as seven J. Acoust. Soc. Am. papers to disclose the outcome in an objective way to the public. A brief outline of the history of our collaboration and its background are presented.

4aNS3. BBN and structural acoustics. Richard H. Lyon (RH Lyon Corp, 691 Concord Ave., Cambridge, MA 02138)

Leo Beranek, Dick Bolt, and Bob Newman founded a company based on technical excellence and client service—BBN. The early services were oriented to noise control and architectural acoustics, but these led fairly quickly over about a decade into several related fields. One such field, now called “structural acoustics,” arose from activities in noise control where the radiated sound due to vibrations of the machine caused problems. Commercial work on such problems was later augmented by work for the US Navy, the Air Force, and NASA. In the mid- and late-1950s Ira Dyer built the group that during the course of about the next decade developed the field of structural acoustics, with emphasis on statistical modeling and with applications to ships, aircraft, and space launch vehicles. The author will present some of his personal remembrances of this second decade, with particular emphasis on the development of statistical energy analysis.

4aNS4. Ocean acoustics at BBN and beyond. Ira Dyer (Dept. of Ocean Eng., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, dyerira@msn.com)

Earlier known exclusively as underwater acoustics, ocean acoustics was pursued almost from BBN's beginning, the firm created by Richard Bolt and Leo Beranek. Ship noise, including that of submarines, and also sonar detection performance are perhaps still classified as underwater acoustics, while sound propagation and noise in the ocean have come to be classified as ocean acoustics. Given its creators, it is no surprise that contributions at BBN to ocean acoustics to about 1970 were based on fundamental concepts in architectural acoustics. These are in essence a set of geometrical ideas within which complicated multi-mode waves propagating in rooms can be reduced to simple theoretical and measurable aggregates expressing the mean-square wave motion. Thus, ray-averaged theories of propagation, wave coherence, and noise in the ocean were developed and used at BBN, as they were elsewhere, and are still in use today. By about 1970 it became clear that improved measurement tools, and increased understanding of the ocean state, including the bottom, could be exploited by theories and measurements of the phase as well as the mean-square motion. This story is traced from the perspective of those at BBN in the early years, to the present state of ocean acoustics.

10:00–10:20 Break


Leo Beranek and the firm Bolt Beranek and Newman have played a defining role in the formulation of noise policy in America. The firm that he and Richard Bolt founded in 1948 with fewer than a half-dozen others grew to become the world’s largest acoustical consulting firm with more than 2000 employees. Two decades later in 1971, Leo Beranek was a key founder of the Institute of Noise Control Engineering of the U.S.A. The Institute, which inaugurated the INTER-NOISE series of annual noise congresses in 1972 under Beranek’s direction, played a major role in the enactment by the Congress of the Noise Control Act of 1972. NCA-72 identified the Environmental Protection Agency as the leading Federal agency with oversight responsibilities responsible for implementing the noise policies defined by the Congress. In 1981, funding for EPA's noise program was withdrawn. Since then, leadership at the Federal level for implementing a coordinated national noise policy has been absent, but a dozen Federal agencies remain active in the noise field. With the exception of aircraft, no product emission regulations on major sources of noise are enforced today. To rectify this situation, Leo Beranek has recently been playing a leadership role in a concerted effort to rejuvenate America’s national noise policy.

4aNS6. Standards for acoustical instruments. Alan H. Marsh (16072 Santa Barbara Ln., Huntington Beach, CA 92649-2155)

Prior to 1930, sounds were classified from subjective judgments of loudness. General-purpose instruments that could make objective measurements of sounds were not yet developed, nor were there any national or international standards with specifications for the performance characteristics of such instruments. In recognition of the need for instruments that could provide objective measurements of a sound, as well as the need for a national consensus on the characteristics of a sound to be measured, the newly founded Acoustical Society of America established Sectional Committees in 1930 to develop such standards under the procedures of the American Standards Association (now the American National Standards Institute). A draft of a standard for sound level meters prepared by the subcommittee for Sound Levels and Sound Level Meters was considered at a meeting of the Sectional Committee in...
May 1934. After incorporating revisions in response to comments and suggestions, the draft standard for sound level meters was approved for publication by the American Standards Association on 17 February 1936. This paper describes the development of sound level meters and other instruments; along with related national and international standards, in the years since that auspicious beginning.

11:00

4aNS7. History of acoustical consulting at BBN. Eric W. Wood and Eric E. Ungar (Acentech, 33 Moulton St., Cambridge, MA 02138, ewood@acentech.com)

A book is being written describing the history of acoustical consulting at Bolt Beranek and Newman Inc. (BBN). It highlights people that joined the firm during the early years, the unique culture they established, the major clients and projects, noteworthy contributions made by the staff, and 40 years later the transition of BBN's acoustical consulting practice to Acentech. The book is based on interviews of colleagues and friends, primarily by Deborah Melone, a former BBN editor. This presentation reviews the early days of the consulting partnership formed by Bolt and Beranek in 1948, its initial growth, and its achievements.

11:20

4aNS8. Books on acoustics. Neil A. Shaw (Menlo Scientific Acoustics, Inc., P.O. Box 1610, Topanga, CA 90290, menlo@ieee.org)

The legacy of a man is not limited to just his projects. His writings in many cases are a more lasting, and a definitely more accessible, monument. For 60 years, Leo L. Beranek has produced books on acoustics, acoustic measurements, sound control, music and architecture, noise and vibration control, concert halls, and opera houses in addition to teaching and consulting. His books are standard references and still cited in other books and in technical and professional articles. Many of his books were among, if not, the first comprehensive modern treatment of the subject and many are still foremost. A review of Dr. Beranek’s many books as well as some anecdotes about the circumstances and consequences of same will be presented.

11:35

4aNS9. My 65 years in acoustics. Leo L. Beranek (975 Memorial Dr., Ste. 804, Cambridge, MA 02138-5755, beranek@leo.ieee.org)

My entry into acoustics began as research assistant to Professor F. V. Hunt at Harvard University. I received my doctorate in 1940 and directed the Electro-Acoustic Laboratory at Harvard from October 1940 until September 1945. In 1947, I became a tenured associate professor at MIT, and, with Richard H. Bolt, formed the consulting firm Bolt and Beranek, that later included Robert B. Newman, becoming BBN. My most significant contributions before 1970 were design of wedge-lined anechoic chambers, systematization of noise reduction in ventilation systems, design of the world’s largest muffler for the testing of supersonic jet engines at NASA’s Lewis Laboratory in Cleveland, speech interference level, NC noise criterion curves, heading New York Port Authority’s noise study that resulted in mufflers on jet aircraft, and steep aircraft climb procedures, and publishing books titled, Acoustical Measurements, Acoustics, Noise Reduction, Noise and Vibration Control, and Music, Acoustics and Architecture. As President of BBN, I supervised the formation of the group that built and operated the ARPANET (1969), which, when split in two (using TCP/IP protocol) became the INTERNET (1984). Since then, I have written two books on Concert Halls and Opera Houses and have consulted on four concert halls and an opera house.

THURSDAY MORNING, 27 MAY 2004

CONFERENCE ROOM L, 9:00 TO 11:45 A.M.

Session 4aPA

Physical Acoustics: Guided Waves, Resonators and Structures

Sameer Madanshetty, Chair

Mechanical Engineering Department, Kansas State University, Rathbone Hall, Manhattan, Kansas 66506-5106

Contributed Papers

9:00

4aPA1. Optical measurement of acoustic streaming in a waveguide. Tetsushi Biwa (Nagoya Univ., Chikusa-ku, Nagoya 464-8603, Japan, biwa@nuap.nagoya-u.ac.jp), Michael W. Thompson, and Anthony A. Atchley (Penn State Univ., University Park, PA 16802)

The generation of time-averaged fluid velocities by acoustic waves is known as acoustic streaming. The recent development of traveling-wave thermoacoustic engines has accelerated the need for a deeper understanding of acoustic streaming. In these devices, mass flow associated with traveling-wave streaming results in a large heat leak. However, no quantitative measurements of traveling-wave streaming have been found in the literature. In the present work, we study acoustic streaming induced by traveling- and standing-wave fields in a cylindrical waveguide. Drivers attached to each end of the waveguide are driven at the same frequency. By tuning the relative magnitudes and phases of the voltages supplied to these drivers, both traveling- and standing-wave fields can be generated. Laser Doppler anemometry has been used to make quantitative measurements of acoustic streaming induced by a standing wave in a resonator [M. W. Thompson and A. A. Atchley, in Nonlinear Acoustics at the Beginning of the 21st Century, edited by O. V. Rudenko and O. A. Sapozhnikov (Moscow, 2002), Vol. 1, pp. 183–190]. We apply this experimental technique to the traveling-wave field in this waveguide, and compare the local acoustic variables and the streaming.
4aPA2. Optical measurement of acoustic streaming in a standing wave with a temperature gradient. Michael W. Thompson and Anthony A. Atchley (Penn State Grad. Prog. in Acoust., University Park, PA 16802, mwt126@psu.edu)

Laser Doppler anemometry (LDA) with burst spectrum analysis (BSA) is applied to the measurement of Rayleigh-type acoustic streaming in a cylindrical standing-wave resonator constructed partly of glass and filled with air. The resonator is driven sinusoidally, and the axial component of the streaming-velocity field is measured along the resonator’s axis and across its diameter. At large acoustic amplitudes, distortion of the streaming-field is observed. This distortion is attributed to two effects: (1) fluid inertia and (2) a steady temperature gradient resulting from thermoacoustic heat transport along the resonator’s inner wall. Measurements of the streaming field are made at several different acoustic amplitudes and with the resonator either wrapped in foam insulation, suspended within an air-filled tank, or surrounded by a water jacket in order to control the magnitude of the temperature gradient.

9:30

4aPA3. Demonstration by optical visualization of acoustic re-radiation from $A_4$ and $S_0$ waves on submerged shells. P. K. Raju, A. C. Ahily, H. Cao (Dept. of Mech. Eng., Auburn Univ., Auburn, AL 36849), and H. Überall (Catholic Univ., Washington, DC 20064)

Elastic waves propagating on thin shells may be classified like for the Lamb waves on plates ($A_0$, $S_0$, $A_1$, $S_1$, …), and the Scholte-Stoney wave ($A$) in the fluid loading. The present study deals with evacuated shells of semi-infinite extent and a uniformly curved front surface, on which acoustic pulses are incident head-on. The incident signals cause the generation of the mentioned shell waves at a critical angle of incidence; these may be observed by their reradiation into the fluid at the same critical angle. Our demonstration of the reradiated pulses consists in a numerical evaluation of the incident and reradiated fields, and a visualization of the corresponding pulses in a tank experiment employing the Schlieren method. While the A wave could not be observed because of its rapid decay following its generation, we were able to demonstrate by both methods the generation of reradiated pulses of the $A_0$ and the $S_0$ wave, at the same time verifying the value of the critical angle of their generation.

9:45


The objective was to develop a numerical model for investigating the effect of obstructions in the opening on the frequency and amplitude of flow-excited cavity pressure fluctuations. A hybrid Reynolds-averaged Navier–Stokes/large eddy simulation (RANS/LES) method was used. The RANS/LES method ensures that important turbulence scales outside of the boundary layer are resolved, while permitting more economical RANS modeling of the smaller turbulent scales immediately adjacent to walls. Solutions were obtained at the speed where the cavity Helmholtz resonance was excited and at a speed well below resonance. A solution was also obtained where the cavity opening was partially obstructed. Results were compared to experimental data obtained for the same configuration. In all cases, the frequency of the shear layer oscillations was accurately predicted. Resonant cases also showed excellent agreement on the amplitude of the first harmonic. Some initial difficulties in predicting the amplitude in the nonresonant case are attributed to excessive dissipation in the LES calculations. Nonresonant cases are being resimulated using a less dissipative advective scheme and SGS turbulence model. A comparison of some flow field results showed reasonable agreement in certain cases.

10:00–10:15 Break

10:15

4aPA5. Experimental and numerical investigation of flow through an oscillated acoustic resonator. Christopher C. Daniels (Univ. of Akron, Akron, OH 44325), Joshua Finkbeiner, Bruce M. Steinertz (NASA Glenn Res. Ctr., Cleveland, OH 44135), Mahesh Athavale, and Maciej Pinda (CFD Res. Corp., Huntsville, AL 35805)

An acoustic resonator was oscillated experimentally at the fundamental gas resonant frequency to develop standing pressure waves. The conical shaped resonator contained openings that provided an air passage from a pressurized cavity through the resonator to the ambient environment. For several pressure differentials applied across the resonator, the rate of air flow is reported for no resonator oscillation, and for on-resonant and near-resonant frequency oscillations. When compared to no oscillation and near-resonant frequency oscillation at all pressure differentials, the standing waves within the resonator reduced the flow of air through the system when oscillated on-resonance. A two-dimensional numerical model was developed using a commercial CFD package to simulate the gas flow within the system. The mass flow of air through the oscillating resonator was matched using the numerical simulations. For a low value of differential pressure, the simulations showed the reversal of gas flow into the high pressure cavity during a part of the cycle, while allowing free flow of air during other cycle phases. For a high value of differential pressure, no flow reversal was observed. The goal of the study was to determine the applicability of nonlinear acoustics to advanced seal concepts at NASA Glenn Research Center.

10:45

4aPA6. Vibroacoustics of three-dimensional drum silencer. Lixi Huang (Dept of Mech. Eng., The Hong Kong Polytechnic Univ., Kowloon, Hong Kong)

When low-frequency sound waves travel down a duct in which a segment of hard walls is replaced by membranes backed by side branch cavities, they are reflected as a result of sound radiation by the induced membrane vibration. The reflection is effective over a broad frequency band when high tension is applied in the axial direction of the duct. The device is thus called a drumlike silencer, and its existing vibroacoustics theory is based on a two-dimensional model in which the membrane behaves as a string. This study extends the theory to three dimensions, in which the membrane covers a finite rectangular cavity with all edges fixed. It is shown analytically and validated experimentally that the fixed edges of the drum silencer have no effect on the silencing performance of the device working in the cutoff frequency range of the duct and when there is no tension in the transverse direction. Finite-element method is used to simulate the full fluid–membrane coupling and the results are validated by known solutions. The validated numerical method is then applied to demonstrate the acoustic benefit of using a circular cavity for a rectangular duct.

10:45

4aPA7. Fluid-structure interaction in fast breeder reactors. A. A. Mitra, D. N. Manik (IIT Bombay, India), and P. A. Chellapandhi (IGCAR, Kalpakkam, India)

A finite element model for the seismic analysis of a scaled down model of Fast breeder reactor (FBR) main vessel is proposed to be established. The reactor vessel, which is a large shell structure with a relatively thin wall, contains a large volume of sodium coolant. Therefore, the fluid structure interaction effects must be taken into account in the seismic design. As part of studying fluid-structure interaction, the fundamental frequency of vibration of a circular cylindrical shell partially filled with a liquid has been estimated using Rayleigh’s method. The bulging and sloshing frequencies of the first four modes of the aforementioned system have been obtained using the Rayleigh–Ritz method. The finite element formulation of the axisymmetric fluid element with Fourier option (required due to seismic loading) is also presented.
The excitation of a liquid-filled cavity by ultrasonic waves was described as a parametric phenomenon [L. Adler and M. A. Breazeale, J. Acoust. Soc. Am. 48, 1077–1083 (1970)]. A standing ultrasonic wave is produced in the cavity by a drive transducer at one end and a rigid reflector at the other. Variations in the cavity length lead to frequencies lower than the drive frequency. Such a situation can be described by a modified Mathieu equation whose solution can be used to predict a threshold for parametric oscillation. Originally the threshold was assumed to decrease with frequency [L. Adler, Ph.D. dissertation, The University of Tennessee, 1969]. The apparatus used by Adler and Breazeale recently was refined for accurate measurement of the threshold amplitude for parametric excitation between 1 and 10 MHz. The measurement showed that in this range the threshold amplitude actually increases with increasing drive frequency. The results are compared with existing models.

11:15


This paper examines a localized mode of sound in a planar waveguide between rigid walls with a pair of identical Helmholtz resonators connected. Assuming the waveguide extends infinitely, a two-dimensional problem to the wave equation is solved within linear, lossless theory. By the localized mode are meant time-harmonic, stationary oscillations without radiation damping, which are connected vis-à-vis to the upper and lower walls, an antisymmetric localized mode can exist but no symmetric mode exists. In the antisymmetric mode, the sound pressure changes out of phase at frequencies lower than the lowest cutoff frequency of the duct modes and also a natural frequency of the resonator. The solution for the localized mode is represented by superposition of an infinite number of antisymmetric, evanescent duct modes. When the resonators are staggered on both walls, the localized mode disappears.

11:30


Edge resonance in elastic bodies of finite sizes is a specific phenomenon without analogs in wave fields in electrodynamics and acoustics. This phenomenon was examined by studying normal modes of finite elastic cylinders and plates. With excitation of the edge resonance, an unusual wave effect is formed. The eigenfrequency of an elastic body is independent of its geometrical size and, in some cases, even increases with increasing body dimension. Eigenfrequency of edge resonance is into the frequency range where only one propagating wave exists in the corresponding infinite cylinder or layer. It gives a basis for the general qualitative conclusion that edge resonance is a result of intensive excitation of evanescent waves. However, such an obvious idea does not open the nature of the edge mode formation. The analysis of longitudinal and shear components of propagating wave gives way to understanding specific features of reflection of normal wave in semi-infinite waveguides. The determining role of effects of transformation of longitudinal wave in shear wave and vice versa at reflection from free boundary is shown. Dependence of quantitative characteristics of these effects from Poisson’s ratio allows estimation of the quality factor of edge resonance and changes corresponding eigenfrequency.

### Session 4aPP

**Psychological and Physiological Acoustics: Loudness and Binaural Perception**

**Mary Florentine, Cochair**

*Department of Speech–Language Pathology and Audiology, Northeastern University, 360 Huntington Avenue, Boston, Massachusetts 02115*

H. Steven Colburn, Cochair

*Department of Biomedical Engineering, Boston University, 44 Cumming Street, Boston, Massachusetts 02215*

### Contributed Papers

#### 8:00

4aPP1. A study of listening habits in adolescents: Correlating stated loudness preferences with actual listening levels. Laura Warren, Jean Warren (Columbia College Chicago, 1684 Linden St., Des Plaines, IL 60018, warr808@aol.com), and Dominique Cheenne (Columbia College Chicago, Des Plaines, IL 60018)

Evidence suggests that children are damaging their hearing in substantial numbers [Niskar et al., J. Am. Med. Assoc. (1998)]. Conventional thinking would suggest that cultural norms and attitudes contribute to a desire in children to model what they have seen in the media, thus implying that they would be listening to music at levels that are considered harmful. Our study focused on a gender-balanced group of 316 elementary-age students and aimed at assessing a correlation between an attitudinal survey related to loud music and the children’s own listening levels. The study was broader in scope and in sample size than previous work [Fucci, 138th ASA Meeting, 11/99]. Findings were both surprising and encouraging, citing that a majority of children who expressed favoritism towards loud music listened to the presented samples at lower levels than expected. The study also proposes a set of listening level distribution curves that may prove useful for future studies with older participants.
4aPP2. Extracting magnitude estimations of loudness from pairwise judgments. Eugene Galanter (Columbia Univ., 460 Riverside Dr., New York, NY 10027, eg53@columbia.edu)

Four problems limit widespread applications of magnitude estimation scales. There is first a serious question about the meaning of the judgments. Second, the judgments appear to be intrinsically unreliable. Third, amalgamation methods used to strike averages from several observers are poorly understood. Finally, it is difficult to know how to evaluate a single entity as distinct from a stimulus domain by these methods. A new psychophysical method is described and demonstrated that can scale any perceptual or attitudinal continuum. Observers give a numerical estimate of the magnitude of a randomly selected stimulus relative to another such stimulus to which a computer has assigned a random number. The stimulus pairs vary at random from trial to trial. Ratios of these computer-person number pairs estimate the slope of the psychophysical function. The slope lets us normalize the judgments which can then be mapped onto the stimulus domain. Loudness functions for individuals are shown with none of the cusps or singularities of traditional magnitude estimations from individuals. [Work supported in part by NASA.]

8:30

4aPP3. Directional loudness of narrow-band noises in an anechoic sound field. Ville P. Sivonen and Wolfgang Ellermeier (Dept. of Acoust., Aalborg Univ., Fredrik Bajers Vej 7 B5, DK-9220 Aalborg, Denmark, vps@acoustics.au.dk)

In order to investigate the effect of sound incidence angle on loudness across a larger set of parameters than have been used in most previous studies, a listening experiment was carried out using a loudspeaker setup in an anechoic chamber. Eight subjects, whose absolute hearing thresholds and head-related transfer functions (HRTFs) were measured, participated in a total of 22 sessions each. On each trial their task was to judge which of two narrow-band noises sounded louder. These judgments were used in an adaptive procedure to find loudness matches between a frontal reference location and seven other sources, positioned both in the horizontal and median planes. Sound incidence angle, center frequency, and overall SPL were varied in the procedure. The results show that loudness is not constant over sound incidence angles, with matches varying over a range of 10 dB, and showing considerable frequency dependency. The pattern of results also varies substantially between subjects, but can be accounted for by interindividual variations in the listeners’ HRTFs. [Work supported by Brüel & Kjær Sound & Vibration Measurement A/S.]

8:45

4aPP4. Perceptual and procedural learning in interaural cue discrimination. Helena Constantinides and David R. Moore (Univ. Lab. of Physiol., Parks Rd., Oxford OX1 3PT, UK, hconstantinides@doctors.org.uk)

Training in stimulus detection, discrimination, or identification generally leads to an improvement in performance, apparently in two phases. Learning during the early, rapid phase (within hours) has been attributed predominantly to procedural or task learning. Perceptual (true stimulus) learning is believed to occur mainly in the slower, later phase. Two experiments were undertaken to determine whether significant perceptual learning also occurs during the early phase. In the first, four groups of listeners were trained for 1 h in interaural level difference (ILD) discrimination. Stimuli were presented within a different, two- or three-interval task for each group. The following day, complete generalization of learning from the trained to the untrained tasks was demonstrated. In the second experiment, two groups of listeners were trained in an ILD discrimination task. In one group the stimulus was at a fixed, suprathreshold level (ILD = 15 dB), and in the other the stimulus was varied adaptively around the listener's discrimination threshold. The post-train mean stimulus discrimination threshold in the suprathreshold group was 3.9 dB, and in the threshold group was 2.4 dB (p < 0.01). These findings suggest perceptual learning contributes significantly to the early, rapid phase of performance improvement in interaural cue discrimination.

9:00


Listeners are remarkably sensitive to interaural incoherence. They can easily distinguish between a coherence of 1.00 and a coherence of 0.99. Interaural incoherence leads to the sensation of apparent source width; it is also held to be responsible for the masking level difference and for the creation of binaural pitch. However, incoherence per se is only a statistical description of signals; it is not indicative of any particular auditory binaural property. To discover the relevant binaural attribute(s), three-interval oddity experiments were performed using narrow-band noises that were perfectly coherent or slightly incoherent. The listener's task was to identify the incoherent noise. First, it was shown that different frozen noises with identical values of incoherence could often differ greatly in detectability. Subsequent experiments studied the detectability of incoherent frozen noises that were particularly strong or particularly weak according to selected binaural difference functions. These functions tested different rules for combining interaural phase differences (IPD) and interaural level differences (ILD)—either weighted sums of squares or rules that combine IPD and ILD to form a fluctuating lateralized image. The goal was to discover the best predictor of incoherence detectability. [Work supported by the NIDCD Grant DC 00181.]

9:15


The lateralization of the Huggins pitch (HP) was measured using a direct estimation method. The background noise was initially N0 or N π, and then the laterality of the entire stimulus was varied with a frequency-independent interaural delay, ranging from −1 to + 1 ms. Two versions of the HP boundary region were used, stepped phase and linear phase. When presented in isolation, without the broadband background, the stepped boundary can be lateralized on its own but the linear boundary cannot. Nevertheless, the lateralizations of both forms of HP were found to be almost identical functions both of the interaural delay and of the boundary frequency over a two-octave range. In a third experiment, the same listeners lateralized sine tones in quiet as a function of interaural delay. Good agreement was found between lateralizations of the HP and of the corresponding sine tones. The lateralization judgments depended on the boundary frequency according to the expected hyperbolic law except when the frequency-independent delay was zero. For the latter case, the dependence on boundary frequency was much slower than hyperbolic. [Work supported by the NIDCD grant DC 00181.]

9:30

4aPP7. Earedness: Left-eared and right-eared listeners. William M. Hartmann, Peter Xinya Zhang (Dept. of Phys. and Astron., Michigan State Univ., 4230 BPS Bldg., East Lansing, MI 48824), and John F. Culling (Cardiff Univ., Cardiff CF1 3YG, UK)

The Huggins pitch (HP) stimulus known as HP-, is created with a broadband background noise having an interaural phase difference of zero, together with a narrow boundary region wherein the interaural phase varies with frequency. At the spectral center of the boundary region the interaural phase is 180 deg. Therefore, HP- is symmetrical with respect to the two ears. Despite the symmetry, most listeners hear the HP image strongly lateralized to one side of the head. Some hear it on the right; others hear it on the left. Two surveys, involving 51 listeners, found that these perceptions do not change when the headphones are reversed. Extensive experiments with five listeners found that the lateralization directions were usually insensitive to variations in the frequency of the boundary region (more than two octaves). The left or right preference was strong enough that listeners chose alias locations (differing from a more central location by 360 deg) on the preferred side when various frequency-independent interaural delays and phase shifts were added to the HP.
stimulus. The experiments suggest that given ambiguous stimuli, listeners exhibit cuedness—a preference similar to, but not as strong as, handedness. [Work supported by the NIDCD Grant DC 00181.]

9:45

Noninvasive measurements utilizing magnetoencephalography (MEG) have been used to study how sound stimulus features are represented in the human brain. These measurements have successfully revealed how, for example, tone frequency, periodicity, and intensity are encoded. Here, the auditory-evoked magnetic fields in change of the magnitude of the interaural cross correlation (IACC) were analyzed. The IACC of the stimuli was controlled by mixing two independent bandpass noises in appropriate ratios. The auditory stimuli were binaurally delivered through silicon tubes and earpieces inserted into the ear canals. All source signals had the same sound-pressure level. Nine volunteers with normal hearing took part in this study. The auditory-evoked fields were recorded using a neuramagneto-meter in a magnetically shielded room. Combinations of a reference stimulus (IACC=1.0) and test stimuli (IACC=0.2,0.6,0.85) were presented alternately at a constant 0.5-s interstimulus interval, and the MEGs were recorded and averaged more than 50 times. The results showed that the peak amplitude of N1m, which was found above left and right temporal lobes around 100 ms after the stimulus onset, significantly decreased with increasing IACC. The N1m latencies were not affected by IACC.

10:00–10:15 Break

10:15
4aPP9. Psychophysical calibration of auditory range control in binaural synthesis with independent adjustment of virtual source loudness. William L. Martens (Faculty of Music, McGill Univ., Montreal, QC H3A 1E3, Canada, wlm@music.mcgill.ca)

This paper reports the results of a study designed to evaluate the effectiveness of synthetic cues to the range of auditory images created via headphone display of virtual sound sources processed using individualized HRTFs. The particular focus of the study was to determine how well auditory range could be controlled when independent adjustment of loudness was also desired. Variation in perceived range of the resulting auditory spatial images was assessed using a two-alternative, forced choice procedure in which listeners indicated which of two successively presented sound sources seemed to be more closely positioned. The first of the two sources served as a fixed standard stimulus positioned using a binaural HRTF measured at ear level, 1.5 m from the listeners head at an azimuth angle of 120 deg. The second source served as a variable loudness comparison stimulus processed using the same pair of HRTFs, with the same interaural time difference but with a manipulated interaural level difference. From the obtained choice proportions for each pairwise comparison of stimuli, numerical scale values for auditory source range were generated using Thurstone’s Case IV method for indirect scaling. Results provide a basis for calibrated control over auditory range for virtual sources varying in loudness.

10:30
4aPP10. Development of the sound localization cues in cats. Daniel J. Tollin (Dept. of Physiol., Univ. of Wisconsin—Madison, 1300 University Ave., Madison, WI 53706, tollin@physiology.wisc.edu)

Cats are a common model for developmental studies of the psycho-physical and physiological mechanisms of sound localization. Yet, there are few studies on the development of the acoustical cues to location in cats. The magnitude of the three main cues, interaural differences in time (ITDs) and level (ILDs), and monaural spectral shape cues, vary with location in adults. However, the increasing interaural distance associated with a growing head and pinnae during development will result in cues that change continuously until maturation is complete. Here, we report measurements, in cats aged 1 week to adulthood, of the physical dimensions of the head and pinnae and the localization cues, computed from measurements of directional transfer functions. At 1 week, ILD depended little on azimuth for frequencies <6–7 kHz, maximum ITD was 175 μs, and for sources varying in elevation, a prominent spectral notch was located at higher frequencies than in the older cats. As cats develop, the spectral cues and the frequencies at which ILDs become substantial (>10 dB) shift to lower frequencies, and the maximum ITD increases to nearly 370 μs. Changes in the cues are correlated with the increasing size of the head and pinnae. [Work supported by NIDCD DC05122.]

10:45

This study investigated aspects of the precedence effect (PE) known as fusion and localization dominance in children 4–5 years of age. Stimuli were three, 25-ms noise bursts (2-ms rise/fall times) with 250-ms ISI. On PE conditions the lead stimulus was presented from one of six locations in azimuth and the lag was at 0 deg. Lead-lag delays varied from 5 to 100 ms. Localization was measured using an identification paradigm. Fusion was measured separately whereby subjects reported whether a single auditory event or two auditory events were perceived. Children reported sounds on 75% of trials (fusion threshold) at delays ranging from 15 to 35 ms. Below fusion thresholds, the localization of the lead was similar to that of single-source stimuli. Above fusion thresholds lead localization was significantly degraded, persisting out to 100 ms. Localization of the lag was poor at all delays on which it was reported as being heard. According to these results localization dominance (difficulty localizing the lag) in children persists at greater delays than fusion, which is consistent with findings obtained in adult subjects. The range of delays over which these effects are robust in children is longer than the range observed in adults.

11:00

To determine the mechanism by which direct-to-reverberant energy ratio (D/R) is discriminated, a previous experiment (Larsen et al., 2003) reported INDS at three different D/R values (∼10, 0, +10 dB). Of three proposed mechanisms, the data favored a model based on detection of spectral variation of the magnitude of the source-receiver transfer function. However, a model based on detection of spectral centroid of the received signal could not be rejected conclusively. In order to determine the relative salience of spectral variance versus spectral centroid cues, the current study reports findings from two experiments. These data will be useful in the development of a general model for distance perception. D/R was fixed at 0 dB, because D/R discrimination is most acute (Larsen et al., 2003) and spectral variations are largest (Jetz, 1979) at that ratio. In experiment one, signals (300 ms Gaussian noise; 10 ms onset/offset time) were modified such that spectral variation cues were removed by roving the compression of the spectral envelope variations. In the second experiment, spectral centroid cues were removed by using band-limited signals (300 ms Gaussian noise; 10 ms onset/offset time) and roving the center frequency. Listener sensitivity in both these experiments will be compared to model predictions.
4aPP13. The influence of target-masker similarity on across-ear interference in dichotic listening. Douglas Brungart and Brian Simpson (Air Force Res. Lab., 2610 Seventh St., Wright-Patterson AFB, OH 45433)

In most dichotic listening tasks, the comprehension of a target speech signal presented in one ear is unaffected by the presence of irrelevant speech in the opposite ear. However, recent results have shown that contralaterally presented interfering speech signals do influence performance when a second interfering speech signal is present in the same ear as the target speech. In this experiment, we examined the influence of target-masker similarity on this effect by presenting ipsilateral and contralateral masking phrases spoken by the same talker, a different same-sex talker, or a different-sex talker than the one used to generate the target speech. The results show that contralateral target-masker similarity has the greatest influence on performance when an easily segregated different-sex masker is presented in the target ear, and the least influence when a difficult-to-segregate same-talker masker is presented in the target ear. These results indicate that across-ear interference in dichotic listening is not directly related to the difficulty of the segregation task in the target ear, and suggest that contralateral maskers are least likely to interfere with dichotic speech perception when the same general strategy could be used to segregate the target from the masking voices in the ipsilateral and contralateral ears.

11:30

4aPP14. The relative immunity of high-frequency transposed stimuli to low-frequency binaural interference. Leslie R. Bernstein and Constantine Trahiotis (Dept. of Neurosci. and Dept. of Surgery (Otolaryngol.), Univ. of Connecticut Health Ctr., Farmington, CT 06030)

We have recently demonstrated that high-frequency transposed stimuli, having envelopes designed to provide high-frequency channels with information similar to that normally available in only low-frequency channels, yield threshold-ITDs and extents of laterality comparable to those obtained with conventional low-frequency stimuli. This enhanced potency of ITDs conveyed by high-frequency transposed stimuli, as compared to conventional high-frequency stimuli, suggested to us that ITDs conveyed by transposed stimuli might be relatively immune to the presence of low-frequency binaural interferers. To investigate this issue, threshold-ITDs and extents of laterality were measured with a variety of conventional and transposed targets centered at 4 kHz. The targets were presented either in the presence or absence of a simultaneously gated diotic noise centered at 500 Hz, the interferer. As expected, the presence of the low-frequency interferer resulted in substantially elevated threshold-ITDs and reduced extents of laterality for the conventional high-frequency stimuli. In contrast, these interference effects were either greatly attenuated or absent for ITDs conveyed by the high-frequency transposed targets. The results will be discussed in the context of current models of binaural interference. [Work supported by NIH DC 04147, NIH DC04073, NIH DC 002304.]

11:45

4aPP15. Performance benefits of adaptive, multimicrophone, interference-canceling systems in everyday environments. Joseph G. Desloge, Martin J. Zimmer, and Patrick M. Zurek (Sensimetrics Corp., 48 Grove St., Somerville, MA 02144, desloge@sens.com)

Adaptive multimicrophone systems are currently used for a variety of noise-cancellation applications (such as hearing aids) to preserve signals arriving from a particular (target) direction while canceling other (jammer) signals in the environment. Although the performance of these systems is known to degrade with increasing reverberation, there are few measurements of adaptive performance in everyday reverberant environments. In this study, adaptive performance was compared to that of a simple, non-adaptive cardioid microphone to determine a measure of adaptive benefit. Both systems used recordings (at an Fs of 22 050 Hz) from the same two omnidirectional microphones, which were separated by 1 cm. Four classes of environment were considered: outdoors, household, parking garage, and public establishment. Sources were either environmental noises (e.g., household appliances, restaurant noise) or a controlled noise source. In all situations, no target was present (i.e., all signals were jammers) to obtain maximal jammer cancellation. Adaptive processing was based upon the Griffiths-Jim generalized sidelobe canceller using filter lengths up to 400 points. Average intelligibility-weighted adaptive benefit levels at a source distance of 1 m were, at most, 1.5 dB for public establishments, 2 dB for household rooms and the parking garage, and 3 dB outdoors. [Work supported by NIOSH.]
4aSA2. The transmission loss through curved sandwich composite structures. Sebastian Ghinet, Nouredine Atalla (Dept. of Mech. Eng., Universite de Sherbrooke, 2500 Blvd. Universite, Sherbrooke, QC J1K 2R1, Canada), and Haisam Osman (The Boeing Co., Huntington Beach, CA 92647)

The principal aim of this work is to present a model for the transmission loss of sandwich composite cylindrical shells. The effects of membrane, bending, and transverse shearing as well as rotational inertia are considered in all of the layers composing the structure. The elastic constants of any layer are related to the orthotropic angle-ply defined as the angle of the principal directions of the layers material to the global axis of the shell. Fundamental relations are expressed using the dynamic equilibrium relations of the unit forces in the structure. The structural impedance, critical frequencies and ring frequency are computed numerically in the general case of symmetrical laminated composite shell. Their expressions are developed in a wave approach context. A general eigenvalue approach to compute the dispersion curves of such structures is presented. Using these curves, the radiation efficiency, the modal density, the group velocity an the resonant and nonresonant transmission loss are computed and used within SEA framework to predict the sound transmission loss of these structures. Comparisons with existing models and experimental data are also discussed.

8:45

4aSA3. Theoretical investigation of noise transmission into a finite cylinder. Deyu Li and Jeffrey S. Vipperman (Dept. of Mech. Eng., Univ. of Pittsburgh, Pittsburgh, PA 15261)

A new mathematical model for characterizing noise transmission into a finite elastic cylindrical structure with application to a ChamberCore composite cylinder is presented. A plane wave obliquely impinges on the structure, the external sound field is approximated by the solution for an infinite cylinder, and the internal sound field is solved with the structural and acoustic modal interaction method. The noise reduction spectrum for characterizing noise transmission into the cylinder is defined, and the analytical model for the calculation of the noise reduction spectrum is developed. The analytical results show that the cavity resonances dominate the noise transmission into the finite cylinder, and the longitudinal acoustic modes play an important role in the noise transmission at the low frequencies. These results are matched with experimental results.

9:00


Controlling the sound radiation from a source can be either performed at the source or in the transmission path or both. Modifications of the transmission path can be in the form of a close fitting enclosure. Simple models are available to estimate the effectiveness of the close fitting enclosure. However, if the sound source is submerged in water, the presence of the fluid within and external to the close fitting enclosure, and the type of absorption material within the enclosure, may influence the enclosure effectiveness. Modeling the fluid solely by its characteristic impedance may not lead to reasonable results. The influence of the fluid on the behavior of the close fitting enclosure should be taken into account in the modeling. Numerical results are presented that compare the insertion loss estimated using “simple” close fitting enclosure models to that from multi-layer theory. [Work supported by ONR.]

4aSA5. The general 2-D moments via integral transform method for acoustic radiation and scattering. Jerry R. Smith, Jr. (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, smithjr@nsncc.navy.mil) and Mark S. Mirozni (The Catholic Univ. of America, Washington, DC 20064)

The moments via integral transform method (MITM) is a technique to analytically reduce the 2-D method of moments (MoM) impedance double integrals into single integrals. By using a special integral representation of the Green’s function, the impedance integral can be analytically simplified to a single integral in terms of transformed shape and weight functions. The reduced expression requires fewer computations and reduces the fill times of the MoM impedance matrix. Furthermore, the resulting integral is analytic for nearly arbitrary shape and weight function sets. The MITM technique is developed for mixed boundary conditions and predictions with basic shape and weight function sets are presented. Comparisons of accuracy and speed between MITM and brute force are presented. [Work sponsored by ONR and NSWCCD ILIR Board.]

9:30

4aSA6. Improved theory for acoustic radiation from a stiffener attached to a plate. Robert C. Haberman (BBN Technologies, Old Mystic Mill, 11 Main St., Mystic, CT 06355, rhaberman@bbn.com)

A flexural wave in a plate that interacts with a stiffener produces sound. Physically the stiffener provides concentrated line forces and moment onto the plate that generates a subset of waves with wave numbers less than the acoustic wave number. Typically the stiffener is modeled as a beam containing bending and torsional waves and mathematically coupled to the flexural waves in the plate. In reality, however, the stiffener exhibits resonant modes and is coupled to in-plane compression and shear waves in the plate. These effects produce in-plane forces at the plate–stiffener line junction (i.e., shear tractions), and high frequency-dependent interaction forces and torques that result in complex frequency-dependent sound waves along with scattered and transmitted waves in the plate and stiffener. The theory associated with resonant stiffeners fully coupled to all three waves types in a plate with heavy fluid loading is presented, along with several numerical examples. The examples will demonstrate the conditions under which the simpler models are valid and conditions that require more sophisticated modeling. Application of the theory to problems in statistical energy analysis is briefly discussed.

9:45

4aSA7. Prediction of the sound reduction index with the modal theory. Alain Tisseyre, Cecile Courre, Andre Moulinier, and Thomas Buzzi (Tisseyre & Associes, 16 chemin de Manel, 31 400 Toulouse, France)

A procedure of sound reduction index calculation was developed on the basis of the modal theory. This calculation uses the study of the vibratory behavior and of the radiation of the plate. The vibratory study makes it possible to characterize the displacement of the wall. The latter is developed over a base: sinusoidal or polynomial. The use of a sinusoidal base makes it possible to obtain satisfactory results without limitation of frequency. In this case, the only possible boundary condition corresponds to a plate simply supported. The use of a polynomial base only makes it possible to obtain results in the low frequencies. On the other hand, in this case, several boundary conditions are possible: embedded, simply supported. In both cases, modeling several types of walls is possible: isotropic or orthotropic wall, single or multiple walls. Walls, orthotropic due to their geometry, were more particularly studied. It is the case in particular of alveolar slabs, bricks and ribbed sheets. A formulation of the acoustic transparency was established and programmed, and various results are presented. Currently, a calculation algorithm is developed in order to allow polynomial base modeling on a more significant frequency range.
10:00–10:15  Break

10:15


Fluid-loaded vehicle structures, such as fuselages and hulls, often have spatially periodic discontinuities such as braces, ribs, and attachments. The structural motion, the acoustic radiation and scattering, and the interior sound field are of interest. Calculating the motion of fluid-loaded structures is a difficult task because of the high complexity and a disparity of length scales requiring high numerical resolution. Discontinuities cause the structural response to occur in a broad spectrum of spatial wavenumbers, and to exhibit stop-band and pass-band behavior. Structural discontinuities broaden the spatial wavenumber spectrum, causing both supersonic (radiating) and subsonic (nonradiating) waves. An analysis method called local-global homogenization (LGH) is used to predict directly the low wavenumber smooth response of periodic fluid-loaded structures in a self-contained manner. The low wavenumber part of the response is efficiently coupled to the acoustic field, since low wavenumbers correspond to supersonic phase speeds. In the LGH reformulation, an infinite order operator that must be truncated for numerical solution governs the equivalent smooth global problem. The numerical implementation is described, including the treatment of boundary conditions, and sample calculations are compared to exact solutions. The size of the computational problem is dramatically reduced by the LGH analytical reformulation.

10:30


Efficient calculation of vehicle interior noise is a challenging task. Classical acoustic boundary element calculations become costly at high frequencies due to the very large number of elements required and must be solved repeatedly for broadband applications. An alternative energy-intensity boundary element method has been formally developed that employs uncorrelated broadband directional intensity sources to predict mean-square pressure distributions in enclosures. The boundary source directivity accounts for local correlation effects and specular reflection. The method is applicable to high modal density fields, but it is not restricted to the usual low-absorption, diffuse, and quasi-uniform assumptions. The approach can accommodate fully specular reflection, or any combination of diffuse and specular reflection. This new method differs from the classical version in that the element size is large compared to an acoustic wavelength and equations are not solved on a frequency-by-frequency basis. These differences lead to an orders-of-magnitude improvement in computational efficiency. In vehicle interiors the sources are typically the vibrating walls of the enclosure. A special treatment for wall vibration sources has been developed for use with the new boundary element method. Calculations of spatially varying mean-square pressures agree well with computationally intensive modal solutions.

10:45


Analytical-numerical matching (ANM) is an analysis scheme that combines a low-resolution global numerical solution with a high-resolution local analytical solution and a matching solution to form a uniformly valid composite solution. The application of ANM to harmonically oscillating bodies in a fluid leads to a novel reformulation of boundary-value problems in fluid-dynamics and acoustics and to a new kernel encountered in the integral equation is handled simply within the analytical local solution. The numerical implementation utilizes a smoothed Green’s function solution to the governing equation with a distributed source term. Because the singular behavior has been removed from the numerical aspect of the problem, the approach converges rapidly and exhibits insensitivity to node (control point) location. The method allows low-resolution numerics to be combined with analytical corrections to obtain high accuracy solutions using a robust calculation scheme. The method has recently been extended to include viscous effects within the local solution. The appropriately modified global solution remains irrational and can be expressed in terms of a smoothed potential. Sample results are shown for radiation from plates up to high frequencies. Ongoing research on ANM BEM is described, including work to include non-linear convection within the viscous flow effects.

11:00

4aSA11. Acoustic emission analysis of shuttle thermal protection system. John Lane, Jeffery Hooker, Christopher Immer (ASRC Aerosp., MS: ASRC-10, KSC, FL 32899), and James Walker (NASA, Huntsville, AL 35812)

Acoustic emission (AE) signals generated from projectile impacts on reinforced and advanced carbon/carbon (RCC and ACC) panels, fired from a compressed-gas gun, identify the type and severity of damage sustained by the target. This type of testing is vital in providing the required return to flight (RTF) data needed to ensure continued and safe operation of NASA’s Space Shuttle fleet. Conventional AE analysis techniques require time domain processing of impulse data, along with amplitude distribution analysis. It is well known that identical source excitations can produce a wide range of AE signal amplitudes. In order to satisfy RTF goals, it is necessary to identify impact energy levels above and below damage thresholds. Spectral analysis techniques involving joint time frequency analysis (JTFA) are used to reinforce time domain AE analysis. JTFA analysis of the AE signals consists of short-time Fourier transforms (STFT) and the Huang–Hilbert transform (HHT). The HHT provides a very good measure of the instantaneous frequency of impulse events dominated by a single component. Identifying failure modes and cracking of fibers from flexural and/or extensional mode acoustic signals will help support in-flight as well as postflight impact analysis.

11:15

4aSA12. The elastic shell T-matrix theory evaluated. Michael Werby (NRL Code 7181, Stennis Space Ctr., MS 39529) and H. Uberall (Catholic Univ. of America, Washington, DC 20064)

The T-matrix method initiated by Waterman was extended in a clever paper by Peterson and the Varadans in 1980 to the complicated problem of elongated elastic shells. There have been some questions on whether the method converges for thin shells due to the impression that one must be able to inscribe a spherical surface in the annular region of the shell. The method may be derived by making use of several of the constraining equations that arise naturally and we show that the expression derived in 1980 is generally correct at least for objects with mirror symmetry. We present some details of the theoretical development with some calculations.

11:30

4aSA13. Reciprocity in the wave reflection and transmission problem. Yuri I. Bobrovnitskii (Dept. of Vibroacoustics, Mech. Eng. Res. Inst., 4 Maly Kharitonievsky Str., 101990 Moscow, Russia, bobrovnii@orc.ru)

Matrices of reflection and transmission coefficients of plane waves in media or normal waves in waveguides are, in general case, not symmetric. When all the waves are of the propagating type, the matrix can be symmetrized by normalizing the wave amplitudes with the wave power flow. For evanescent (inhomogeneous) waves this is not valid because of their zero power flow. This difficulty is overcome in the present paper. The main result is that a reflection and transmission matrix becomes symmetric if the wave amplitudes are normalized with the certain energy-like quan-
tunity that coincides with the power flow in the case of propagating waves. The result is valid for all types of waves (including those with complex wave numbers) and for all media and waveguides where the classical reciprocity theorem is valid. All symmetry relations, known in the literature for reflection and transmission coefficients follow from the result as particular cases. The result is useful in analysis of multimodal sound fields in composite media. It is illustrated in examples with inhomogeneous waves in fluids and solid waveguides.

THURSDAY MORNING, 27 MAY 2004

IMPERIAL BALLROOM B, 8:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Poster Session III

Melissa Epstein, Chair

Biomedical Sciences, University of Maryland Dental School, 666 West Baltimore Street, Baltimore, Maryland 21201

Contributed Papers

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

4aSC1. Aeroacoustics of [s]. Michael S. Howe (College of Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, mshowe@bu.edu) and Richard S. McGowan (CReSS LLC, Lexington, MA 02420)

The theory of the sibilant fricative [s] is formulated and solved as a mathematical problem of aeroacoustics. Air is forced through the constrictions between the tongue and the hard palate by the intra-oral pressure, forming a jet that strikes the upper incisors and leaves the mouth through a gap between the upper and lower incisors. The principal source of sound is the diffraction of jet turbulence pressure fluctuations by the incisors. The spectrum of these pressure fluctuations incident on the teeth is modeled analytically using an empirical formula adapted from boundary layer theory. Predictions are made of the far field acoustic pressure spectrum by reference to measured and estimated values of vocal tract dimensions and intra-oral pressure. Predicted spectra compare well with observations. The principal spectral peaks are determined by vocal tract physiology anterior to the tongue–palate constriction. The theory furnishes the field acoustic pressure spectrum by reference to measured and estimated values of vocal tract dimensions and intra-oral pressure. Predicted spectra compare well with observations. The principal spectral peaks are determined by vocal tract physiology anterior to the tongue–palate constriction. The theory furnishes the field acoustic pressure spectrum by reference to measured and estimated values of vocal tract dimensions and intra-oral pressure. Predicted spectra compare well with observations. The principal spectral peaks are determined by vocal tract physiology anterior to the tongue–palate constriction. The theory furnishes the mathematical model in a nonmathematical manner. [Work supported, in part, by Grant NIDCD-01247 to CReSS LLC.]


This study examines how monolingual French speakers produced the stop voicing distinction in syllable-initial and syllable-final stops embedded in various sentence contexts. Voicing-related differences in percentages of closure voicing, durations of aspiration, closure, and vowel were analyzed as a function of two experimental variables: the voicing class of the sound adjacent to the target stop [voiced vowel (\pa/-/a/ context), voiceless consonant (/pa/-/s/ context)], and the position of the stop within a syllable (syllable-initial, -final). Results from ANOVA showed that despite variations among speakers, group-based patterns surfaced in three contexts (i.e., syllable-initial stops in the /pa/-/a/ and /pa/-/s/ contexts and syllable-final stops in the /pa/-/a/ context): /b, d, g/ were more aspirated, preceded by longer vowels and were more frequently phonated than /p, t, k/. Closure durations for /f, d, g/ were shorter than those for /p, t, k/ in the /pa/-/a/ context only. Group patterns were not found for syllable-final stops in the /pa/-/s/ context. Results from discriminant analyses indicated that closure voicing was the variable that contributed the most to the phonological voicing distinction in all conditions.


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4aSC4. The effect of emphatic stress on CV coarticulation. Golnaz Modarresi (Dept. of Linguist., B5100, Univ. of Texas, 1 University Station, Austin, TX 78712-0198, golie@mail.utexas.edu) and Harvey M. Sussman (Univ. of Texas, Austin, TX 78712-0198)

The effect of emphatic stress on CV coarticulation was investigated in the speech of one male and one female native speaker of American English using locus equation slope as a measure of CV coarticulation. Stressed real word C1V2C2 tokens where C1=/b,d,g/ and V2=/i, e, u, a, æ, u, o, ø/ were put in carrier sentences with the, thirty, or two preceding the test word. Each sentence was read three times in a normal manner and three times with emphasis on the test token. This resulted in a total of 486
tokens per speaker (3 stop consonants * 3 V₁ contexts * 9 V₂ contexts * 2 emphasis patterns * 3 repetitions). Locus equation slopes were derived by plotting F₂ onset of C₁ against V₂ F₂ mid-vowel frequency and fitting a regression line to data points. Consistent duration closure, V₂ duration, Fᵡ₂, and amplitude were also measured. Despite a significant increase in the acoustic correlates of emphasis, locus equation slopes remained constant as a function of emphasis and varied as a function of place of articulation. This study provides further evidence of the stability of locus equation slopes as phonetic descriptors of stop place of articulation. [Work supported by NIH.]

4aSC5. The stability of locus equation slopes across stop consonant voicing/aspiration. Harvey M. Sussman (Dept. of Linguist., B5100, Univ. of Texas, 1 University Station, Austin, TX 78712-0198, sussman@mail.utexas.edu) and Golnaz Modarresi (Univ. of Texas, Austin, TX 78712-0198)

The consistency of locus equation slopes as phonetic descriptors of stop place in CV sequences across voiced and voiceless aspirated stops was explored in the speech of five male speakers of American English and two male speakers of Persian. Using traditional locus equation measurement sites for F₂ onsets, voiceless labial and coronal stops had significantly lower locus equation slopes relative to their voiced counterparts, whereas velars failed to show voicing differences. When locus equations were derived using F₂ onsets for voiced stops that were measured closer to the stop release burst, comparable to the protocol for measuring voiceless aspirated stops, no significant effects of voicing/aspiration on locus equation slopes were observed. This methodological factor, rather than an underlying phonetic-based explanation, provides a reasonable account for the observed flatter locus equation slopes of voiceless labial and coronal stops relative to voice cognates reported in previous studies [Molis et al., J. Acoust. Soc. Am. 95, 2925 (1994); O. Engstrand and B. Lindblom, PHONUM 4, 101–104]. [Work supported by NIH.]

4aSC6. The Nationwide Speech Project: A multi-talker multi-dialect speech corpus. Cynthia G. Clopper and David B. Pisoni (Speech Res. Lab., Dept. of Psych., Indiana Univ., Bloomington, IN 47405, cclopper@indiana.edu)

Most research on regional phonological variation relies on field recordings of interview speech. Recent research on the perception of dialect variation by naive listeners, however, has relied on read sentence materials in order to control for phonological and lexical content and syntax. The Nationwide Speech Project corpus was designed to obtain a large amount of speech from a number of talkers representing different regional varieties of American English. Five male and five female talkers from each of six different dialect regions in the United States were recorded reading isolated words, sentences, and passages, and in conversations with the experimenter. The talkers ranged in age from 18 and 25 years old and they were all monolingual native speakers of American English. They had lived their entire life in one dialect region and both of their parents were raised in the same region. Results of an acoustic analysis of the vowel spaces of the talkers included in the Nationwide Speech Project will be presented. [Work supported by NIH.]

4aSC7. Effects of coda voicing and aspiration on Hindi vowels. Claire Lampp and Heidi Reklis (Dept. of Linguist., Univ. of North Carolina at Chapel Hill, CB #3155, Chapel Hill, NC 27599, lampp@email.unc.edu)

This study reexamines the well-attested coda voicing effect on vowel duration [Chen, Phonetics 22, 125–159 (1970)], in conjunction with the relationship between vowel duration and aspiration of codas. The first step was to replicate the results of Maddieson and Gandour [UCLA Working Papers Phonetics 31, 46–52 (1976)] with a larger, language-specific data set. Four nonsense syllables ending in [oen]- followed by [k, kh, g, gh] were read aloud in ten different carrier sentences by four native speakers of Hindi. Results confirm that longer vowels precede voiced word-final consonants and aspirated word-final consonants. Thus, among the syllables, vowel duration would be longest when preceding the voiced aspirate [gh]. Coda voicing, and thus, vowel duration, have been shown to correlate negatively to vowel F₁ in English and Arabic [Wolf, J. Phonetics 6, 299–309 (1978); de Jong and Zawaydeh ibid., 30, 53–75 (2002)]. It is not known whether vowel F₁ depends directly on coda voicing, or is determined indirectly via duration. Since voicing and aspiration both increase duration, F₁ measurements of this data set (which will be presented) may answer that question.

4aSC8. The effect of consonantal context on intensity distribution in vowels. Ewa Jacewicz and Robert Allen Fox (Dept. of Speech and Hearing Sci., The Ohio State Univ., Columbus, OH 43210, jacewicz.1@osu.edu)

Overall vowel intensity varies intrinsically with vowel quality and in terms of immediate consonantal context. These variations affect the levels of formants F₁–F₄ and F₀, which change in consonantal contexts according to a vowel—specific pattern [Jacewicz, J. Acoust. Soc. Am. 114, 2395 (2003)]. The current study further examines the differential distribution of acoustic energy across and within continuous frequency bands in the vowel as a function of vowel quality and consonantal context. The analysis technique used here was adopted from Shuijter and Van Heuven [J. Acoust. Soc. Am. 100, 2471–2485 (1996)], who measured differences in spectral intensity distribution across frequency bands as a function of stress. The intensity distribution of the vowels i, i, æ, u, w in American English were measured across eight frequency bands in a stressed [C1VC2] context (where C₁ = C₂). Contexts were chosen which have the greatest (e.g., voiceless fricatives) and the smallest (e.g., voiced stops) effect on changes in relative amplitude of formant frequencies and F₀ for each particular vowel. Preliminary results indicate that intensity variations across frequency bands as a function of consonantal context have a potential to change the overall spectral balance. [Work supported by NIDCD R03 DC005560–01.]

4aSC9. Subglottal coupling and vowel space. XueMin Chi and Morgan Sonderegger (Speech Commun. Group, RLE, MIT, Cambridge, MA 02139, smore@mit.edu)

A model of acoustic coupling between the oral and subglottal cavities predicts discontinuities in vowel formant prominences near resonances of the subglottal system. One discontinuity occurs near 1300–1500 Hz, suggesting the hypothesis that this is a quantal effect [K. N. Stevens, J. Phonetics 17, 3–46 (1989)] dividing speakers’ front and back vowels. Recordings of English vowels (in /HdV/ environments) for several male and female speakers were made, while an accelerometer attached to the neck area was used to capture the subglottal waveform. Statistics on our subglottal resonance measurements are given and compared with prior work. Qualitative agreement is shown between the resonator model and diphthong data with time-varying F₂ for several speakers. Comparison of the second vowel formant and second subglottal formant tracks across all speakers, analysis of the formant spaces spanned by each speaker’s vowel data, and a survey of vowel formant data for a sample of the world’s languages support the possibility that a speaker’s second subglottal resonance divides front and back vowels. Possible implications for theories of vowel inventory structure [e.g., J. Lijencrants and B. Lindblom, Language 48, 839–862 (1972)] are discussed. [Work supported by NIH Grant DC00075.]

We measured the formant shifts of a vowel in the context of a nasal and investigated whether human perception is able to compensate for such shifts. According to the acoustic theory, nasal coupling causes a modification on the spectrum, including formant frequency shift. The first goal of this study is to confirm that the formant frequencies actually shift due to nasalization. Based on several measurements of formant frequencies of various vowels in nasal contexts, we confirmed that the first formant (F1) tends to shift in a more central direction when nasalized. In English, vowels should be perceived as the same phoneme regardless of nasalization. In other words, listeners might have the capability to compensate for such formant shifts. The second goal of this study is to examine this compensation effect by a perceptual experiment. For stimuli, we synthesized a nonnasal vowel V0 that has the same formant frequencies as a nasalized vowel V1. A continuum was also synthesized between V0 and the nonnasalized version of V1. Results show V1 is more correctly identified than V0, which suggests the existence of the compensation effect.

4aSC11. Voiceless stop duration under narrow focus and in clear speech. Jeanette A. Ortiz (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208), Ann R. Bradlow, and Janet B. Pierrehumbert (Northwestern Univ., Evanston, IL 60208)

Talkers typically hyperarticulate when producing individual speech segments under conditions of sentence-level, or narrow focus and of clear speech, which is speech produced in response to known sound perception difficulties on the part of the listener. This study investigated whether narrow focus and clear speech elicit similar changes in the speech signal. A comparison was conducted between the effects of these two phenomena on the durations of voiceless stops /k/ and /t/ in word-initial and word-medial positions of trochaic, disyllabic words. Both narrow focus and clear speech yielded longer closure and aspiration durations when /k/ and /t/ were in word-initial position. For word-medial stops, narrow focus yielded no change in closure duration and a lengthening of aspiration duration for both /k/ and /t/. In contrast, clear speech yielded no change (/t/) or an increase (/k/) in closure duration, and, more interestingly, significant decreases in aspiration duration for both /k/ and /t/. The different effects of narrow focus and clear speech on voiceless stop production suggest that the local hyperarticulation of sentence-level focus and the global hyperarticulation of clear speech probably do not arise from the same underlying organizational structures.

4aSC12. Rosa’s roses: Reduced vowels in American English. Edward Flemming (Dept. of Linguist., Stanford Univ., Stanford, CA 94305-2150, flemming@stanford.edu)

Beginning phonetics students are taught that American English has two contrasting reduced vowels, transcribed as [ə] and [i], illustrated by the unstressed vowels in the minimal pair Rosa’s versus roses. However, little seems to be known about the precise nature or distribution of these vowels. This study explores these questions through acoustic analysis of reduced vowels in the speech of 12 American English speakers. The results show that there is a fundamental distinction between the mid central [ə] vowel that can occur in unstressed word-final position (e.g., in Rosa), and high reduced vowels that occur in most other unstressed positions, and might be transcribed as [i]. The contrast between pairs like Rosa’s and roses derives from this difference because the word-final [ə] is preserved when an inflectional suffix is added, so the schwa of Rosa’s is similar to the final vowel of Rosa, whereas the unstressed vowel of roses is the high [i] reduced vowel quality found elsewhere. So the standard transcription of the reduced vowel contrast is justified, but the widespread use of [ə] to transcribe word-internal reduced vowels is misleading—mid reduced vowels are generally only found in stem-final position.

4aSC13. Child directed speech, speech in noise and hyperarticulated speech in the Pacific Northwest. Richard Wright, Lesley Carmichael, Alicia Beckford Wassink, and Lisa Galvin (Dept. of Linguist., Univ of Washington, Box 354340, Seattle, WA 98195-4340, rawright@u.washington.edu)

Three types of exaggerated speech are thought to be systematic responses to accommodate the needs of the listener: child-directed speech (CDS), hyperspeech, and the Lombard response. CDS (e.g., Kuhl et al., 1997) occurs in interactions with young children and infants. Hyperspeech (Johnson et al., 1993) is a modification in response to listeners difficulties in recovering the intended message. The Lombard response (e.g., Lane et al., 1970) is a compensation for increased noise in the signal. While all three result from adaptations to accommodate the needs of the listener, and therefore should share some features, the triggering conditions are quite different, and therefore should exhibit differences in their phonetic outcomes. While CDS has been the subject of a variety of acoustic studies, it has never been studied in the broader context of the other “exaggerated” speech styles. A large crosslinguistic study was undertaken that compares speech produced under four conditions: spontaneous conversations, CDS aimed at 6–9-month-old infants, hyperarticulated speech, and speech in noise. This talk will present some findings for North American English as spoken in the Pacific Northwest. The measures include f0, vowel duration, F1 and F2 at vowel midpoint, and intensity.

4aSC14. Formants of Japanese function particles. Setsuko Shirai (Dept. of Linguist., Univ of Washington, P.O. Box 354340, Seattle, WA 98195-4340)

There is much debate about whether vowel centralization results from reduction of undershoot. The results of research conducted by Keating and Huffman (1984) indicated that Japanese vowels in prose were formed closer to the center of vowel chart than in words. However, they did not provide any statistical information and the information about the preceding consonants in the prose. Thus, there was a possibility that the preceding consonants led to the vowel centralization through undershoot. To address this question, I conducted research, in which Japanese function vowels [a, e, o] were compared with content vowels. The results showed that F1 of function /a/ following /g/ (average 609.8 Hz) was statistically lower than content /a/ (average 696.3 Hz) [F(1,65)=73.40, p<.0001]. However, there were no significant differences of F2 of /a/ following /d/ and F2 of /a/ following /t/ between function and content. At first, it appeared centralization played a role. However, close examination of the results indicated that vowel undershoot was the source of the /a/ centralization. Japanese function /a/ was statistically shorter than content /a/ and there was a significant correlation between duration and normalized vowel displacements for /a/ [Pearson’s r = 0.466, p<.0001]. This short duration caused the difference of F1 between content /a/ and function /a/.

4aSC15. Word-length and context effects on the acoustics of /ai/. Chandan R. Narayan (Dept. of Linguist., Univ of Michigan, 4080 Frieze Bldg., Ann Arbor, MI 48109, cnarayan@umich.edu)

The acoustics of American English /ai/ are investigated. This study investigated whether speakers maintain an invariant slope of the F2 transition across the vowel shortening effects of increased word length and the vowel lengthening effects of following obstruent voicing. Six speakers recorded minimal /ai/ triplets varying in word length and post-vocalic place and voicing (i.e., hide, Heidi, Heidelberg; hype, hyper, hyperness). Analysis of both temporal and spectral characteristics revealed systematic effects of word length and voicing. With increasing word length /ai/ duration significantly decreased across all variables, except in voiceless contexts where there was not a significant decrease from two- to three-syllable words. Before voiced obstruents F2 transition onset increased significantly from one- to two-syllable words with offset frequencies remaining stable. In the three-syllable word condition, the F2 transition offset dropped significantly. Importantly, F2 transition slope remained stable.
The existence of the “bimodal” pattern in schwas in nonrhotic dialects through an acoustic experiment. It is predicted that there is a significant difference in formant values between lexical schwas and function schwas. Results to date indicate a significant difference in them between schwas in lexical versus function words, both between historical schwas and those derived from final /r/ reductions. Data from several additional nonrhotic subjects will be presented. Implications for intrusive r as well as for the phonological treatment of function words will be discussed. [Work funded by NSERC and SSHRC.]

4aSC18. Bimodal schwa: Evidence from acoustic measurements. Noriko Yamane-Tanaka, Bryan Gick, and Sonya Bird (Interdisciplinary Speech Res. Lab., Dept. of Linguist., Univ. of BC, E270-1866 Main Mall, Vancouver, BC V6T 1Z1, Canada, nrkyamane@aol.com)

The question of whether schwa is targeted or targetless has been the subject of much debate (Browman et al., 1992; Browman and Goldstein, 1995; Gick, 1999, 2002). Gick (2002) found that there is a pharyngeal constriction during schwa relative to rest position, and concluded that schwa is shwa targetless. This experimental evidence showed a “bimodal” pattern in schwa in a nonrhotic speaker, indicating that the subject has distinct schwas in lexical words and function words. The present study examines

4aSC19. Parametric synthesis of Korean alveolar stops. Gwanhi Yun (Dept. of Linguist., Univ. of Arizona, P.O. Box 210028, Tucson, AZ 85721-0028, ghyn@email.arizona.edu)

Korean alveolar stops were synthesized through MfTalk system to evaluate the parameters. Because of the insufficiency of reliable acoustic cues to distinguish three-way Korean stops, synthesized stops were expected to lack intelligibility and naturalness. In the first synthesis, AH (amplitude of aspiration during transition from C to the following V), duration of transition, formant frequencies, bandwidths of formants, and /T0 of the following vowels were employed as main parameters only at two anchor points, i.e., at the burst point and onset of the following vowel. As expected, the plain stops (33%) were not satisfactorily discriminated from tense stops (39%), and naturalness was below chance. However, the discrimination between plain/tense and aspirated stops was not so bad. Further, the intelligibility and naturalness of aspirated stops were so low. Thus, to compensate for lack of reliable parameters, two parameters were readjusted: (1) duration of AH and (2) duration of transition. The identification results showed that the longer AH and transition, the more accurate the perception of aspirated stops. Thus, it indicates that AH and duration of transition might work as a trade-off relation for the perception of aspirated stops, and so a relevant combination of parameters in SynthWork may synthesize more accurate Korean aspirated stops.

4aSC20. One-hand control of a speech synthesizer. Harold A. Cheyne II, Robert E. Beaudoin, Thomas E. von Wiegand, Kenneth N. Stevens (Sensimetrics Corp., 48 Grove St., Ste. 305, Somerville, MA 02144-2500, harold@sens.com), and Patrick M. Zurek (Sensimetrics Corp., Somerville, MA 02144-2500)

The long-term objective of this research is the development of a one-hand-controlled speech synthesizer, to give laryngectomized and other speech-impaired persons a means of producing higher-quality speech with less effort than currently available methods such as an electrolarynx or a text-to-speech system. To demonstrate the feasibility of a one-hand-controlled speech synthesizer, a system was constructed using a hand-held device similar to a pen connected to an articulated arm for measuring six degrees of freedom (three Cartesian and three rotational dimensions) as the user interface to an Hlsyn-based speech synthesizer. Through this interface, the user controls parameters for the first three formants, pitch, subglottal pressure, and glottal area. Parameter control was introduced progressively in that order to four participants who underwent training to produce synthesized speech composed of a subset of English phonemes: vowels, semivowels, diphthongs, /l/, and the glottal stop. The complexity of the synthesized speech targets also grew from monosyllabic utterances to short phrases over the training. After training, a separate group of four listeners compared the naturalness and intelligibility of the synthesized speech to the same utterances produced by the participants with a text-to-speech system. [Work supported by NIDCD Grant Number R43 DC006134-01.]
In a previous study exploring American English question intonation, we found that some speakers deviated considerably from expected question prosody. In this study, we focus on listener-rated acceptability of the various prosodic patterns observed for yes/no and wh questions. A variety of intonational patterns realized in both question utterances recorded from five female and three male professional speakers and in questions synthesized from several TTS voices of both genders was presented to listeners. Subjects judged the acceptability of each utterance in the context of a dialogue between a travel agent and customer. We hypothesized that question utterances with the expected intonational features (phrase-final fall in wh questions, phrase-final rise in yes/no questions) would be rated as more acceptable than question utterances with deviating intonational features, and that this result would hold for both natural and synthetic speech conditions. In addition, following our previous results, we hypothesized that the unexpected intonation pattern of phrase-final falls for yes/no questions would be more acceptable for lower-pitched than for higher-pitched voices. We also varied the prominence of the interrogative pronouns in synthetic wh questions in order to see whether simulating their high intonational prominence in natural wh questions improved the acceptability of synthetic wh questions.

Experiments indicate that non-nasal obstructions in human utterances can be replaced by “surrogate” segments, either produced by formant synthesis or recorded from other speakers, with virtually no change in speech quality or speaker identity [Hertz, Proc. IEEE 2002 Workshop on Speech Synthesis (2002)]. While the durational and spectral properties of the surrogate segments must be broadly appropriate to their target context, no speaker-specific tailoring is required. This paper describes follow-on experiments studying the perceptual consequences of replacing nasal consonants in human utterances with surrogate segments from different phonetic contexts, either synthesized or spoken by other speakers. These experiments indicate that the manipulated speech sounds natural when surrogate segment durations, and the formant transitions and nasalization characteristics of adjacent vowels, are appropriate. In certain contexts F0 is also perceptually salient. The spectral characteristics of surrogate nasal murmurs are often unimportant. In many cases, the perceived speech quality, phoneme identity, and speaker identity are unaffected even by a surrogate from a phoneme differing from the original. This paper highlights the perceptual results and explains their relevance to hybrid synthesis techniques that employ cross-speaker waveform concatenation and/or integrate waveform concatenation with formant synthesis. Utterances that exemplify these results will be played.

The goal of this work is to develop principles of overlapping gestures in obstructant-sonorant sequences in the word-initial position and sonorant-obstruct sequences in the word-final position. Consonant clusters such as sm in small are phonetically represented as a sequence of individual elements, but the exact perceptual representation is unclear. The modification during the production of these overlapping gestures may be driven partly by perceptual salience and partly by vocal tract aerodynamics. When two consonants occur next to each other, the same gestures may be made as for only one consonant. The aerodynamics of the vocal tract may account for the modification in the timing of the articulators during production and this modification can be incorporated as rules into HLsyn, a higher-level quasiarticulatory speech synthesizer that takes as inputs the pressures and

4aSC24. Advances in the acoustic correlates for nasals from analysis of MRI data. Tarun Pruthi and Carol Espy-Wilson (Dept. of Elect. and Computer Eng., AVW Bldg., Univ. of Maryland, College Park, MD 20742, tpruthi@glue.umd.edu)

MRI data for nasals is being used to simulate the nasal murmur spectrum in order to improve our understanding of nasals in speech. The data consists of American English nasals /n/, /l/, and /ŋ/ from one speaker [Story et al., J. Acoust. Soc. Am. 100, 537–554 (1996)]. A computer simulation model developed in our lab is being used for the purpose [Zhang and Espy-Wilson, J. Acoust. Soc. Am. (2004)]. A sufficiently good match has been obtained between the simulated and real spectra, particularly during the low frequencies (below 2000 Hz). Currently, only the maxillary and sphenoidal sinuses have been included in the model. The match between the two spectra is expected to improve further when frontal and ethmoidal sinuses are included in the model. It is our belief that this study will give us a better understanding of the acoustic manifestations of nasal manner and place in the nasal murmur spectrum and help us achieve our goal of finding speaker-independent acoustic parameters (APs) for them. It would also be interesting to see if this study can help us in finding speaker-dependent APs from the nasal murmur spectrum for our speaker recognition system. [Work supported by NSF Grant No. BCS0236707.]


A Matlab-based computer program for vocal tract acoustic response calculation (VTAR) has been developed. Based on a frequency-domain vocal tract model [Z. Zhang and C. Espy-Wilson, J. Acoust. Soc. Am. (2004)], VTAR is able to model various complex sounds such as nasals, rotics, and liquids. With input in the form of vocal tract cross-sectional area functions, VTAR calculates the vocal tract acoustic response function and the formant frequencies and bandwidths. The user-friendly interface allows directed data input for defined categories: vowels, nasals, nasalized sounds, consonant, laterals, and rotics. The program also provides an interface for input and modification of arbitrary vocal tract geometry configurations, which is ideal for research applications. [Work supported by NIH Grant 1 R01 DC05250-01.]

4aSC26. Obstruent-sonorant consonant sequences—Analysis by synthesis. Xiaomin Mou (MIT Speech Commun. Group, 77 Massachusetts Ave., 36-513, Cambridge, MA 02139, xmmou@mit.edu)

The goal of this work is to develop principles of overlapping gestures in obstructant-sonorant sequences in the word-initial position and sonorant-obstruct sequences in the word-final position. Consonant clusters such as sm in small are phonetically represented as a sequence of individual elements, but the exact perceptual representation is unclear. The modification during the production of these overlapping gestures may be driven partly by perceptual salience and partly by vocal tract aerodynamics. When two consonants occur next to each other, the same gestures may be made as for only one consonant. The aerodynamics of the vocal tract may account for the modification in the timing of the articulators during production and this modification can be incorporated as rules into HLsyn, a higher-level quasiarticulatory speech synthesizer that takes as inputs the pressures and
the flows of the vocal tract. Acoustic information extracted from the speech waveform is mapped into inputs for HLsyn. This analysis by synthesis approach is a method to develop a more precise picture of the planning stage during speech production where the acoustic phonetics must be carefully planned and modified to achieve the correct target sounds. [Work funded by a grant provided by NIH.]

4aSC27. Text-to-phonemic transcription and parsing into mono-syllables of English text. Yugal Jusgir Mullick, S. S. Agrawal, Smita Tayal, and Manisha Goswami (CSIO, CSIRComplex, 2nd Fl., Pusa Campus, New Delhi, India)

The present paper describes a program that converts the English text (entered through the normal computer keyboard) into its phonemic representation and then parses it into mono-syllables. For every letter a set of context based rules is defined in lexical order. A default rule is also defined separately for each letter. Beginning from the first letter of the word the rules are checked and the most appropriate rule is applied on the letter to find its actual orthographic representation. If no matching rule is found, then the default rule is applied. Current rule sets the next position to be analyzed. Proceeding in the same manner orthographic representation for each word can be found. For example, “reading” is represented as “rEdiNX” by applying the following rules:

- r→r move 1 position ahead
- e→Ed move 3 position ahead
- i→i move 1 position ahead
- n→NX move 2 position ahead, i.e., end of word.

The phonemic representations obtained from the above procedure are parsed to get mono-syllabic representation for various combinations such as CVC, CVCC, CV, CVCCV, etc. Example, the above phonemic representation will be parsed as rEdiNX→ /rE/ /dINX/. This study is a part of developing TTS for Indian English.


It is by now widely accepted that the articulation of speech is influenced by the prosodic structure into which the utterance is organized. Furthermore, the effect of prosody on F0 realization has been shown to be mainly phonological [Beckman and Pierrehumbert (1986); Selkirk and Shen (1990)]. This paper presents data from the F0 realizations of lexical tones in Standard Chinese and shows that prosodic factors may influence the articulation of a lexical tone and induce phonetic variations in its surface F0 contours, similar to the phonetic effect of prosody on segment articulation [de Jong (1995); Keating and Foureron (1997)]. Data were elicited from four native speakers of Standard Chinese producing all four lexical tones in different tonal contexts and under various focus conditions (i.e., under focus, no focus, and post focus), with three renditions for each condition. The observed F0 variations are argued to be best analyzed as resulted from prosodically driven differences in the phonetic implementation of the lexical tonal targets, which in turn is induced by pragmatically driven differences in how distinctive an underlying tonal target should be realized. Implications of this study on the phonetic implementation of phonological tonal targets will also be discussed.

4aSC29. Variation of Taiwanese tones in conversation. Pei-Yu Hsieh and Jane Tsay (Inst. of Linguist., Cheng Chung Univ., 160 San-Xing, Min-Xiong, Chia-Yi 621, Taiwan, ROC, g9015001@ccu.edu.tw)

In laboratory research on tonal coarticulation in Taiwanese, one study [H.-B. Lin, Ph.D. dissertation, Univ. of Connecticut (1988)] reported a perseveratory effect but no anticipatory effect, while another [S.-H. Peng, J. Phonetics 25, 371–400 (1997)] found a significant anticipatory effect. Peng also found tonal variation due to prosodic positions. Unlike these previous laboratory studies, this study attempts to investigate tonal coarticulation and prosodic effects on Taiwanese tones using natural conversations from the Taiwanese Spoken Corpus (Tsay and Myers, 2004), of which 56 min of recorded conversations were analyzed. Consistent with Lin, the results showed that tone is more affected by the preceding tone than by the following tone. The slope is more influenced by the preceding tone as well. As for prosodic effects, the results confirmed Peng, showing that F0 is the lowest in utterance-final position, while in other phrase-final positions it is slightly lower than in non-phrase-final position. This study thus demonstrates the results obtained in the laboratory do indeed carry over into actual conversation.

4aSC30. Contrasting the effects of duration and number of syllables on the perceptual normalization of lexical tones. Valter Ciocca, Alexander L. Francis, and Teresa S.-K. Yau (Div. of Speech & Hearing Sci., Univ. of Hong Kong, 5/F, 34 Hospital Rd., Hong Kong, HKSAR, China, vciocca@hkusua.hku.hk)

In tonal languages, syllabic fundamental frequency (F0) patterns ("lexical tones") convey lexical meaning. Listeners need to relate such pitch patterns to the pitch range of a speaker ("tone normalization") to accurately identify lexical tones. This study investigated the amount of tonal information required to perform tone normalization. A target CV syllable, perceived as either a high level, a low level, or a mid level Cantonese tone, was preceded by a four-syllable carrier sentence whose F0 was shifted (1 semitone), or not shifted. Four conditions were obtained by gating one, two, three, or four syllables from the onset of the target. Presentation rate (normal versus fast) was set such that the duration of the one, two, and three syllable conditions (normal carrier) was equal to that of the two, three, and four syllable conditions (fast carrier). Results suggest that tone normalization is largely accomplished within 250 ms or so prior to target onset, independent of the number of syllables; additional tonal information produces a relatively small increase in tone normalization. Implications for models of lexical tone normalization will be discussed. [Work supported by the RGC of the Hong Kong SAR, Project No. HKU 7193/00H.]

4aSC31. Effects of native language experience on perceptual learning of Cantonese lexical tones. Alexander L. Francis (Dept. of Audiol, and Speech Sci., Purdue Univ., Heavilion Hall, 500 Oval Dr., West Lafayette, IN 47907, francisa@purdue.edu), Valter Ciocca (The Univ. of Hong Kong, Hong Kong, China), and Lian Ma (Beijing Medical Univ., Beijing, China)

In a tonal language syllabic pitch patterns contribute to lexical meaning. Perceptual assimilation models of cross-language perception predict speakers of another tonal language should assimilate Cantonese lexical tones to native tonal categories, affecting identification, discrimination and acquisition. For nontonal language speakers, two possibilities exist. If pitch information is ignored, vowels with different tones should assimilate to the same native category, lowering performance. If tonal information is attended but unused in native categorization, Cantonese tones could be nonassimilable and therefore easily discriminated, and possibly easily identified or learned. Here, native speakers of Mandarin Chinese and American English were trained to identify Cantonese words differing in lexical tone. Discrimination and identification were tested before and after training. Both groups initially performed well on upper register tones (high level, high rising, mid level) and poorly on lower (low falling, low level, low rising). Mandarin listeners improved most at identifying low falling tones; English listeners improved most on low level and low rising tones. Training primarily appeared to improve listeners’ ability to make categorical decisions based on direction of pitch change, a feature reportedly under-attended by English speakers, but preferred by Mandarin speakers. [Work supported by research funding from The University of Hong Kong.]
4aSC32. Musical experience and Mandarin tone discrimination and imitation. Terry L. Gottfried, Ann M. Staby, and Christine J. Ziener (Dept. of Psych., Lawrence Univ., Appleton, WI 54912-0599, Terry.L.Reu-Gottfried@lawrence.edu)

Previous work [T. L. Gottfried and D. Riester, J. Acoust. Soc. Am. 108, 2604 (2000)] showed that native speakers of American English with musical training performed better than nonmusicians when identifying the four distinctive tones of Mandarin Chinese (high-level, mid-rising, low-dipping, high-falling). Accuracy for both groups was relatively low since listeners were not trained on the phonemic contours. Current research compares musicians and nonmusicians on discrimination and imitation of unfamiliar tones. Listeners were presented with two different Mandarin words that had either the same or different tones; listeners indicated whether the tones were same or different. Thus, they were required to determine a categorical match (same or different tone), rather than an auditory match. All listeners had significantly more difficulty discriminating between mid-rising and low-dipping tones than with other contrasts. Listeners with more musical training showed significantly greater accuracy in their discrimination. Likewise, musicians’ spoken imitations of Mandarin tones (model tokens presented by a native speaker) were rated as significantly more native-like than those of nonmusicians. These findings suggest that musicians may have abilities or training that facilitate their perception and production of Mandarin tones. However, further research is needed to determine whether this advantage transfers to language learning situations.

4aSC33. Teaching tone and intonation with the Prosody Workstation using schematic versus veridical contours. George D. Allen (College of Nursing, Michigan State Univ., East Lansing, MI 48824) and John B. Eulenberg (Michigan State Univ., East Lansing, MI 48824)

Prosodic features of speech (e.g., intonation and rhythm) are often challenging for adults to learn. Most computerized teaching tools, developed to help learners mimic model prosodic patterns, display lines representing the veridical (actual) acoustic fundamental frequency and intensity of the model speech. However, a veridical display may not be optimal for this task. Instead, stereotypical representations (e.g., simplified level or slanting lines) may help by reducing the amount of potentially distracting information. The Prosody Workstation (PW) permits the prosodic contours of both models and users’ responses to be displayed using either veridical or stereotypical contours. Users are informed by both visual displays and scores representing the degree of match of their utterance to the model. American English-speaking undergraduates are being studied learning the tone contours and rhythm of Chinese and Haua utterances ranging in length from two to six syllables. Data include (a) accuracy of mimicking of the models’ prosodic contours, measured by the PW; (b) quality of tonal and rhythmic production, judged by native speaker listeners; and (c) learners’ perceptions of the ease of the task, measured by a questionnaire at the end of each session.

4aSC34. Tone clarity in mixed pitch/phonoate type tones. Jean E. Andruski (Audiol. & Speech-Lang. Pathol., Wayne State Univ., 581 Manoogian Hall, 906 W. Warren Ave., Detroit, MI 48202, j_andruski@wayne.edu)

Lexical tone identity is often determined by a complex of acoustic cues. In Green Mong, a Hmong-Mien language of Southeast Asia, a small subset of tones is characterized by phonation type in addition to pitch height, pitch contour, and duration, which characterize the remaining tones of the language. In tones that incorporate multiple cues to tonal identity, what makes a tone clear, or easy to recognize? This study examines acoustic and perceptual data to address this question. Six native speakers of Green Mong were asked to produce 132 phonological CV words in sentence context, using a conversational speaking style. Seventeen native speakers of the language were then asked to categorize three tones which have similar falling contours, but are differentiated by phonation type (breathy, creaky, and modal). Tokens that were correctly identified by 100% of the listeners were compared with tokens that were relatively poorly identified. Data indicate that the breathy- and creaky-voiced tones are less susceptible to identification errors than the modal-voiced tone. However, the clearest tokens of the three tones are also differentiated by details of pitch contour shape, and by duration. Similarities and differences between acoustic cue values for the best and worst tokens will be discussed.

4aSC35. Temporal and spectral cues in Mandarin tone recognition. Ying-Yee Kong and Fan-Gang Zeng (Dept. of Cognit. Sci. and Dept. of Otalaryngol., Univ. of California–Irvine, Irvine, CA 92697)

Temporal cues contribute significantly to Mandarin tone recognition, but the relevance of formant frequencies is debatable. This study investigates the relative contribution of temporal and formant cues to tone recognition. Three sets of Mandarin stimuli were created. Recorded whispered speech was used to test the contribution of the formant cue, 1- and 8-band noise-modulated speech with 50-Hz envelope cutoff frequency was used to test the contribution of the temporal envelope cue, and 1- and 8-band speech with 500-Hz cutoff frequency was used to test the contribution of the periodicity cue. Four normal-hearing native Mandarin speakers participated. In quiet, subjects achieved an average 87% (8-band) and 82% (1-band) correct with the periodicity cue (500-Hz cutoff), but only 72% (8-band) and 55% (1-band) correct without the periodicity cue (50-Hz cutoff). Whispered speech produced an average 72% correct. From 10 to 10-dB signal-to-noise ratios, this pattern of results was largely preserved, with whispered speech and 8-band conditions without the periodicity cue showing similar performances, which was significantly better than the 1-band without the periodicity cue, but poorer than the 8-band with the periodicity cue. These results suggest all three cues contribute to Mandarin tone recognition, and there is a trade-off among these cues.

4aSC36. Tonal production of early and late Canadian Cantonese bilinguals. Connie K. So (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC V5A 1S6, Canada)

Previous studies show that native speakers have exhibited changes in both phonetic and phonemic characteristics in their L1 production and perception after living in an L2 community for years. Little is known if the phenomenon is also present in the tonal production of native speakers of tone languages who grew up in an English-speaking environment. This research examines both the phonetic and phonemic features of the six Cantonese tones (i.e., vowel duration, pitch contour, and tonal space) produced by two groups of adult Canadian Cantonese bilinguals: early (AOA: less than 1 year) and late (AOA: 10–13 years). The data collected were compared with those of a group of native Cantonese speakers (AOA: greater than 18 years). Analyses revealed that (i) both the early and the late bilinguals produced significantly longer vowel durations than did the comparison group, and that (ii) their pitch contours and tonal space were significantly different from those of the native speakers. In addition, the early bilinguals’ tonal productions were more deviant from the native productions than those of the late bilinguals.

4aSC37. Influence of Mandarin tone exposure on the processing of intonation by 14-year-old American adolescents: An fMRI study. Jo-Fu Lotus Lin, Toshiaki Imada, Patricia Kuhl (Dept. of Speech and Hearing Sci. and Inst. for Learning and Brain Sci., Box 357988, Univ. of Washington, Seattle, WA 98195), and Yue Wang (Simon Fraser Univ., Burnaby, BC V5A 1S6, Canada)

This study investigated, for American adolescents, whether the learning of non-native speech contrasts in one prosodic domain (Mandarin Chinese intonation) would influence the processing of non-native contrasts in another prosodic domain (Mandarin Chinese intonation). Two groups of 14-year-old American teenagers were tested using the functional magnetic resonance imaging (fMRI) technique, including eight who had received a
mediate priming. Further, if lexical discriminability modulates the degree and form-based representations are abstract, variation in VOT should not indicate that multiple linear regression is a better predictor of intelligibility using speech measures from the remaining 41 children. Preliminary results show that words with high-frequency counterparts should show larger specificity effects than words without counterparts and words with low-frequency counterparts. Implications of these results for the specificity and abstractness of phonetic representations in long-term memory will be discussed.

4aSC38. Representational specificity of within-category phonetic variation in the mental lexicon. Min Ju and Paul A. Luce (Dept. of Psych., Univ. at Buffalo, Buffalo, NY 14260, mju@buffalo.edu)

This study examines whether within-category variation in voice onset time (VOT) is encoded in long-term memory and affects subsequent word recognition, and whether these effects are modulated by the degree of lexical discriminability. Four long-term repetition-priming experiments were conducted using words containing word-initial voiceless stops varying in VOT. The magnitude of priming was compared between same and different VOT conditions in words with voiced counterparts (pat/bat) and words without voiced counterparts (cow/gow), and in words with high-frequency counterparts (cably/gab) and words with low-frequency counterparts (cab/gab). If veridical representations of each episode are preserved in memory, variation in VOT should have demonstrable effects on the magnitude of priming. However, if within-category variation is discarded and form-based representations are abstract, variation in VOT should not mediate priming. Further, if lexical discriminability modulates the degree of encoding of within-category variation, words with counterparts and words with high-frequency counterparts should show larger specificity effects than words without counterparts and words with low-frequency counterparts. A weighted combination of speech-acoustic measures may provide an objective assessment of speech intelligibility in deaf children that could be used to evaluate the benefits of sensory aids and rehabilitation programs. This investigation compared the accuracy of two different approaches, multiple linear regression and a simple neural net. These two methods were applied to identical sets of acoustic measures, including both segmental (e.g., voice-onset times of plosives, spectral moments of fricatives, second formant frequencies of vowels) and suprasegmental measures (e.g., sentence duration, number and frequency of intersentence pauses). These independent variables were obtained from digitized recordings of deaf children’s imitations of 11 simple sentences. The dependent measure was the percentage of spoken words from the 36 McGarr Sentences understood by groups of naive listeners. The two predictive methods were trained on speech measures obtained from 123 out of 164 8- and 9-year-old deaf children who used cochlear implants. Then, predictions were obtained using speech measures from the remaining 41 children. Preliminary results indicate that multiple linear regression is a better predictor of intelligibility than the neural net, accounting for 79% as opposed to 65% of the variance in the data. [Work supported by NIH.]

4aSC39. Predicting the intelligibility of deaf children’s speech from acoustic measures. Rosalie M. Uchanski (Dept. of Otolaryngol., Washington Univ. School of Medicine, St. Louis, MO 63110), Ann E. Geers (Univ. of Texas at Dallas, Dallas, TX), Christine M. Brenner (Moog Ctr. for Deaf Education, St. Louis, MO), and Emily A. Tobey (Univ. of Texas at Dallas, Dallas, TX)

A weighted combination of speech-acoustic measures may provide an objective assessment of speech intelligibility in deaf children that could be used to evaluate the benefits of sensory aids and rehabilitation programs. This investigation compared the accuracy of two different approaches, multiple linear regression and a simple neural net. These two methods were applied to identical sets of acoustic measures, including both segmental (e.g., voice-onset times of plosives, spectral moments of fricatives, second formant frequencies of vowels) and suprasegmental measures (e.g., sentence duration, number and frequency of intersentence pauses). These independent variables were obtained from digitized recordings of deaf children’s imitations of 11 simple sentences. The dependent measure was the percentage of spoken words from the 36 McGarr Sentences understood by groups of naive listeners. The two predictive methods were trained on speech measures obtained from 123 out of 164 8- and 9-year-old deaf children who used cochlear implants. Then, predictions were obtained using speech measures from the remaining 41 children. Preliminary results indicate that multiple linear regression is a better predictor of intelligibility than the neural net, accounting for 79% as opposed to 65% of the variance in the data. [Work supported by NIH.]

THURSDAY MORNING, 27 MAY 2004

CONFERENCE ROOM E, 8:25 TO 11:55 A.M.

Session 4aSP

Signal Processing Acoustics and Underwater Acoustics: Advances in Sonar and Imaging Techniques Including Interferometric, Synthetic and Tomographic Apertures

John Impagliazzo, Chair

John Impagliazzo Consulting, 2 Kimberly Drive, Wakefield, Rhode Island 02879-3804

Chair’s Introduction—8:25

Invited Papers

8:30

4aSPI. Time domain beamforming ASIC for a handheld bistatic imaging sonar system. Alice M. Chiang, Steven R. Broadstone, and John M. Impagliazzo (Teratech Corp., 77-79 Terrace Hall Ave., Burlington, MA 01803)

A high-resolution, handheld imaging sonar system has been developed by Teratech Corporation for the U.S. Navy. This is a 192-channel, dual-frequency bistatic sonar for Navy divers performing search and survey missions for underwater explosives. The goal is to provide the most compact and energy-efficient imaging system for the divers. The low power and small volume are a result of the development of Teratech’s Charge Domain Processing (CDP) technology. This technology has led to the development of a low-power 64-channel beamformer chip. As a result, only three beamformer chips are needed for the 192-element array. Until now, implementation of small, low-power sonar systems containing this many elements and forming enough beams to create an image was considered impossible. Test results and images obtained in Teratech’s acoustic test tank will be presented. [Work sponsored by ONR and OSD Small Business Innovative Research Program, Program Manager, Mr. Bruce Johnson, Naval Explosive Ordnance Disposal Technology Division.]
4aSP2. Interferometry and computer-aided tomography as an acoustic analysis tool. Ron J. Wyber (Midship Systems Pty. Ltd., 24 Farrer Pl., Oyster Bay, NSW 2252, Australia) and Brian G. Ferguson (Maritime Operations Div. DSTO, Pyrmont, 2009 NSW, Australia)

By using a wide band signal to measure the impulse response of a target it is generally possible to resolve the reflected signal into a sequence of impulses associated with discrete scattering features or structural waves. If two hydrophones, with an appropriate vertical separation, are used to receive the signal from a target, it is possible to determine the vertical position of the source for each impulse in the sequence from the relative phase of the analytic signal received at each of the hydrophones. The amplitude and phase information measured as the target rotates can then be used to form a three-dimensional image in which the position of the strong scattering sources is highlighted. Examples are given showing how the images formed provide information allowing features in the measured impulse response to be related to different scattering mechanisms such as specular target highlights, multiple reflections or structural waves. By characterizing the acoustic hotspots in the image in this way it is possible to provide feedback to the designer of a stealth target quantifying the contribution of individual features on the target to the overall target strength.

Contributed Papers

9:10
4aSP3. Acoustic Mining Imaging (AMI) project: An underwater acoustic camera for use in mine warfare. Colin Ellis and Ed Murphy (Thales Underwater Systems Pty. Ltd., 274 Victoria Rd., Rydalmere, NSW 2116, Australia, colin.ellis@au.thalesgroup.com)

This paper is submitted to detail the advances in sonar and imaging techniques and synthetic apertures being made in Australia by Thales Underwater Systems within a Australian Defence Acquisition Project termed Acoustic Mining Imaging (AMI). This paper will detail the development of the AMI underwater acoustic camera for the detection, classification and characterization of mines and other underwater objects in turbid water where optical imaging is ineffective. It will explain the history of the development from a DSTO concept to the current system. It will detail how the acoustic camera provides real-time millimetric resolution three-dimensional images and how the design has had to meet practical operational constraints to allow seamless mounting onto a small, remotely controlled underwater vehicle for mine disposal operations. An overview of the processing architecture and technical complexity of the AMI will be provided, including detail on the design and development of the required submillimeter transducers and 2D matrix array. The paper will reflect on the technical and operational challenges that had to be addressed. Finally, trial results will be presented that will demonstrate the real-time capability of the acoustic camera in various environments.

9:25
4aSP4. Array processing methods for identifying buried objects. Salah Bourennane (Institut Fresnel/GSM, UMR CNRS 6133, France)

Underwater object identification has been of great interest for a few years to acousticians (detection of boulders) or mines (detection of buried mines). Image and signal processing succeed in identifying objects lying on the sea bottom, however identification of an object buried in sediment remains complex. The purpose of this paper is to develop a complete identification of objects embedded in the sediment using array processing methods. We use a parametric source whose properties are based on the water nonlinear propagation characteristics, the signals are received on a uniform linear antenna. The higher order statistics of the recorded data and the high resolution methods are used to localize the objects and to estimate their ranges. This paper presents also a procedure which estimates discriminant parameters from the signals in order to classify the different objects. The developed algorithms are applied on the experimental underwater acoustic data. The obtained results show the good performance of our algorithms.

9:40
4aSP5. Sonar image enhancement via acoustic color. Nicola Neretti, Nathan Intrator (IBNS, Brown Univ., Providence, RI 02912), and Quyen Huynh (CSS, NSWCA, Panama City, FL 32407)

A novel method to enhance synthetic aperture sonar (SAS) images using the acoustic color of the returns from different targets is presented. The method enhances the SAS image using the internal absorption properties of targets in a way that enhances the differences between different types of targets (target signature). First, the algorithm detects the highlights in the sonar image as is found from the beam-formed processed image. These highlights represent edges of targets. Then the analysis is performed only at the echo returns which correspond to edges of targets. For each edge the raw data corresponding to the return from that edge is found. This data which has not yet been pulse-compressed using the pinging signal is analyzed using a detailed frequency representation. Then, the frequency representation of the pinging signal is subtracted from it so that a resulting frequency representation of the difference between the pinging signal and the returning signal is obtained. A normalization of the returned echo is performed before the subtraction to compensate for the lower energy of the returning echo. The same method can be extended to a more detailed analysis of internal structures of targets.

9:55
4aSP6. Bathymetry and seafloor acoustic backscatter imagery with a volume search sonar. Daniel S. Brogan and Christian P. de Moustier (Cfr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, daniel.brogan@unh.edu)

Volume search sonars designed for mine hunting applications could be used for environmental sensing, particularly seafloor relief and texture. This capability is explored with a system that transmits a stepped FM pulse over a 243 deg vertical fan beam centered on nadir. It receives with 27 pairs of beams, symmetrically steered about nadir in the fore-aft direction and spaced at 7.16-deg intervals across track. The receive beam pair geometry allows simultaneous views of the seafloor in forward, vertical, and aft profiles. Pulse compression, monopulse processing techniques, and temporal and spatial filtering are used to estimate bathymetry and seafloor acoustic backscatter imagery of a sandy bottom and a muddy bottom. Three monopulse techniques have been investigated: conjugate product, difference over sum, and reduced beamwidth which is the most promising for this application. Results are presented for data collected while surveying at roughly 25 knots, showing the combined effects of acoustic geometry and survey speed on the resolution of the bathymetry and acoustic backscatter imagery and on bottom coverage. [Work supported by NRL Grant No. N00173-00-1-G912.]

10:10–10:25 Break

10:25
4aSP7. Investigations into the application of a new sonar system for assessing fish passage in Alaskan rivers. Deborah Burwen (Alaska Dept. of Fish and Game, 333 Raspberry Rd., Anchorage, AK 99518), Suzanne Maxwell (Alaska Dept. of Fish and Game, Soldotna, AK 99669), and Carl Pflisterer (Alaska Dept. of Fish and Game, Fairbanks, AK 99701)

Over a two-year period (2002–2003) the Alaska Department of Fish and Game (ADF&G) has evaluated a relatively new sonar technology at several sites in Alaska to determine its applicability to counting migrating fish in rivers. The new system, called a dual frequency identification sonar (DIDSON), is a high-definition imaging sonar designed and manufactured.
The skeletal density of several massive reef coral species is strongly correlated with ocean temperature, and it has recently been proposed as a paleoclimate proxy offering access to multicityrury-long records of sea surface temperature. The presence of annual growth bands (from a few mm up to 1 or 2 cm wide) provides a chronological tool that enables changes in skeletal density to be traced through time. In this work, we investigate the feasibility of imaging the coral skeletal density using ultrasound. As the structure of coral is very close to that of trabecular bone, techniques successfully applied to bone imaging were assessed on coral. Ultrasonic measurements in the MHz frequency range were carried out in the laboratory on machined coral samples, using backscattering and through-transmission configuration. Parametric images were generated from the derivation of ultrasonic parameters such as acoustic impedance, velocity, backscatter coefficient, and attenuation. As x-ray absorption is directly related to the density of the coral, x-ray images were obtained as reference images. Preliminary results show qualitative agreement between x-ray images and ultrasound parametric images, indicating the potential for ultrasound to image coral. [Work supported by NSF through the Centre for Subsurface Sensing and Imaging Systems, Award Number EEC-9980821.]

10:55

4aSP9. Fourier and wavelet domain denoising of active sonar echoes. Peter G. Cable (BBN Technologies, 11 Main St., Mystic, CT 06355-3641), Sheila Shah, and Gary Butler (BBN Technologies, Arlington, VA 22209-3801)

Active sonar classification performance improves significantly when echo-to-background ratios increase above 10–15 dB. To achieve the improved echo waveform fidelity implied by increasing echo-to-background, preclassification processing methods are sought to improve echo waveform estimates. For this purpose a class of nonlinear techniques termed denoising, applied to efficient Hilbert space representations of transient signals, has been shown to yield nearly optimal estimation procedures for noise corrupted signals of unknown smoothness [D. L. Donoho and I. M. Johnstone, Biometrika 81 (1994)]. We have applied several versions of Fourier and wavelet domain denoising to noisy low-frequency target echoes and, for echoes near detection threshold, have demonstrated signal representation improvements equivalent to increases in echo-to-background of 4 dB. The theoretical foundations of denoising, including a new threshold algorithm, will be outlined and measures of performance for waveform estimation will be reviewed and discussed. The experimental methodology used and the results obtained for the test sonar echoes will be summarized and target classification implications of the results obtained from the analysis discussed. [Work supported by ONR.]

11:10

4aSP10. Survey of active sonar simulations. Diana F. McCammon (McCammon Acoust. Consulting, 475 Baseline Rd., RR3, Waterville, NS B0P 1VO, Canada)

In lieu of well-characterized measured data sets, time series simulations are often employed for testing active sonar systems. This paper reviews the simulation techniques and models that are being employed today and discusses some of the issues of fidelity that may arise with regard to range dependence, bistatic geometries, reverberation statistics and bottom interactions.

11:25

4aSP11. Imaging flaws in thin multilayered metal sheets using an array of guided mode transducers. Karl A. Fisher (Lawrence Livermore Natl. Lab., Livermore, CA, 94551)

In a multilayered elastic structure, acoustic energy can be focused into a particular layer by identifying a suitable guided wave mode. Once this guided mode is identified, low profile surface mounted interdigital transducers can be designed to excite preferentially this mode in the multilayered elastic structure. This has led to the development of a novel eight-channel linear array based on guided mode interdigital transducers. The array has been used to generate data for a time domain reconstruction algorithm. Resulting images reveal 1–2-mm length cracks in the center layer of a 300-μm-thick composite steel plate.

11:40


Leak detection and location in pressure vessels is commonly performed using microphones to detect airborne ultrasound generated by the leak turbulence. For spacecraft, this turbulence is generated outside the spacecraft and cannot be detected inside because the leak velocity is approximately the speed of sound. Instead, to detect and locate leaks in on-orbit spacecraft we monitor leak-generated guided ultrasonic waves within the plate-like spacecraft skin. We use cross-correlation to measure the deterministic behavior of the leak-generated noise. Measured leak-into-vacuum noise signals from two adjacent transducers, each correlated with the signal from a third reference transducer, are fed into headphones as a stereo pair. The direction to the leak can be determined by rotating the two transducers, or equivalently by selecting element pairs in a dense array. The leak can be precisely located through triangulation.
Underwater Acoustics: Acoustic Propagation and Internal Waves

Peter H. Dahl, Chair
Applied Physics Laboratory, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105-6698

Chair’s Introduction—7:55

Contributed Papers

8:00

4aUW1. Comparison between ocean-acoustic fluctuations in parabolic-equation simulations and estimates from integral approximations. Stanley M. Flattié and Michael D. Vera (Phys. Dept., Univ. of California at Santa Cruz, Santa Cruz, CA 95064, sflatte@ucsc.edu)

Line-integral approximations to the acoustic path integral have been used to estimate fluctuations due to internal waves. Approximations for the root-mean-square (rms) fluctuation and the bias of travel time, rms fluctuation in vertical arrival angle, and the spreading of the acoustic pulse out to 1000-km range are here compared to estimates from simulations that use the parabolic equation (PE). [See S. M. Flattié and M. D. Vera, J. Acoust. Soc. Am. 114, 697–706 (2003).] Integral-approximation (IA) estimates of rms travel-time fluctuations were within statistical uncertainty at 1000 km for the Slic89 profile, and in disagreement by between 20% and 60% for the Canonical profile. Bias estimates were accurate for the first few hundred kilometers of propagation, but often disagreed beyond. The PE structure functions of travel time with depth were quadratic for vertical separations of 20 m or less, in qualitative agreement with the IA estimates. Implications of these results will be discussed. [Work supported by ONR.] Currently at Scripps Institution of Oceanography.

8:15


Recent shallow water experiments suggest that 3-D ocean variability can produce significant intensity variations with a broad spectrum of time scales. In particular, nonlinear internal waves (NIWs) can cause acoustical ducting when the NIW and acoustic propagation directions are nearly at right angles. The sound-speed curvature induced by NIWs play an important role in controlling acoustic focusing, as shown by recent numerical simulations. However, NIWs are not perfect plane waves and an acoustic signal propagating roughly perpendicular to an NIW will encounter a fluctuating duct. We present 3-D numerical simulations in which multiple NIW packets are used to emulate the along-duct variability and its effect on acoustic intensity changes. A rough hierarchy of NIW feature characteristics, based on their intensity influence, will be described. [Work supported by ONR.]

8:30

4aUW3. Blind prediction of broadband coherence time at basinscales. John L. Spiesberger (Dept. of Earth and Environ. Sci., 240 S. 33rd St., Univ. of Pennsylvania, Philadelphia, PA 19104, johns@sas.upenn.edu), Frederick Tappert (Univ. of Miami, Miami, FL 33149), and Andrew R. Jacobson (Princeton Univ., NJ 08544)

A blind comparison with data is made with a model for the coherence time of broadband sound (133 Hz, 17 Hz bandwidth) at 3709 km. Coherence time is limited by changes in the ocean because the acoustic instruments are fixed to the Earth on the bottom of the sea with time bases maintained by atomic clocks. Although the modeled coherence time depends on the difficult problem of correctly modeling relative signal-to-noise ratios, normalized correlation coefficients of the broadband signals for the data (model) are 0.90 (0.83), 0.72 (0.59), and 0.51 (0.36) at lags of 2, 4, and 6.2 mins respectively. In all these cases, observed coherence times are a bit longer than modeled. The temporal evolution of the model is based on the linear dispersion relation for internal waves. Acoustic propagation is modeled with the parabolic approximation and the sound speed insensitive operator.

8:45


Acoustic intensity was observed to fluctuate in time and space after both one-way transmission and bistatic scattering from moored targets in the New Jersey continental shelf during the Main Acoustic Clutter Experiment (MAE) 2003 [Lai et al., 114, 2312 (2003)]. Oceanographic measurements made during the experiment also showed significant temporal and spatial variability in sound speed structure, indicating the presence of oceanic instabilities such as internal waves. Here the theoretical approach of Ratilal and Makris [114, 2428 (2003)] is applied to estimate the expected acoustic intensity given the measured statistics of the oceanographic field data. The resulting estimates span a wide range of possible values due to undersampling of the oceanographic data but are still useful in helping to understand the fundamental mechanisms responsible for the observed fluctuations.

9:00

4aUW5. Analytic mean and variance of forward propagated field through random internal waves and subbottom anomalies with Rayleigh–Born scattering. Purnima Ratilal, Tianrun Chen, and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

The mean and variance of the field propagated through a waveguide containing random internal waves or subbottom anomalies is modeled using a modal solution [Ratilal and Makris, J. Acoust. Soc. Am. 114, 2428 (2003)] that analytically expresses the effects of dispersion, attenuation,
and redistribution of modal energy due to multiple scattering in the forward direction. The scatter function of an elemental volume of inhomogeneity in density and compressibility is modeled using the Rayleigh–Born approximation. Simulations in both continental shelf and deep water waveguides quantitatively show how the relative magnitude of coherent versus incoherent intensity varies with range in typical internal wave fields and subbottom environments.

9:15
4aUW6. Frequency dependence of the sound fluctuations in the presence of internal solitary waves in SWARM'95. Mohsen Badiey (College of Marine Studies, Univ. of Delaware, Newark, DE 19716), Boris Katsev, Sergey Pereselkov (Voronezh State Univ., Voronezh 394006, Russia, katz@phys.vsu.ru), and James Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

In this paper the propagation of sound pulses in a shallow-water region in the presence of internal soliton train (IS) traveling approximately across an acoustic track of 15 km is studied. Broadband sound pulses were produced by an airgun (shots with frequency band about 30–120 Hz) and an electromechanical (J15) sound source (LFM sweeps with frequency band 50–200 Hz). Experimental data show significant correlation of sound fluctuations with IS motion. In addition, difference in the excitation source causes different modal energy partitioning in the waveguide. Modeling is carried out within the framework of the theory of vertical modes and horizontal rays (or PE in the horizontal plane). Calculations give a good agreement for temporal fluctuations of the sound energy for the sources used. It is shown that frequency dependence of intensity fluctuations is connected with the frequency dependence of refraction index of horizontal rays due to the 3D nature of these fluctuations. [Work supported by ONR and RFBR, Grant 03-05-65458.]

9:30
4aUW7. Acoustic field propagation through a time-evolving buoyant jet. Steven Finette, Roger Oba (Naval Res. Div., Naval Res. Lab., Washington, DC 20375, finette@wave.nrl.navy.mil), Patrick Gallacher, and Steve Piacsek (Naval Res. Lab., Stennis Space Ctr., MS 39529)

Previous theoretical work regarding internal wave/ acoustic wave interaction has shown that in the case of soliton (i.e., nonoverturning) propagation, significant effects on acoustic propagation occur via both amplitude and phase components. Acoustic flow visualization data obtained recently in the South China Sea (Orr and Mignerey, J. Geophys. Res. 108 (2003) indicate the presence of internal bores with associated Kelvin–Helmholtz instabilities at the base of the mixed layer. These overturning bores are examples of nonhydrostatic dynamics associated with significant vertical acceleration of the fluid. In order to explore the effect of overturning on acoustic field propagation, we have used a nonhydrostatic hydrodynamic model to simulate the temporal evolution of internal bores. The initial condition is a stationary front separating two regions characterized by slightly different but homogeneous densities. The front is released at \( t = 0 \) and the gravity-induced flow evolves into an internal bore with associated Kelvin–Helmholtz instabilities, or rotors, behind the leading edge of the bore. Results for transmission loss and signal gain degradation over a frequency range of 200–500 Hz will be presented. [Research sponsored by ONR.]

9:45

Intensities fluctuations of direct and refracted paths measured during a field test near San Clemente Island in August 2002 are presented. Signals used include 20 and 40 kHz CW pulses with 0.14- and 1.0-ms durations. Source to receiver separation was 1 km. Towed CTD data, also collected during the experiment, is presented. Acoustic signals will be analyzed using a matched filter. Temperature variations will be computed as mean-square fluctuations of the index of refraction. The intensity fluctuation records have lower scintillation indices than may be expected from the existing theory [e.g., Uscinsky et al., J. Acoust. Soc. Am. 74 (1983)]. However, our frequencies are one to two orders of magnitude higher than those used in experiments for which this theory was developed and our propagation ranges are considerably shorter. More significantly, the acoustic data records are of considerably shorter duration (~30 s) than previous experiments. To address the issue of the short record’s inability to detect larger structures in the water column, filtering is applied to the sound speed field before the calculation of the correlation length (patch size). [Work supported by ONR Code 321US.]

10:00–10:15 Break

10:15
4aUW9. Source localization and inversion in the presence of internal waves. Ralph N. Baer and Michael D. Collins (Naval Res. Lab., Washington, DC 20375, baer@nrl.navy.mil)

It is often possible to localize an acoustic source in a medium with environmental uncertainties by including environmental parameters in the search space [J. Acoust. Soc. Am. 90, 1410–1422 (1991)]. This approach has recently been applied to problems involving stochastic environmental parameters [J. Acoust. Soc. Am. 114, 2401 (2003)]. The parameters of the Garrett–Munk spectrum were held fixed, and the source was localized by generating replica fields for various realizations of the internal wave field. In the present work, source localization simulations are performed with the parameters of the Garrett–Munk spectrum included in the parameter space. The possibility of solving an inverse problem to estimate the internal wave parameters is also considered. [Work supported by the ONR.]

10:30
4aUW10. Sea surface effects on reverberation vertical coherence and inverted bottom acoustic parameters in the East China Sea. Ji-Xun Zhou, Xue-Zhen Zhang (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332), Peter H. Dahl, and Jeffrey A. Simmen (Univ. of Washington, Seattle, WA 98105)

Wideband reverberation measurements were repeatedly conducted at the center of the ASAEX site in the East China Sea, on 3 and 5 June, 2001. Between these two measurement days, wind speed and rms wave height changed significantly, going from about 1 m/s to 0.1 m to 10 m/s and 0.35 m, respectively. This paper will compare reverberation vertical coherence (RVC) and RVC-inverted equivalent bottom acoustic parameters in frequency range of 100–1200 Hz for the two measurements [J. X. Zhou and X. Z. Zhang, J. Acoust. Soc. Am. 113, 2204 (2003)]. The difference of equivalent bottom reflection losses, obtained from two measurements, is well explained by the supporting sea surface data. The results show that an inversion of seabottom acoustic parameters from shallow-water long-range reverberation (or sound propagation) should take the surface condition into account, especially for higher sea states and higher frequencies. [Work supported by ONR and NNSF of China.]

10:45

Signals akin to head waves were obtained from explosive sources during the joint U.S. and China Yellow Sea experiment, conducted in August 1996. The sources were deployed at ranges from 0.7 to 37 km in waters approximately 75 m deep. Data were recorded on a 16-element vertical line array with element spacing of 4 m and the deepest element at depth 66 m. This talk will focus on measurements obtained at source ranges less than 1 km, because the amplitude of the head wave arrivals decreases rapidly with range. The arrival time and power spectra were first used to identify and distinguish ground-, water-, and head-wave arrival types. This
latter category was then further studied in the context of pure head waves, noninterfering head waves, and interference head waves. The experimental data are also compared with simulated signals obtained via a Fourier synthesis of a narrow-band complex parabolic equation (PE) field, using the RAM PE code. The overall influence of compressional velocity, velocity gradient, and attenuation in the sediment, and frequency dependency of head wave arrivals are discussed. [Research supported by ONR Ocean Acoustics.]

11:00

4aUW12. Evidence for nonlinear frequency dependence of attenuation in an East China Sea environment. David P. Knobles, Robert A. Koch (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029), James H. Miller, and Gopu R. Potty (Univ. of Rhode Island, Narragansett, RI 02882)

Acoustic data analyzed from two locations in the central East China Sea indicate a nonlinear frequency dependence of the sediment attenuation. The acoustic measurements are in the form of transmission loss (TL) versus range for frequencies in the 50–800-Hz band and time series generated by explosive sources recorded on a vertical line array (VLA). The seabed can be characterized by a sediment that has a significant sand content. It is discovered that a frequency dependence of $f^n$, with $n$ at least as large as 1.8, is needed to explain both an optimal propagation effect of the TL at the lower frequencies and the temporal spread of the time series measured on the VLA. A linear frequency dependence generates too much loss at the lower frequencies than the reported measurements and impulse responses that have too small a time spread. Complicating features of the inference of the frequency dependence of the attenuation from forward propagation measurements include the deduction of the source spectrum for individual shots, the source depth, and geoacoustic layering structures that are capable of explaining both measured TL and the time series.

11:15

4aUW13. Low-frequency reverberation signal in shallow water in presence of internal waves. Boris Katsnelson, Sergey Pereselkov (Voronezh Univ., 1, Universitetskaya Sq., Voronezh 394006, Russia, pereselkov@hotmail.com), and Valery Petnikov (General Phys. Inst., Moscow 113000, Russia)

The reverberation of low-frequency (100–400 Hz) sound in shallow water is considered. It is supposed that reverberation signals are formed by summarizing from circle area (radius about 50 km) as result of sound field backscattering on bottom surface. The developed model of reverberation takes into account mode coupling caused by background internal waves in shallow water and depending on direction of backscattered signal. Influence of internal waves on space-time structure of reverberation signal is considered for two cases: single-modal and multi-modal. The computer simulation shows that internal waves lead to significant temporal fluctuations of reverberation signal in both cases. The results of computer simulation are compared with data of experimental research in the Barents Sea. [Work supported by RFBR, Grants 02-02-16509 and 03-05-64568.]

11:30

4aUW14. Environmental uncertainty in modeling of shallow-water reverberation. T. W. Yudichak and D. P. Knobles (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

Simulated monostatic shallow-water reverberation is produced by a broadband model that accounts for propagation and scattering in a waveguide that is horizontally stratified on average, but subject to slightly rough interfaces and small, random spatial fluctuations in density and sound speed. This model has been used to infer parameters characterizing these inhomogeneities (scattering parameters) from experimental data, assuming a specific background geoacoustic profile. In realistic situations, however, the background environment may either not be known with precision or may be too variable to admit a range-independent characterization. It is even possible that several geoacoustically distinct alternative sets of environmental parameters are plausible. This may occur, for instance, when geoacoustic inversion is used to determine a background profile. In this presentation, the effects of environmental uncertainties, including the existence of alternative geoacoustic profiles, on scattering parameters inferred from reverberation data are examined. [Work supported by ONR.]
Session 4pAA

Architectural Acoustics: Alternative Acoustic Environments for Performing Arts Presentations

Christopher Jaffe, Cochair
Jaffe Holden Acoustics, Inc., 114 A Washington Street, Norwalk, Connecticut 06854

Neil Rolnik, Cochair
Jaffe Holden Acoustics, Inc., 114 A Washington Street, Norwalk, Connecticut 06854

Chair’s Introduction—2:45

Invited Papers

2:50

4pAA1. Engaging spaces: Intimate electro-acoustic display in alternative performance venues. Curtis Bahn and Stephan Moore (Arts Dept./EAR Studios, Rensselaer Polytechnic Inst., 110 8th St., Troy, New York, NY 12180, crb@rpi.edu)

In past presentations to the ASA, we have described the design and construction of four generations of unique spherical speakers (multichannel, outward-radiating geodesic speaker arrays) and Sensor-Speaker-Arrays, (SenSAs: combinations of various sensor devices with outward-radiating multichannel speaker arrays). This presentation will detail the ways in which arrays of these speakers have been employed in alternative performance venues—providing presence and intimacy in the performance of electro-acoustic chamber music and sound installation, while engaging natural and unique acoustical qualities of various locations. We will present documentation of the use of multichannel sonic diffusion arrays in small clubs, “black-box” theaters, planetariums, and art galleries.

3:10

4pAA2. Alternative acoustic environments for the generation of reverberation. Alexander U. Case (fermata audio + acoustics, P.O. Box 1161, Portsmouth, NH 03802)

The musicians and engineers who create popular recorded music view reverberation as a signal processing effect to be added to any and all elements of a multitrack production. Devices such as digital reverbs, spring reverbs, and plate reverbs are tools of the recording trade, synthesizing reverberlike sounds for performance through loudspeakers. Acoustic reverberation makes its way into recorded music through the use of a reverb chamber. A small room is used to generate reverb. With cubic volume well below that of a performance hall, it works the “other side” of the Sabine equation, being built of highly sound reflective materials. A purpose-built room for the generation of reverb is a luxury not many studios can afford. Clever use of stairwells, bathrooms, and basements is easier on the recording studios balance sheet. This work evaluates the repurposing of these alternative spaces for the generation of reverb in popular recorded music.

3:30

4pAA3. Not your grandfather’s concert hall. Russell Cooper (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854, rcooper@jhacoustics.com), Richard Malenka (Carnegie Hall, New York, NY 10019-3210), Charles Griffith (Polish Partnership, New York, NY 10014), and Steven Friedlander (Auerbach, Pollack, Friedlander, New York, NY 10018-5509)

The opening of Judy and Arthur Zankel Hall on 12 September 2003, restores Andrew Carnegie’s original 1891 concept of having three outstanding auditoriums of different sizes under one roof, and creates a 21st-century venue for music performance and education. With concerts ranging from early music to avant-garde multimedia productions, from jazz to world music, and from solo recitals to chamber music, Zankel Hall expands the breadth and depth of Carnegie Hall’s offerings. It allows for the integration of programming across three halls with minifestivals tailored both to the size and strengths of each hall and to the artists and music to be performed. The new flexible space also provides Carnegie Hall with an education center equipped with advanced communications technology. This paper discusses the unique program planned for this facility and how the architects, theatre consultants, and acousticians developed a design that fulfilled the client’s expectations and coordinated the construction of the facility under the floor of the main Isaac Stern Auditorium without having to cancel a single performance.

3:50

4pAA4. A composer’s eye for room acoustics. Alvin Lucier (Dept. of Music, Wesleyan Univ., 42 Pinewood Terrace, Middletown, CT 06457, alucier@wesleyan.edu)

The physical acoustician has a number of techniques that can be used to document the acoustic signature of an enclosed space. A composer may approach this matter from a different perspective. This paper discusses and sonically illustrates several physical acoustic aural approaches to developing the signature of a room using unamplified speech and music as source signals. Other methods involving multiple echograms and natural impulse sources are also described. Portions of a score composed for five instruments will be played. In this piece, parts were written for four acoustic instruments as well as for the room in which the performance was presented.
4pAA5. Open secrets. Malcolm Holzman (Hardy Holzman Pfeiffer & Assoc., 902 Broadway, New York, NY 10010, mholzman@hhpa.com), Eve Beglarian (EVBVD Music, New York, NY 10011), and Curtis Bahn (Rensselaer Polytechnic Inst., Troy, NY 12180-3590)

In 1998, the Schools of Architecture and Humanities at the Rensselaer Polytechnic Institute initiated a collaborative project between students of both schools to develop a musical performance piece that would be presented in an environment to be designed and built in the Art Gallery of the School of Architecture. Students from both Schools were to work together on developing the content and conceptualizing the built environment. Architects Malcolm Holzman and Nestor Bottino of the firm of HHPA were brought in as advisors to the project, as was Composer Eve Beglarian of Twisted Tutu. Curtis Bahn, Director of the iEAR Studio at RPI, was Academic Coordinator for the project, together with Beth Weinstein, Professor, School of Architecture, and Grethe Holby, who directed the performance. This paper will review the academic raison d’etre for organizing such a program, discuss the history of the project as it developed, and present a video tape of portions of the finished performance.

THURSDAY AFTERNOON, 27 MAY 2004

Session 4pAB

Animal Bioacoustics: Infrasonic Communication by Animals: Signal Propagation, Generation, Reception and Function

Edmund Gerstein, Cochair
Leviathan Legacy, Inc., 1318 S.W. 14th Street, Boca Raton, Florida 33486

Roger S. Payne, Cochair
Ocean Alliance, 191 Weston Road, Lincoln, Massachusetts 01773

Chair’s Introduction—1:00

Invited Papers

1:05

4pAB1. Infrasound propagation and coupling between air, soil, and water. Henry Bass, Carrick Talmadge, Craig Hickey (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, Univ., MS 38677, pabass@olemiss.edu), and Francine Desharnais (DRDC Atlantic, Dartmouth, NS, Canada)

Propagation of low-frequency sound is enhanced by very low absorption in air, water, and soil. Some examples of absorption expected in air, water, and soil are presented. Reduction in sound level between source and receiver is typically the result of spreading. In all media of interest, layering can give rise to trapped waves that decay as cylindrical waves as opposed to spherical waves. Examples of the effect of layering on propagation in the atmosphere will be given for day/night conditions. At very low frequency, scattering by rough terrain, moderate size turbulence, and waves have a reduced effect on signal amplitude and phase. Turbulence carried by the wind can result in pressure variations at the receiver that can sound like acoustic signals. An effective receiving system must deal with that wind noise. Identifying the mechanism for communications at low frequency is complicated by the exchange of acoustic energy between air, soil, and water. Some characteristics of this coupling and losses that might be encountered along different paths will be discussed.

1:25

4pAB2. Eavesdropping on elephants. Katy Payne (Bioacoust. Res. Prog., Cornell Lab of Ornithology, 159 Sapsucker Woods Rd., Ithaca, NY 14850, kp17@cornell.edu)

The Elephant Listening Project is creating an acoustic monitoring program for African forest elephants, an endangered species that lives in dense forests where visual censusing is impossible. In 2002, a 2 ½-month continuous recording was made on an array of autonomous recording units (ARUs) surrounding a forest clearing in the Central African Republic. Each day between 10 and 160 forest elephants (Loxodonta cyclotis), the subjects of Andrea Turkalo’s 13-year demographic study, were present on the clearing.
Thousands of vocalizations were recorded, most of which contained infrasonic energy. The calls were located in space using software developed in the Bioacoustics Research Program. During daytime hours simultaneous video recordings were made. GPS time-synchronization of video cameras and the ARUs made it possible to identify the elephants responsible for many calls and to examine associated circumstances and behaviors. Recordings were also made on a second acoustic array, permitting a preliminary estimate of propagation and an indication of source level for selected elephant calls. Automatic detection of elephant calls is increasing the feasibility of analyzing long acoustic recordings, and paving the way for finer-tuned analyses, with an ultimate goal of describing forest elephants’ acoustic repertoire.

1:45

4pAB3. Elephant low-frequency vocalizations propagate in the ground and seismic playbacks of these vocalizations are detectable by wild African elephants (*Loxodonta africana*). Caitlin E. O’Connell-Rodwell, Jason D. Wood, Roland Gunther, Simon Klemperer (Dept. of Geophysics, Stanford Univ., Stanford, CA 94305, ceoconnell@stanford.edu), Timothy C. Rodwell, Sunil Puria, Robert Sapolsky (Stanford Univ., Stanford, CA 94305), Colleen Kinzley (Oakland Zoo), Byron T. Arnason (Tezar, Inc., Austin, TX 76235), and Lynette A. Hart (Univ. of California, Davis, CA 95616)

Seismic correlates of low-frequency vocalizations in African and Asian elephants propagate in the ground at different velocities, with the potential of traveling farther than their airborne counterparts. A semblance technique applied to linear moveouts on narrow-bandpass-filtered data, coupled with forward modeling, demonstrates that the complex waves observed are the interference of an air wave and a Rayleigh wave traveling at the appropriate velocities. The Rayleigh wave appears to be generated at or close to the elephant, either by coupling through the elephant’s body or through the air near the ground. Low-frequency elephant vocalizations were reproduced seismically and played back to both a captive elephant and to elephant breeding herds in the wild, monitoring the elephants’ behavioral responses, spacing between herd members and time spent at the water hole as an index of heightened vigilance. Breeding herds detected and responded appropriately to seismically transmitted elephant warning calls. The captive studies promise to elucidate a vibrotactile threshold of sensitivity for the elephant foot. Elephants may benefit from the exploitation of seismic cues as an additional communication modality, thus expanding their signaling repertoire and extending their range of potential communication and eavesdropping beyond that possible with airborne sound.

2:05

4pAB4. The role of infrasounds in maintaining whale herds. Roger S. Payne (Ocean Alliance, 191 Weston Rd., Lincoln, MA 01773)

For whales and dolphins a basic social unit is the herd. In several species, herds have been observed to maintain the same speed, direction, and membership overnight, and while swimming in waters of near-zero visibility—evidence that individuals can stay together using nonvisual cues. The most likely such cue is sound. If whale herds are held together with sound, yet we define herds as groups of whales seen moving together, then we are using visual criteria to judge what is an acoustic phenomenon, and our conclusions about a most basic unit of cetacean social structure, the herd, are at least incomplete, and, quite possibly, worthless. By calling herds, heards, we remind ourselves that sound controls herd size. We then consider that some whale infrasound can propagate across deep water at useful intensities (even in today’s ship-noise-polluted ocean) for thousands of kilometers. The distance to which blue and fin whale sounds propagate before falling below background noise is given, and the possible advantages these whales obtain from such sounds is explored. The conclusion is that by sharing information on food finds infrasonically, fin and blue whales may have developed a way to divide up the food resources of an entire ocean.

2:25

4pAB5. Baleen whale infrasonic sounds: Natural variability and function. Christopher W. Clark (Lab. of Ornithology, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, cwc2@cornell.edu)

Blue and fin whales (*Balaenoptera musculus* and *B. physalus*) produce very intense, long, patterned sequences of infrasonic sounds. The acoustic characteristics of these sounds suggest strong selection for signals optimized for very long-range propagation in the deep ocean as first hypothesized by Payne and Webb in 1971. This hypothesis has been partially validated by very long-range detections using hydrophone arrays in deep water. Humpback songs recorded in deep water contain units in the 20–100 Hz range, and these relatively simple song components are detectable out to many hundreds of miles. The mid-winter peak in the occurrence of 20-Hz fin whale sounds led Watkins to hypothesize a reproductive function similar to humpback (*Megaptera novaeangliae*) song, and by default this function has been extended to blue whale songs. More recent evidence shows that blue and fin whales produce infrasonic calls in high latitudes during the feeding season, and that singing is associated with areas of high productivity where females congregate to feed. Acoustic sampling over broad spatial and temporal scales for baleen species is revealing higher geographic and seasonal variability in the low-frequency vocal behaviors than previously reported, suggesting that present explanations for baleen whale sounds are too simplistic.

2:45

4pAB6. Do manatees utilize infrasonic communication or detection? Edmund Gerstein (Florida Atlantic Univ., Charles E. Schmidt College of Sci., 777 Glades Rd., Boca Raton, FL 33431 and Leviathan Legacy, Inc., Boca Raton, FL 33486), Laura Gerstein (Leviathan Legacy, Inc., Boca Raton, FL 33486), Steve Forsythe (Naval Undersea Warfare Ctr., Newport, RI 02841), and Joseph Blue (Leviathan Legacy, Inc., FL 32806)

Some researchers speculate Sirenians might utilize infrasonic communication like their distant elephant cousins; however, audiogram measurements and calibrated manatee vocalizations do not support this contention. A comprehensive series of hearing tests conducted with West Indian manatees yielded the first and most definitive audiogram for any Sirenian. The manatee hearing tests were also the first controlled underwater infrasonic psychometric tests with any marine mammal. Auditory thresholds were measured from 0.4 to 46 kHz, but detection thresholds of possible vibrotactile origin were measured as low as 0.015 kHz. Manatees have short hairs
on their bodies that may be sensitive vibrotactile receptors capable of detecting particle displacement in the near field. To detect these signals the manatee rotated on axis, exposing the densest portion of hairs toward the projector. Manatees inhabit shallow water where particle motion detection may be more useful near the water’s surface, where sound pressures are low due to the Lloyd mirror effect.

With respect to intraspecific communication, no infrasonic spectra have been identified in hundreds of calibrated calls. Low source levels and propagation limits in shallow-water habitats suggest low-frequency manatee calls have limited utility over long distances and infrasonic communication is not an attribute shared with elephants.

3:05–3:20 Break

3:20

4pAB7. Low-frequency sounds and amphibious communication in *Hippopotamus amphibious*. William E. Barklow (Framingham State College, Framingham, MA 01760)

Hippos make sounds in both air and underwater, and, with their heads in an amphibious position (eyes and nostrils above water but mouth and throat below), are able to transmit sounds to both media simultaneously. Hippos on the surface respond to the surface component by calling. Hippos underwater consistently surface and call in a chorus that can spread in air from one territory to the next for many kilometers. They produce several low-frequency, high-amplitude (100 dB re: 20 μPa) sounds. The grunt, their most common call, has a 30- to 60-Hz fundamental, and the huff and some tonal sounds end with an abrupt drop in frequency to 20- to 30-Hz. These sounds are usually given amphibiously, but the high-pass filter characteristics of shallow water attenuates the low frequencies of the underwater component. Hippos also emit these and other sounds when they are completely submerged. These are inaudible in air, but they produce a fountain on the surface accompanied by a 10- to 20-Hz sound. They also produce this effect with plosive blows underwater without other sounds. Similar “bubble blasts” have been reported in gray whales. The function of these sounds is not clear, but they may facilitate long-distance “chain chorusing.”

3:40

4pAB8. Ground sounds: Seismic detection in the golden mole. Peter M. Narins (Dept. of Physiol. Sci., UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095-1606, pnarins@ucla.edu) and Edwin R. Lewis (Univ. of California, Berkeley, CA 94720)

The Namib Desert golden mole is a nocturnal, surface-foraging mammal, possessing a massively hypertrophied malleus which presumably confers low-frequency, substrate-vibration sensitivity through inertial bone conduction. Foraging trails are punctuated with characteristic sand disturbances in which the animal’s head dips under the sand. The function of this behavior is not known but it is thought that it may be used to obtain a seismic fix on the next mound to be visited. To test this, we measured the local seismic vibrations both on the top of a mound and on the flats. The spectrum recorded on the flats shows a relatively low-amplitude peak at about 120 Hz, whereas the spectral peak recorded from the mound is nearly 17 dB greater in amplitude and centered at 310 Hz. This suggests that mounds act as seismic beacons for the golden moles that would be detectable from distances corresponding to typical intermound distances of 20–25 m. In addition, out of the 117 species for which data are available, these golden moles have the greatest ossicular mass relative to body size (Mason, personal communication). Functionally, they appear to be low-frequency specialists, and it is likely that golden moles hear through substrate conduction. [Work supported by NIH.]

Contributed Papers

4:00

4pAB9. Interactive patterns of vocal communication in African elephant herds (*Loxodonta africana*). Caitlin E. O’Connell-Rodwell (Dept. of Geophys., Bio-X Program, Stanford Univ., Clark Bldg. E-150, 318 Campus Dr., Stanford, CA 94305-5437, cococonnell@stanford.edu), Megan T. Wyman, Lynette A. Hart, and Shay Redfield (Univ. of California, Davis, CA 95616)

This study examines the interactive nature of vocalizations produced within African elephant herds during waterhole visits at Etosha National Park, Namibia. The temporal organization and physical characteristics of 1025 vocalizations, documented within 14.8 h of acoustic field recordings, were analyzed using a variety of statistical tests. Temporal distribution analyses indicated that calls were clumped in time (Kolgoror–Smirnov, average significant \( p = 0.00158 \)) and the majority of the calls occurred after herds began the process of departing a waterhole (Wilcoxon signed rank, \( p = 0.0012 \)). In addition, 84% of all calls began within 30 s of another call. Based on the analysis of the temporal patterning of calls, a conversational series of vocalizations was defined as calls separated by less than 30 s of silence. The majority of calls in a conversational series are overlapping or contiguous, indicating that multiple elephants are involved in the exchange. A variety of physical characteristics (duration, frequency, frequency modulation, and frequency at peak amplitude) was compared across three different call types. These conversational call patterns may function to maintain intragroup cohesion and coordination as well as intergroup sharing and competition reduction. [Work supported by: Namibia Nature Foundation, Etosha Ecological Institute, UC Davis Block, Jastro Shields, and Faculty Research Grants.]

4:15

4pAB10. How many rumbles are there? Acoustic variation and individual identity in the rumble vocalizations of African elephants (*Loxodonta africana*). Joseph M. Solits, Anne Savage (Animal Programs, Disney’s Animal Kingdom, P.O. Box 10000, Lake Buena Vista, FL 32830, Joseph.solits@disney.com), and Kirsten M. Leong (Cornell Univ., Ithaca, NY)

The most commonly occurring elephant vocalization is the rumble, a frequency-modulated call with infrasonic components. Upwards of ten distinct rumble subtypes have been proposed, but little quantitative work on the acoustic properties of rumbles has been conducted. Rumble vocalizations \((N = 269)\) from six females housed at Disney’s Animal Kingdom were analyzed. Vocalizations were recorded from microphones in collars around subject necks, and rumbles were digitized and measured using SIGNAL software. Sixteen acoustic variables were measured for each call, extracting both source and filter features. Multidimensional scaling analysis indicates that there are no acoustically distinct rumble subtypes, but that there is quantitative variation across rumbles. Discriminant function analysis showed that the acoustic characteristics of rumbles differ across females. A classification success rate of 65% was achieved when assigning unselected rumbles to one of the six females \((\text{test set} = 64 \text{ calls})\) according to the functions derived from the originally selected calls \((\text{training set} = 205 \text{ calls})\). The rumble is best viewed as a single call type with graded variation, but information regarding individual identity is encoded in female rumbles.
Cetaceans produce sounds at opposite ends of the frequency spectrum. The laryngeal role in odontocete sound production (echolocation, communication) remains unclear. Mysticete infrasonic are presumed to be laryngeal in origin, but production mechanisms are unknown. To address this, we examined postmortem larynges in 6 mysticete species (3 genera) and compared them to our odontocete collection (20 species/15 genera). Results indicate that the rostral portion of the odontocete larynx is elongated, narrow, rigid, and normally positioned intranarially. This portion of the mysticete larynx is comparatively shortened, open, pliable, and in Megaptera may be retracted from its intranarial position. Internally, mysticete vocal folds are thick, paired, and oriented horizontally, compared with the thin, usually unpaired, and vertically oriented odontocete fold. Mysticetes may generate low frequency sounds via pneumatically driven fold vibrations, which then pass to attached laryngeal sac walls, through overlying throat pleats, to water. Rorqual mysticetes may also vibrate paired corniculate flaps while regulating airflow into the nasal region. Infrasonic pulses may pass through adjacent soft palate, skull, or nasal cartilages to water. Laryngeal anatomy in mysticetes and odontocetes appears highly divergent. These morphological differences may correlate to adaptations for producing infrasonic (mysticete) or ultrasonic (odontocete) communication. [Work supported by ONR: N00014-96-1-0764, ONR: N00014-99-1-0815, and AMNHSOF].

4:30
4pAB11. Anatomy of infrasonic communication in baleen whales: Divergent mechanisms of sound generation in mysticetes and odontocetes. Joy S. Reidenberg and Jeffrey T. Laitman (Ctr. for Anatomy and Functional Morphology, Mail Box 1007, Mount Sinai School of Medicine, New York, NY 10029-6574, joy.reidenberg@mssm.edu)

4:45
4pAB12. Spectrogram analysis of low to mid frequency marine mammal clicks. George E. Ioup, Juliette W. Ioup, James P. Larue (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148), Natalia A. Sidorovskaia (Univ. of Louisiana at Lafayette, Lafayette, LA 70504-4210), Stan A. Kuczaj, Grayson H. Rayborn, and Christopher D. Walker (Univ. of Southern Mississippi, Hattiesburg, MS 39406)

Previous investigators have proposed explanations for some sperm whale click structure and pointed out that the separation of individual pulses within the click might be used to determine approximately the size of the sperm whales. Recently, Mohl et al. [J. Acoust. Soc. Am. 114, 1124–1154 (2003)] have shown that echo-location click structure is highly dependent on the received angle. In data measured by the Littoral Acoustic Demonstration Center using bottom-moored hydrophones in the northern Gulf of Mexico in the summers of 2001 and 2002, rich click structures were observed in the spectrograms of many click trains, some of which exhibit strikingly consistent spectral nulls across the train. Although this structure in the spectra could be due to propagation effects, investigations to date suggest this possibility is highly unlikely, as discussed in the next abstract. Therefore it is at least plausible that the structure could be used to identify individual animals. This is known to be a difficult problem in the case of sperm whales because of the angle dependence of at least some of their clicks. These difficulties are discussed, as is the possible use of the spectrograms of the clicks to identify individuals. [Research supported by ONR.]
A collaborative investigation of midwater zooplankton aggregations in a coastal fjord was conducted in November 2002. Midwater aggregations of zooplankton in a coastal fjord were sampled and mapped using a calibrated, three-frequency (38, 120, and 200 kHz) vessel-based echo-sounder system, a multinet towed zooplankton net (BIONSESS), and a high-resolution in situ camera system (ZOOVIS). Dense daytime layers of euphausiids and amphipods near 70- to 90-m depth were found in the lower reaches of the inlet, especially concentrated by tidal flows around a sill which rises above the layer. Quantitative euphausiid and amphipod backscattering measurements, combined with in situ species, size, and abundance estimates, were found to agree closely with recent size- and orientation-averaged fluid–cylinder scattering models produced by Stanton et al. Also, in situ scattering measurements of physonect siphonophores were found to have a much stronger low-frequency (38 kHz) scattering strength, in agreement with a simple bubble scattering model. [Work supported by Dr. J. Eckman, ONR code 322BC.]

Coral reefs serve as the habitat for demersal mesozooplankton and small fishes that migrate into overlying waters at night but spend daylight hours within the reef in part because that habitat provides protection against visual predators. These movements structure energy, mass, and nutrient exchange between the reef habitat and the surrounding waters. Information to date has, however, been predominately qualitative and has not taken advantage of recent advances in biological oceanographic sampling instrumentation. Moreover, what sampling has been done has not been rigorously coupled to synoptic time series of ambient oceanographic conditions. To begin to fill this critical gap we deployed for two 1-month-periods multi-frequency acoustic (TAPS) and optical integrated environmental sensor packages to continuously measure and record the abundance and size distribution of organisms while concomitantly measuring water column chlorophyll, fluorescence, transmittance, temperature, and salinity. During the two periods of high-resolution sampling, and in the intervening six months, we also deployed bottom-mounted ADCP units yielding both vertical current structure and backscatter amplitude distributions. Both video footage and traditional net or pump samples were obtained for ground-truth purposes. [Support provided by the National Marine Fisheries Service/Southeast Fisheries Science Center.]

The seasonal, and in particular, winter distribution of Calanus finmarchicus and its predators were investigated using multifrequency acoustics in the Irminger Sea, North Atlantic, as part of Marine Productivity, a UK contribution to GLOBEC. The distribution of Calanus, which over-winters deeper than 500 m, well below the maximum over-wintering Calanus abundances. Calanus predators, euphausiids, e.g., Meganyctiphanes norvegica and Myctophids, were observed within the Irminger Sea, which suggests that the Calanus predators may be deepening their winter depth to coincide with the maximum over-wintering Calanus abundances.
During a collaborative investigation of zooplankton aggregations near a coastal fjord (Knight Inlet, British Columbia), surveys were conducted using down-looking echosounders and a digital imaging system (ZOOVIS). Two surveys conducted at night revealed that the presence or absence of external illumination on the vessel had a pronounced influence on measured acoustical scattering. The extremely short latency between shifts in illumination and changes in acoustical scattering suggested that differences in the orientations of scatterers were responsible for this phenomenon. High-resolution, in situ images of euphausiids (Euphausia pacifica) from ZOOVIS indicated that these organisms were present in a range of orientations ranging from horizontal to vertical, relative to the incident acoustical beam. Theoretical scattering models based on digitizations of in situ ZOOVIS images suggest that the magnitude of the observed changes in scattering, in response to altered illumination, may be accounted for by differences in euphausiids orientation. [Research supported by the ONR, Code 322BC.]

**Behavioral observations of in situ copepods with a multibeam sonar.** Jules Jaffe (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA 92039-0238, jules@mpl.ucsd.edu), Amatzia Genin, and Moti Ohevia (Hebrew Univ., Eilat 88103, Israel)

In order to count and track copepods in three dimensions a multibeam sonar has been developed which operates at a frequency of 1.6 MHz. The sonar uses an analog version of a crossed array in order to localize animals in 2–4-liter volumes. The system uses an 8 × 8 beam configuration to localize animals at subcentimeter accuracy in range and centimeter accuracy across track. Sensitivity was measured in a test tank via the injection of animals, and it was found that the system was capable of measuring reflections of animals as small as 1–2 mm at ranges of at least 2 m. In order to measure the behavior of the animals under different environmental conditions in situ, experiments were conducted in the Gulf of Elat. Here, the sonar was mounted on a tripod and aimed horizontally. Processed results yielded over 200 000 individual tracks that lasted from 2 to 40 s. The results demonstrated that zooplankton retain their depth under up- and down-welling currents of 1–2 cm/s (as measured with injected fluorescein dye). This is unlike their response in the horizontal, where the animals are passively swept with the currents.

2:30–2:45 Break

**Multifrequency analyses of fish distributions in the northwest Atlantic.** J. Michael Jech (NEFSC, 166 Water St., Woods Hole, MA 02543, michael.jech@noaa.gov)

Routine acoustical surveys for estimating Atlantic herring (Clupea harengus) population abundance have been conducted on Georges Bank during the autumn spawning season from 1998 to present. Acoustical data are collected with a Simrad EK500 scientific echo sounder operating at 12 or 18, 38, and 120 kHz, and split-beam (the 12-kHz system is a single beam) transducers. Biological measurements and verification of acoustical scatterers are obtained with a pelagic trawl. Acoustical data are evaluated (scrutinized) manually to remove noise, faulty bottom detections, and to classify acoustical backscattering to species. Species classification is currently subjective, and is based on the experience of the scientists and trawl catches. Objective species classification and automated fish density and abundance estimates are an obvious goal for fisheries surveys using advanced technologies. Classification methods using relationships among frequency-dependent volume backscattering strengths, such as presence–absence and combination–permutation, are described and presented. Results indicate that while classification using these methods and acoustical information alone is not robust, these methods highlight backscattering patterns within aggregations and have the potential to characterize backscattering patterns observed in fisheries acoustics data. [Work supported by NOAA Fisheries and ONR.]

**The acoustic environment of the Florida manatee: Correlation with level of habitat use.** Jennifer L. Miksis-Olds (Univ. of Rhode Island, Grad. School of Oceanogr., Narragansett, RI 02882, jmiksis@geo.uri.edu), James H. Miller (Univ. of Rhode Island, Narragansett, RI 02882), and Peter L. Tyack (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

The Florida manatee is regularly exposed to high volumes of vessel traffic and other human-related noise pollutants because of their coastal distribution. Quantifying specific aspects of the manatees' acoustic environment will allow for a better understanding of how these animals are responding to both natural and human induced changes in their environment. Acoustic recordings and transmission loss measurements were made in two critical manatee habitats: seagrass beds and dredged basins. Twenty-four sampling sites were chosen based on the frequency of manatee presence in specific areas from 2000–2003. Recordings were composed of both ambient noise levels and transient noise sources. The Monterey-Miami Parabolic Equation Model (MMPE) was used to relate environmental parameters to transmission loss, and model outputs were verified by field tests at all sites. Preliminary results indicate that high-use grassbeds have higher levels of transmission loss compared to low-use sites. Additionally, high-use grassbeds have lower ambient noise in the early morning and later afternoon hours compared to low-use grassbeds. The application of noise measurements and model results can now be used to predict received levels, signal-to-noise ratios, and reliable detection of biologically relevant signals in manatee habitats and in the many different environments that marine mammals live.

3:00

**Killer whale caller localization using a hydrophone array in an oceanarium pool.** Ann E. Bowles (Hubbs-SeaWorld Res. Inst., 2595 Ingraham St., San Diego, CA 92109, annb1@san.rr.com), Charles F. Greenlaw, Duncan E. McGhee, and D. Van Holliday (BAE Systems, San Diego, CA 92123)

A system to localize calling killer whales was designed around a ten-hydrophone array in a pool at SeaWorld San Diego. The array consisted of nine IFCC 3212 and one ITT 6050H hydrophones mounted in recessed 30×30 cm² niches. Eight of the hydrophones were connected to a Compaq Armada E500 laptop computer through a National Instruments DAQ 6024E PCMCIA A/D data acquisition card and a BNC-2120 signal conditioner. The system was calibrated with a 139-dB, 4.5-kHz pinger. Acoustic data were collected during four 48–72 h recording sessions, simultaneously with video recorded from a four-camera array. Calling whales were localized by one of two methods, (1) at the hydrophone reporting the highest sound exposure level and (2) using custom-designed 3-D localization software based on time-of-arrival (ORCA). Complex reverberations in the niches and pool made locations based on time of arrival difficult to collect. Based on preliminary analysis of data from four sessions (400+ calls/session), the hydrophone reporting the highest level reliably attributed callers 51%–100% of the time. This represents a substantial improvement over the attribution rates of 5%–15% obtained with single hydrophone recordings. [Funding provided by Hubbs-SeaWorld Research Institute and the Hubbs Society.]
3:30

4pAO10. Contribution of active and passive acoustics to study oceanographic processes feeding whales in a critical habitat of the St. Lawrence Estuary. Yvan Simard, Nathalie Roy (Maurice Lamontagne Inst., Fisheries and Oceans Canada, 850 route de la Mer, Mont-Joli, QC G5H 3Z4, Canada), Yvan Simard, and Cédric Cotte (Univ. du Québec a Rimouski, Rimouski, QC G5L 3A1, Canada, Yvan_Simard@uqar.qc.ca)

The head of the main channel of the continent in eastern Canada is the site of particular oceanographic processes that are responsible for the creation of a persistent feeding ground regularly visited by baleen whales from the Atlantic for centuries. Multifrequency acoustics coupled with ADCP and hydrographic measurements has been used to map the krill and capelin aggregations in 3D and visualize their local concentration process under tidal forcing and upwelling at the channel head. The krill scattering layers, pumped into the area by the strong two-layer estuarine circulation, appear to be concentrated during flood by tidal currents forced against the slopes and upwelling, to which depth-keeping krill is reacting by swimming down. Capelin also tends to concentrate on slopes and neighboring shallows. This highly recurrent process generates rich patches that are contributing with the mean circulation to make this area the richest krill aggregation in Northwest Atlantic. This critical habitat is located in a major continental seaway. Passive acoustics techniques are exploited to locate whale calls and map the use of this area in continuing months, especially by blue and fin whales, with the aim of understanding their movements to improve their protection.

3:45

4pAO11. Sonar off-axis target classification by an echolocating dolphin. Patrick Moore (Space and Naval Warfare Systems Ctr. San Diego, 53560 Hull St. Code 2351, San Diego, CA 92152-5001), Lois Dankiewicz (SAIC BioSolutions, San Diego, CA 92110), and Dorian Houser (BIOMIMETICA, La Mesa, CA 91942)

Dolphin echolocation has evolved over millions of years under selection pressures imposed by a selective niche. The complexity and effectiveness of dolphin echolocation for detection and classification of objects within that niche has useful application to U. S. Naval objectives. In these environments, Navy dolphins are likely to first encounter targets on the edge of their sonar beam during a search. It is unknown, however, if target classification is possible from the off-axis (OA) information alone, or whether a more centrally focused interrogation is necessary. This talk addresses the initial findings of an animal detecting two different targets (cylinder and sphere) presented OA (left and right). Data collection methods will be presented. Outgoing echolocation clicks and echoes are digitized and stored to a PC for acoustic characterization using a high-speed Integrated Circuits Systems, Ltd. 32 channel A/D card, sampling 24 calibrated monitor hydrophones and analog filter-amplifiers arranged in a hemispherical support web in front of the animal. Emitted signals analyzed for various acoustic characteristics are discussed as well as detection performance. Since this is an on-going study, available results to date will be presented.

4:00


An unusual fish chorusing behavior has been observed in nighttime underwater acoustic recordings during the Summer off the Southern California coast. Some characteristics of these choruses, i.e., increases in the ocean sound levels by factors of 2 to 5 for 10 to 20 s followed by 15- to 20-s periods of lower levels, repeating every 30 to 40 s all throughout the night, have been described previously. Recently, in reanalyzing data collected during a set of experiments in which Van Holliday participated, we discovered that the choruses along one 25-km stretch of coastline have characteristics analogous to “The Mexican Wave” performed by spectators at sporting events worldwide. Each cycle of the chorus begins in waters off the Mexican coast and the region of chorusing propagates up-coast until the fish just south of the mouth of San Diego harbor reach a chorus peak 16 to 20 s after their Mexican counterparts down-coast. This pattern repeats at 30- to 40-s intervals. The speed of this upcoast migration is 100 times faster than the 12 m/s human waves in stadia, approaching the 1.5 km/s speed of sound in water. [Work supported by ONR.]

4:15


The Marine Mammal Active Sonar Test was conducted in January 2004 off the central California coast during the gray whale migration. The purpose of the test was to collect data to significantly advance active sonar detection, classification, and tracking of marine mammals out to ranges of 1 mile. The R/V NEW HORIZON was moored in the migration path. Two different sonar systems operating between 210 and 220 dB were deployed off the vessel, the IMAPS phased array sonar system that operates from 20–30 kHz and the MAST mechanical system that uses rotating parabolic transducers operating from 30 to 40 kHz. Marine mammal observers were deployed on the bluffs overlooking the experiment and aboard the NEW HORIZON. The observers tracked the whales using electronic theodolites, providing ground truth for the sonar systems. They also looked for any severe reactions from the animals, and called for a shutdown if any marine mammals were observed within 100 m of the sonar. Here, we discuss the experiment and the results to date, including any reaction of the whale to the sonar system. We will also touch on the legal battle to conduct the experiment, and lessons learned.

4:30

4pAO14. Sources of uncertainty in Doppler sonar measurements of fish speed. Cristina D. S. Tollefsen and Len Zedel (Dept. of Phys. and Phys. Ocean., Memorial Univ. of NF, St. John’s, NF A1B 3X7, Canada)

A 250-kHz, 30-kHz bandwidth coherent Doppler sonar was evaluated to determine sources of uncertainty in fish speed measurements. Three separate tests were undertaken: (1) towtank tests using styrofoam balls to simulate fish, (2) tank tests with live free-swimming fish, and (3) field tests with wild free-swimming fish. The standard deviation in a single speed estimate was 9 cm s⁻¹ for styrofoam balls, 10–11 cm s⁻¹ for swimming fish observed from a dorsal aspect, and 19 cm s⁻¹ for swimming fish observed from a caudal aspect. The variation in precision was primarily due to the different signal-to-noise ratio (SNR) in each test; a larger SNR resulted in a smaller standard deviation. Doppler speed estimates were compared with independent estimates of target speed where possible. An accuracy of ±4 cm s⁻¹ was typical of Doppler speed estimates in all the experiments.
Biomedical Ultrasound/Bioreponse to Vibration: Contrast Agents, Cavitation and Lithotripsy

Charly Thomas, Chair
Department of Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215

Contributed Papers

1:00
4pBB1. Inertial cavitation dose produced ex vivo in rabbit ear arteries with optison. Juan Tu, Andrew Brayman, and Thomas Matula (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St. Seattle, WA 98105, matula@apl.washington.edu)

Ultrasound-induced inertial cavitation (IC) effects were studied ex vivo in rabbit ear arteries with the addition of ultrasound contrast agents (UCAs). Ears were removed from New Zealand white rabbits immediately after being euthanized under a protocol approved by the University of Washington IACUC. The auricular arteries were perfused with varying concentration of UCA (Optison) in saline and exposed to 1.155-MHz pulsed high-intensity focused ultrasound (HIFU) with constant PRF (10 Hz), pulse length (20 cycles), and total treatment time (20 s). Experiments were performed for variable peak negative acoustic pressure (P –) (from 0.19 to 3.31 Mpa) and Optison volume concentration (0% [saline only], 0.1%, 0.2%, 0.5%, and 1%). Cavitation activity was quantified by IC Dose (cumulated root-mean-squared [rms] broadband noise amplitude in a particular band in the frequency domain). The results showed that (1) IC activity was induced much more easily with the addition of Optison, even at low volume concentration, such as 0.1%. (2) IC dose increased significantly with the increasing acoustic pressure and Optison concentration. (3) Higher concentrations of Optison decreased the IC threshold. [Work supported by NIH 8RO1 EB00350-2.]

1:15
4pBB2. Signal-to-noise ratio and attenuation of Optison® microbubbles in blood as a function of imaging frequency. Paolo Zanetti, Constantin-C. Coussios, and Ronald A. Roy (Boston Univ., 110 Cummington St., Boston, MA 02215, pzanetti@bu.edu)

Using an active cavitation detector (ACD), the power backscattered by various concentrations of Optison® microbubbles (signal) was compared to the power backscattered by a 50% hematocrit suspension of red blood cells in saline (noise), as a function of imaging frequency (5–30 MHz). A theoretical model, based on direct experimental measurements of the size distribution of Optison® microbubbles, was developed to predict the signal-to-noise ratio (SNR) of microbubbles in blood, assuming no interactions between the populations of scatterers. The SNR was shown experimentally to decrease with increasing imaging frequency up to a point where Optison® no longer provided image enhancement. Measurements of the SNR were repeated in a suspension of 0.8% hematocrit, which has the same backscattering coefficient as a 50% hematocrit suspension. The SNR for Optison® in 50% hematocrit was found to be lower than for the 0.8% hematocrit suspension at all frequencies, suggesting that the small number density and close proximity of the red blood cells inhibits the acoustic response of the microbubbles. Measurements of sound attenuation through suspensions of red blood cells with or without Optison® were also obtained, indicating that the microbubbles barely contribute to the overall attenuation. [Work supported by the NSF and the ASA.]

1:30
4pBB3. In vitro characterization of echogenic liposomes by acoustic scattering at 3.5–15.0 MHz. Tyrone M. Porter, Christy K. Holland, Saurabh Datta (Dept. of Biomed. Eng., Univ. of Cincinnati, 6166 MSB, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, tyrone.porter@uc.edu), Shaoling Hwang, Robert C. MacDonald (Northwestern Univ., Evanston, IL 60208), and David D. McPherson (Northwestern Univ., Chicago IL 60611)

Echogenic liposomes (ELIP) are phospholipid vesicles that are being developed as ultrasound contrast agents and as vehicles for drug delivery. These particles, when conjugated with antibodies or peptides, can be used for targeted diagnostic imaging and therapy. Evaluating the acoustic properties of ELIP will allow for optimization of its utility as an ultrasound contrast agent. The diameter of the liposomes ranges from 0.25–10.0 μm. In order to measure the backscattering coefficient in vitro, the backscattered power was compared to that reflected from a perfect reflector, a planar air-water interface. The backscatter coefficient and attenuation of ELIP were evaluated as a function of concentration (0.8–9.9×10^7/ml) and frequency (3.5–15.0 MHz). By comparing the values of measured backscatter coefficient to a theoretical model that treats the gas within the liposomes as a free air bubble, a size estimate of the encapsulated gas is provided. Finally, calculation of the scattering to attenuation ratio (STAR) gives a value with which to evaluate the efficacy of ELIP as a contrast agent in cardiovascular ultrasound imaging. The backscatter coefficient shows promise as a sensitive method for determining whether the liposomes are left intact or if they are destroyed during imaging.

1:45
4pBB4. Bubble translation and deformation induced by ultrasound radiation force. Yuriy A. Ilinskii, G. Douglas Meegan, Evgenia A. Zabolotskaya (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029, zhenia@arl.utexas.edu), and Stanislav Y. Emelianov (Univ. of Texas, Austin, TX 78712-1084)

Measurement of small-bubble dynamics has been proposed for the remote evaluation of tissue elasticity [Erpelning et al., Proc. IEEE Ultrasound Symp., 554–557, 2003]. For example, a microbubble can be produced within the cornea during femtosecond laser surgery and its response to a pulsed ultrasonic radiation force can be measured. The bubble’s translation, deformation, and oscillation can be directly related to the mechanical properties of surrounding tissue information that is required for optimization of the surgical procedure. In the work reported here, a model was developed to predict the translation and deformation of an initially spherical bubble in a soft viscoelastic medium as induced by radiation pressure. The extent of bubble translation and deformation is dictated by the elastic stress and viscous forces that oppose the radiation pressure. Numerical simulations predict static, periodic, and transient translation of the bubble in response to continuous, periodic, and pulsed waveforms, respectively. The model also predicts the deformation of the bubble. The results indicate increased deformations with increased bubble translations. Overall, the model can be used to determine the local shear modulus and viscosity of the medium based on measurements of gas-bubble displacement. [Work supported by ARL-UT IR&D.]
4pBB5. Linear contrast agent detection through low frequency manipulation of high frequency scattering properties. Rune Hansen and Bjorn A. Angelsen (Dept. of Circulation and Medical Imaging, Norwegian Univ. of Sci. and Technol., Norway)

In medical ultrasound imaging, contrast agents in the form of encapsulated gas bubbles are injected into the blood to enhance the scattered blood signal which is weak compared to the scattered tissue signal. Obtaining blood information is, from a medical diagnostic point of view, very helpful. Due to the strong linearly back-scattered tissue signal, contrast imaging today relies on the nonlinear scattering properties of the added gas bubbles. These harmonic techniques typically have important limitations in sensitivity, specificity, and image range resolution. The present paper proposes a new method applying the total scattered contrast signal for image reconstruction, thus largely overcoming the problems encountered in contrast harmonic ultrasound imaging techniques. In the new method, contrast signals and tissue signals are differentiated applying a simple pulse subtraction technique which cancels or significantly reduces the scattered tissue signal, whereas the scattered contrast signal, due to assisting transmitted low frequency pulses altering the acoustic scattering properties of the contrast agent, is preserved in this process. The main mechanism through which this imaging technique selects the contrast agent signal is the linear resonant properties of the contrast bubble and the new method is thus mainly a linear contrast agent detection technique.

2:15

4pBB6. Interaction between therapeutic ultrasound propagation and cavitation bubbles. Marko Liebler, Thomas Dreyer, and Rainer Riedlinger (Institut fuer Hoechstfrequenztechnik und Elektronik/Akustik, Univ. of Karlsruhe, Germany, marko.liebler@ihe.uka.de)

In medical applications of high intense focused ultrasound using pressure pulses or continuous wave signals, cavitation is considered to play a significant role for physical and biological effects. To further develop therapeutic applications it is essential to improve the understanding of these cavitation related effects. In this paper a numerical model is presented to simulate the interactions between ultra sonic waves and cavitation bubbles. The FDTD model is based on a two-phase continuum approach for bubbly liquids and combines nonlinear ultrasound propagation with cavitation bubble activity. Experimental and numerical investigations are presented demonstrating the influence of cavitation bubbles on ultrasound propagation. Measurements with a fiber optic hydrophone for pulsed piezoelectric transducers show significant variations in focal pressure waveforms after the first tensile phase of the wave for different gas content. It is supposed that these changes are caused by cavitation effects. Calculations with different bubble densities confirm these experimental results and demonstrate that the first positive pressure part of the wave is not affected by bubble activity. Increasing the gas content leads to a truncated tensile part followed by augmented pressure oscillations. Further on, simulation results for the evolution and impact of cavitation bubble clouds in CW applications are presented.

2:30

4pBB7. Design of multi-frequency cavitation fields for spatial control of cavitation. Sham D. Sokka, Thomas P. Gauthier, and Kullervo Hynynen (Dept. of Radiol., Brigham and Women’s Hospital, Harvard Med. School, Boston, MA 02115)

Cavitation has been implicated as the primary mechanism for a whole host of emerging applications. In all these applications, the main concern is to induce cavitation in perfectly controlled locations in the field; this means specifically to be able to achieve cavitation threshold at the geometrical focus of the transducer without stimulating its near field. In this study, we develop multi-frequency methods to preferentially lower the cavitation threshold at the focus relative to the rest of the field. Three families of multi-frequency driving waveforms are evaluated in a bubble model incorporating rectified diffusion. The results from the optimal waveform analysis are verified by experiment. Finally, the performance of the rest of the acoustic field in suppressing cavitation when cavitation is induced at the focus is investigated theoretically and checked experimentally. This study shows that multi-frequency phased arrays could be used to precisely control cavitation. Cavitation threshold is proved to be almost 1.5 times higher in the near field than at the focus. The concept of cavitation field is introduced and complements cavitation studies concentrating on the focal behavior only.

2:45

4pBB8. Simulation of an acoustically excited bubble near a simulated “cell.” Sheryl M. Gracewski (Dept. of Mech. Eng., Univ. of Rochester, Rochester, NY 14620, gracec@me.rochester.edu), Hongyu Miao, Diane Dalecki (Univ. of Rochester, Rochester, NY 14620), and Morton W. Miller (Univ. of Rochester, Rochester, NY 14642)

A variety of independent studies have reported increased bioeffects such as hemolysis and hemorrhage induced by high-intensity ultrasound when ultrasound contrast agents are present. Therefore, to better understand the role of cavitation, one-, two-, and three-dimensional models have been developed to investigate the interactions between ultrasonically excited bubbles and model “cells.” First, a simple one-dimensional model based on the Rayleigh–Plesset equation was used to estimate upper bounds for strain, strain rate, and areal expansion of a simulated red blood cell. Then, two- and three-dimensional boundary element models were developed (with DynaFlow Inc.) to obtain simulations of asymmetric bubble dynamics in the presence of rigid and deformable spheres. A spherical “cell” near an ultrasonically excited bubble was modeled using Tait’s equation of state for water, surrounded by a “membrane” with surface tension that increased linearly with areal expansion. The effect of a nearby “cell” on bubble response, the resulting pressure field around the “cell,” and “cell” membrane tensions were investigated. Preliminary results were compared with critical values for hemolysis reported in the literature. [Work supported by NIH.]

3:00–3:20 Break

3:20

4pBB9. Model for interaction of bubbles in a cloud near a rigid surface. Evgenia A. Zabolotskaya, Yurii A. Ilinskii, G. Douglas Meegan (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029), and Mark F. Hamilton (Univ. of Texas, Austin, TX 78712-1065)

Bubble clouds produced during lithotripsy undergo complicated motions including bubble interactions that may inhibit kidney stone comminution. Our study of bubble interactions is motivated by high-speed photographs reported by Pischalnikov et al. [J. Acoust. Soc. Am. 114, 2386 (2003)]. In the work reported here, we simulated the observed bubble motion with a model based on the equations derived by Zabolotskaya [Sov. Phys. Acoust. 30, 365 (1984)]. The equations for interaction of two bubbles were generalized and solved numerically for a cluster of n bubbles near a rigid boundary, which represents the stone. The initial spatial distribution of bubbles in three dimensions was assumed to be random. When a short negative pressure pulse was applied, the simulated bubbles grew in size. When two bubbles touched each other, they were merged into a single bubble that conserved mass of the gas. Results are presented in selected planes intersecting the bubble cloud for different instants of time. Bubble interaction was found to reduce the maximum sizes to which the bubbles grow. The bubbles near the rigid boundary are constrained by neighboring bubbles and grow less rapidly, and to smaller sizes, than other bubbles. Interactions within the cloud thus suppress bubble growth and cavitation. [Work supported by ARL:UT IR&D.]
4pBB10. Importance of pulse synchrony to stone comminution in dual-pulse lithotripsy. Yuri A. Pishchalnikov, James A. McAtter (Dept. of Anatomy and Cell Biol., Indiana Univ., 635 Barnhill Dr., Indianapolis, IN 46202, yura@anatomy.iupui.edu), Irina V. Pishchalnikova, Richard J. VonDerHaar, James C. Williams, Jr., Andrew P. Evan (Indiana Univ., Indianapolis, IN 46202), Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Seattle, WA 98105), and Robin O. Cleveland (Boston Univ., Boston, MA 02215)

We have characterized the acoustic output and in vitro stone breakage of the first dual-pulse lithotripter approved for patient treatment. Direx Medical Systems provided their Duet lithotripter and two timing circuits—a standard circuit that fires both electrodes in synchrony and a modified circuit that allows pulse delay. We assessed the effect of synchrony and ~10-µs delayed pulses on breakage of model stones. When synchrony was very close the waveform had a single peak ~70 MPa P', followed by a negative trough ~12 MPa. The waveform from one head was about half this amplitude. Pulses close in timing but not synchronous (~1-2-µs apart) showed amplitude-summed conjoined peaks. Dual pulses fired ~10-µs apart appeared as independent SWs. Simultaneous pulses broke stones better than pulses delivered by only one head (p < 0.001) and better than ~10-µs delayed pulses (p < 0.0001). Delayed dual pulses were also less efficient than pulses delivered by one shock head (p < 0.001). Pulse timing is critical to stone breakage in dual-pulse lithotripsy. When pulse timing is very close, breakage is better than pulses from one source. The observation that asynchronous pulses are less efficient highlights the importance of precision in dual-pulse SW sources. [Work supported by NIH-DK43881, ONRIFO-N00014-04-1-4010.]

3:50

4pBB11. The role of shear and longitudinal waves in the kidney stone comminution by a lithotripter shock pulse. Oleg A. Sapozhnikov (Dept. of Acoust., Faculty of Phys., M.V. Lomonosov Moscow State Univ., Leninskoe Gory, Moscow 119992, Russia, oleg@acs366.phys.msu.ru), Robin O. Cleveland (Boston Univ., Boston, MA 02215), and Michael R. Bailey (Univ. of Washington, Seattle, WA 98105)

Shock wave lithotripsy has been in clinical use for 20 years but there is no consensus as to the main mechanism of kidney stone comminution. Experiments show that several mechanisms might be involved, including cavitation, spallation, and dynamic fatigue. Until recently, little attention was paid to shear elasticity of the stone material, i.e., mechanical load was mainly attributed to the longitudinal waves. In a previous numerical study, we found that shear elasticity resulted in tremendous change in the stress pattern inside cylindrical stones. The numerical model has been extended to study elastic waves in asymmetric inhomogeneous stones. Strains and stresses in the stone are calculated based on the Lamé equation for an isotropic elastic medium. Lithotripter shock waves of various temporal and spatial profiles were considered according to several clinical models of lithotripters. Maximum compression, tensile and shear stresses are predicted as a function of stone dimension and shape. The model predicts that both shear and longitudinal waves play an important role in creating the regions of excess stresses where cracks can be formed. The results of modeling are compared with the experimental observations. [Work supported by ONRIFO, CRDF, NIH-Fogarty, RFBR, NIH, and Whitaker Foundation.]

4pBB12. Comparison between an open-cage electrode lithotripter and an encapsulated electrode lithotripter. Parag V. Chitnis and Robin O. Cleveland (Aerosp. and Mech. Eng. Dept., Boston Univ., 110 Cummington St., Boston, MA 02215, pchitnis@bu.edu)

We compare the acoustic and cavitation field of two electrohydraulic lithotripters: one employed an open-cage electrode (OCE) and the other an encapsulated electrode (ECE). Acoustic pressure was measured with a fiberoptic probe hydrophone and cavitation using a passive detector consisting of two confocal transducers. The focused shock waves (FSW) of the two lithotripters were similar in shape and at 20 kV, peak positive pressure (P+) for the OCE was 23.2 ± 4.4 MPa and peak negative pressure (P-) was 9.0 ± 1.5 MPa. For the ECE the peak pressures were 30.8 ± 9.4 and −8.2 ± 1.3 MPa, respectively. However, the direct waves (DW) were different. The OCE-DW was triangular shaped and almost entirely positive pressure, whereas the ECE-DW consisted of numerous cycles and had a peak negative pressure of 1.3 MPa. For both electrodes the passive cavitation measurements exhibited two acoustic emissions characteristic of inertial collapses. For the OCE the first emission was weak (in comparison to the second emission) and was attributed to scattering of the FSW. For the ECE both emissions were of similar amplitude, indicating that the DW generated cavitation bubbles that were forced to a violent collapse by the FSW. The ECE produces two violent cavitation events for every shock wave. [Work supported by HMT and NIH.]

3:35

4pBB13. Volume measurements of the cell destruction zone and thermal decomposition zone in the focus of a shock wave transducer. Igor Mastikhin (MRI Ctr., Phys. Dept. Univ. of NB, 8 Bailey Dr., Fredericton, NB E3B 5A3, Canada, mast@unb.ca), Vyacheslav Teslenko (Lavrentiev Inst. of Hydrodynamics, Novosibirsk 630090, Russia), and Valery Nikolin (Inst. of Cytology and Genetics, Novosibirsk, 630090 Russia)

Evaluation of the volume of the cell destruction zone is of interest in biomedical applications of shock waves (SW). The volume depends on mechanical properties of the cell membranes and is different for different cell types. In this work, we evaluated the cell destruction volume for two different cell types, tumor cells Crebs-2 and red blood cells. We used 0 70 0.5-s SW pulses with 45-MPa pressure in the focal zone. The concentration of destroyed cells was counted by dyeing in the case of tumor cells, and by spectrometry of released hemoglobin in the case of RBC. The cell destruction volume was calculated from destruction versus pulse number data and measured as 0.0135 ml for tumor cells. For RBC, the volume was 0.021 ml. To evaluate the effective volume of thermal zone, we used EPR signal of stable disulphide biradicals. Under SW action, S–S bonds of the biradicals rupture. The volume measurements were 0.003 ml. Since for that biradical, S–S bonds rupture at temperatures >80°C, and concentration of free radicals was an order lower (measured by spin traps) than of the produced monoradical, the rupture was caused by thermal decomposition. Thermal effects can play a significant role in SW action.
Session 4pED

Education in Acoustics: Careers in Acoustics

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

Chair’s Introduction—2:00

Invited Papers

2:05


Consulting involves both the science of acoustics and the art of communication, requiring an array of inherent and created skills. Perhaps because consulting on architectural acoustics is a relatively new field, there is a remarkable variety of career paths, all influenced by education, interest, and experience. Many consultants juggle dozens of chargeable projects at a time, not to mention proposals, seminars, teaching, articles, business concerns, and professional-society activities. This paper will discuss various aspects of career paths, projects, and clients as they relate to architectural-acoustics consulting. The intended emphasis will be considerations for those who may be interested in such a career, noting that consultants generally seem to thrive on the numerous challenges.

2:20

4pED2. Acoustical consulting—Reflections on a challenging career. David Braslau (David Braslau Assoc., Inc., 1313 5th St. SE, Ste. 322, Minneapolis, MN 55414)

The acoustical consulting profession can be entered in a number of ways. The most direct approach is to obtain a degree in acoustics and join a large consulting firm immediately after graduation. Acoustical consulting can also be entered indirectly from various fields of engineering or physics which can provide a somewhat broader background. These disciplines might include, for example, structural engineering and structural dynamics, mechanics of materials, dynamic behavior of solids or geophysics. Acoustical consulting specialization can be very broad or very narrow as seen from the National Council of Acoustical Consultants capability listing. As an acoustical consultant, one must address a wide range of problems which provides both the challenges and joys of this profession. Technical capabilities and professional judgment are constantly developed from exposure to these problems and through interaction with other members of the profession. Selected case studies including sound isolation in buildings, noise and vibration from blasting, control of noise from environmental sources, acoustical design of classrooms and performing spaces, and product design demonstrate the variety of challenges faced by an acoustical consultant.

2:35

4pED3. What happens when you want to talk to the animals: lessons of a career in animal bioacoustics. Ann E. Bowles (Hubbs-SeaWorld Res. Inst., 2595 Ingraham St., San Diego, CA 92109, annb1@san.rr.com)

Animal bioacoustics (AB) is the study of sound in nonhuman animal biology. I entered the field because I was interested in the evolution of language, and I wanted to study the acoustic communication of whales and dolphins. Topics like this within the scope of AB make the discipline accessible to students and laypeople. Although career opportunities are limited (professionals declaring AB as their primary area represent only 3% of ASA membership [http://www.acoustics.org/WIA, statistics for 2000]), an interest in AB can foster entry into more marketable disciplines. It has been a particularly important avenue for bringing women into careers in traditionally male-dominated subject areas. For example, women represent 14% of the ASA membership and 12% or fewer of those declaring Underwater Acoustics, Engineering Acoustics, and Noise as their primary interest. However, 25% of those declaring AB as their primary interest are women, and AB includes all three topic areas within its scope. Unfortunately, AB is still fairly inaccessible to interested lay professionals such as educators, science writers, and environmental planners. By helping them to develop a deeper understanding of topics in AB, ASA can help them foster careers in acoustics.

2:50


An undergraduate or graduate degree in biomedical engineering prepares students to solve problems at the interface between engineering and medicine. Biomedical engineering encompasses evolving areas such as advanced medical imaging for diagnosis and treatment of disease, tissue engineering for designing and manufacturing biological implants for damaged or diseased tissues and organs, and bioinformatics for determining which genes play a major role in health and disease. Biomedical engineering academic programs produce graduates with the ability to pursue successful careers in the biomedical device industry or to obtain advanced degrees leading to careers in biomedical engineering research, medicine, law or business. Biomedical engineering majors take courses in biology, anatomy, physics, chemistry, engineering, mathematics and medical product design and value life-long learning. Students
learn to work effectively in interdisciplinary teams comprised of individuals with diverse social, cultural and technical backgrounds. Biomedical engineering is becoming increasingly important in imaging and image-guided research. Some examples of innovative ultrasound technology under development are ultrasound devices to accelerate the dissolution of blood clots, advanced surgical instruments with ultrasound guidance and ultrasound contrast agents for targeted drug delivery. Biomedical engineering is a great career choice for technically minded individuals who endeavor to work on applied problems that are medically relevant.

3:05

4pED5. **Life as an acoustician in industry, academia, and government service.** Mardi C. Hastings (ONR, 800 N. Quincy St., Arlington, VA 22217, mardi_hastings@onr.navy.mil)

Acoustics is a science that has very broad applications, which affect all different areas of our lives. During the last 20 years, I have combined family with a career as an acoustics engineer in industry, a tenured faculty member at a university and, most recently, a program manager in a government agency. In these positions I have worked in several areas of acoustics, including noise control, structural acoustics, building acoustics, sound quality, physical acoustics, acoustic materials, underwater acoustics, biomedical ultrasound, physiological acoustics, and bioacoustics. Although the fundamental science of sound is the foundation of all these areas, communication of ideas, problems, and solutions varies greatly from industry to academia to government. Thus knowing the science and how to use it are not enough, as communication skills and the ability to adapt them to changing environments are essential for a successful career. In addition to describing life as an acoustician in industry, academia, and government service, I will present several examples of how even though the acoustic fundamentals are the same, how they are communicated could become a disaster or save the day.

3:20

4pED6. **Acoustics careers for engineers.** Uwe J. Hansen (Indiana State Univ., Terre Haute, IN 47809)

Many acoustics opportunities in industry, government laboratories, and academics rely on a background in mechanical engineering, electrical engineering, or physics. Acoustics deals principally with generation, propagation, and perception of sound. Engineering application, include among many other things, the study and control of structural vibrations, machinery analysis and maintenance, and industrial noise control. Thus, for example, the aircraft industry employs engineers to study vibrational characteristics of items such as turbine blades or entire fuselage assemblies. Among the many techniques utilized are holographic interferometry and modal analysis. Some of these methods will be illustrated.

3:35–3:45 Break

THURSDAY AFTERNOON, 27 MAY 2004

CONFERENCE ROOM K, 1:15 TO 4:55 P.M.

**Session 4pMU**

**Musical Acoustics: New Research on Pre-1929 Instruments**

Thomas D. Rossing, Cochair

*Physics Department, Northern Illinois University, De Kalb, Illinois 60115*

D. Murray Campbell, Cochair

*Department of Physics and Astronomy, University of Edinburgh, Mayfield Road, Edinburgh EH9 3JZ, United Kingdom*

Chair’s Introduction—1:15

**Invited Papers**

1:20

4pMU1. **Pitch bending on the cornet.** D. Murray Campbell (School of Phys., Univ. of Edinburgh, Mayfield Rd., Edinburgh EH9 3JZ, UK, d.m.campbell@ed.ac.uk)

A successful model of a brass wind instrument must be able to describe the playing technique known as “lipping,” in which the player can bend the pitch of a note by modifying the setting and tension of the lips. This technique is of special importance on the lip-reed instrument known as the cornet or cornetto, which was important in the 15th and 16th centuries but fell out of use by the end of the 18th century. The instrument consists of a short wooden tube of approximately conical bore with seven side holes. Chromatic notes must be obtained by cross fingering, and good intonation requires fine control of the lipping technique. Recent experimental studies of lipping on the cornet are described, and implications for modeling of the lip reed are discussed.
4pMU2. The acoustical engineering of brasswind instruments 1779–1929. Arnold Myers (Univ. of Edinburgh, Reid Concert Hall, Bristol Square, Edinburgh EH8 9AG, UK, A.Myers@ed.ac.uk)

At the start of this 150-year period, brass musical instruments were made to traditional designs, which were developed by trial and error, the fittest surviving. Mechanical inventions (most importantly the valve) greatly widened the possibilities for bore engineering. Increasingly through this period, instrument designers were influenced by the developing science of acoustics. By 1929 most of the range of instruments in use today had been developed and acoustical tools were in use in optimizing the design of instruments. Considering the factors of greatest importance in determining the acoustical response of a brass instrument to be bore profile, bell flare (cutoff frequency), and mouthpiece geometry, landmarks in the development of existing instrumental types and the creation of new models are surveyed. The contributions of Stoelzel, C.M. Pace, Sax, Bayley, Blaikley, Webster, and Couturier are discussed. This paper is based on research involving direct examination of several hundred instruments from 1779–1929 located in museums worldwide.


The art of casting bronze bells developed to a high level of sophistication in China during the Shang dynasty (1766–1123 BC). Many chimes of two-tone bells remain from the Western and Eastern Zhou dynasties (1122–249 BC). With the spread of Buddhism from the third century, large round temple bells developed in China and later in Korea, Japan, and other Asian countries. Vibrational modes of some of these bells have been studied by means of holographic interferometry and experimental modal testing. Their musical as well as acoustical properties are discussed.

4pMU4. The design and analysis of new musical bells. Neil M. McLachlan (School of Aerosp., Manufacturing and Mech. Eng., RMIT, GPO Box 2476V, Melbourne Vic. 3000, Australia, neil.mclachlan@rmit.edu.au)

The design and analysis of a series of new musical bells will be presented in this paper. Modal analysis of a wide range of bell-like geometries using FEA revealed the presence and significance of transverse axial modes in unconstrained bell models, leading to a new understanding of the relationships between bell geometry and modal behavior. This understanding was used to adjust simple parametric models of bell geometry to arrive at appropriate geometries to begin numerical shape optimization for the design of bells with a range of desired overtone tunings. Pitch salience is well known to depend on the degree of harmonic relationships between pure tones in complex stimuli. Bells intended to produce a single, highly salient pitch were designed and manufactured with up to the first 7 overtones tuned to the harmonic series. Other bells with overtones tuned to subsets of two or three harmonic series were also designed and manufactured. These bells were intended to produce multiple pitch perceptions of approximately equal strength. Spectral analysis and range of numerical psycho-acoustic models are used to evaluate the sounds of manufactured bells against these design objectives. [I would like to acknowledge the close collaboration of Dr. Anton Hasell of Australian Bell.]

4pMU5. Normal modes of different types of mandolins. David J. Cohen (Cohen Musical Instruments, 9402 Belfort Rd., Richmond, VA 23229) and Thomas D. Rossing (Northern Illinois Univ., DeKalb, IL 60115)

The vibrational modes and sound spectra of some pre-1929 archtop mandolins and pre-1922 Neapolitan (“bowlback”) mandolins have been studied. The results have been compared with those obtained previously on archtop mandolins constructed more recently [D. Cohen and T. D. Rossing, CASJ 4(2), 48–54 (2000), D. Cohen and T. D. Rossing, Acoust. Sci. Tech. 24, 1–6 (2003)]. Some obvious and predictable differences between the Neapolitan mandolins and the archtop mandolins were found. The very stiff bowls of the Neapolitans do not contribute to corpus vibrations below about 1.2 kHz. The ladder-braced top plates of the Neapolitans are also quite stiff, with the (0,0) mode first occurring at or above 500 Hz. The (0,0) modes in archtop mandolins generally occur at lower frequencies. Archtop mandolins with f-holes generally have either longitudinal bracing or X-bracing, with the results that the modes involving cross-grain bending [e.g., (1,0), (2,0), etc.] occur at lower frequencies than the modes involving bending along the grain [e.g., (0,1), (0,2), etc.]. In the ladder-braced Neapolitans, the modes involving cross-grain bending occur at higher frequencies than the modes involving bending along the grain.

3:00–3:10 Break

4pMU6. The acoustics of carved Baltic psaltery. Andres Peekna (Innovative Mech., Inc., 265 Coe Rd., Clarendon Hills, IL 60514-10299, inmmech@comcast.net) and Thomas Rossing (Northern Illinois Univ., DeKalb, IL 60115)

The Baltic psaltery family of plucked string instruments includes the kantele (Finland), the kannel (Estonia), the kokle (Latvia), the kankles (Lithuania), and the wing-shaped gusli (Northwestern Russia). In its archaic, carved form, it has a limited range, 5–13 strings, usually tuned diatonically. By means of electronic TV holography, we studied the modes of vibration of several psalteries based on historic instruments. On the better instruments, the main body resonances are well distributed in frequency so that they support the various strings. Good string-to-soundbox coupling also appears to play a role. A useful method for studying string-to-soundbox coupling involves scanning at intervals as low as 0.1 Hz for narrow peaks within the nominal tuning range of the strings,
and comparing them to their neighboring body resonances, while using electronic TV holography. Predictions of the Helmholtz resonance from sound-hole dimensions and air-cavity volume while neglecting damping in the sound holes yield upper limits when many small sound holes are involved. The locations of the sound holes, as well as their area, are found to have significant effects on sound quality and volume.

3:30

4pMU7. The harpsichord in 1929 and the emperor’s new clothes. Edward Kottick (Univ. of Iowa, Iowa City, IA 52242)

By the end of the 19th century the harpsichord was considered obsolete. Viewed from the vantage point of a century of brilliant advances in piano technology, it was an outworn, outmoded relic of a distasteful past. It was overly delicate in nature; its soundboard was too thin, and thus unstable; its case construction was unsubstantial. Its light, unbushed keyboards were inadequate. Its quaint bird-quill plectra were viewed with disdain, and its sound was thought to be weak and unattractive. Nevertheless, by the year 1929 the harpsichord was enjoying a revival that was at that time in its 40th year, and that revival continues today. If this instrument was considered so inadequate, why has its revival been so successful and long-lasting? All the answers boil down to one word: sound. This paper will explore the contrast of the tonal qualities of the revival harpsichords and the antiques, and the way those qualities changed in response to the shifting tastes of builders, performers, and listeners. Ultimately, changing tastes led to a return to the sound of the classical harpsichord. Embedded in this tale is an important lesson on the dangers of characterizing the tone of any instrument.

3:50

4pMU8. Physical modeling of Tibetan bowls. Jose Antunes (Instituto Tecnologico e Nuclear, Appl. Dyn. Lab., 2686 Sacavem codex, Portugal, jantunes@itn.mces.pt) and Octavio Inacio (Instituto Politecnico do Porto, 4000-045 Porto, Portugal)

Tibetan bowls produce rich penetrating sounds, used in musical contexts and to induce a state of relaxation for meditation or therapy purposes. To understand the dynamics of these instruments under impact and rubbing excitation, we developed a simulation method based on the modal approach, following our previous papers on physical modeling of plucked/bowed strings and impacted/bowed bars. This technique is based on a compact representation of the system dynamics, in terms of the unconstrained bowl modes. Nonlinear contact/friction interaction forces, between the exciter (puja) and the bowl, are computed at each time step and projected on the bowl modal basis, followed by step integration of the modal equations. We explore the behavior of two different-sized bowls, for extensive ranges of excitation conditions (contact/friction parameters, normal force, and tangential puja velocity). Numerical results and experiments show that various self-excited motions may arise depending on the playing conditions and, mainly, on the contact/friction interaction parameters. Indeed, triggering of a given bowl modal frequency mainly depends on the puja material. Computed animations and experiments demonstrate that self-excited modes spin, following the puja motion. Accordingly, the sensed pressure field pulsates, with frequency controlled by the puja spinning velocity and the spatial pattern of the singing mode.

Contributed Papers

4:10

4pMU9. The acoustics of the Bagana. Stephanie Weisser (Universite libre de Bruxelles, CP 175, 50 av. F. Roosevelt, 1050 Brussels, Belgium) and Didier Demolin (Universidade de Sao Paulo, 01060-970 Sao Paulo, Brasil)

The Bagana lyre of Ethiopia is an instrument with a very particular timbre whose acoustic characteristics is determined by U-like leather strips placed between each string and the bridge. These leather strips give a deep buzzing sound that influences pitch duration, intensity, and timbre. The sounds of the instruments were recorded with and without buzzers for comparison. Data were analyzed to characterize the main acoustic features of the instrument. In addition a high-speed camera (400 images/second) was used to describe the different vibratory modes. The main observation is that buzzers enhance the spectrum up to more than 10 kHz; they also modify the attack and the release of the sound. The duration and the loudness of the sound are increased because they enhance the energy’s repartition in the spectrum, especially in the area around 1500–300 Hz. Another effect of the buzzers is that the instrument produces low pitch and loud sounds without having a big resonator or tall harmonic strings.

4:25

4pMU10. On the development of German beating-reed organ pipes during the 19th century. Jonas Braasch (Faculty of Music, McGill Univ., 555 Sherbrooke St. W., Montreal, QC H3A 1B9, Canada, braasch@mail.mcgill.ca)

In the 19th century organ literature, it is often claimed that German organ builders generally adapted the way of building their beating-reed pipes after being influenced by new developments from England and France. To investigate whether this hypothesis is true or false, the reed-pipe sounds of several German historic organs and an English organ by Henry Willis were measured and analyzed. The outcome of the analysis, however, cannot confirm the given hypothesis. Organ builders of the 18th century, such as Gottfried Silbermann for example, were already able to build beating-reed pipes similar in sound to the pipes that are used nowadays in Germany. It is noteworthy that Silbermann used closed shallots in some of his stops, although they are thought to be one of the main inventions in the English and French organ reforms. The use of higher wind pressures, which is also a main part of this reform, on the other hand, never became a common standard in Germany, as was the case for France and Great Britain.

4:40


Glass musical instruments are probably as old as glassmaking. At least as early as the 17th century it was discovered that wine glasses, when rubbed with a wet finger, produced a musical tone. A collection of glasses played in this manner is called a glass harp. Another type of glass harmonica, called the armonica by its inventor Benjamin Franklin, employs glass bowls or cups turned by a horizontal axle, so the performer need only touch the rim of the bowls as they rotate to set them into vibration. We discuss the modes of vibration of both types of glass harmonica, and describe the different sounds that are emitted by rubbing, tapping, or bowing them. Rubbing with a wet finger tends to excite only the (2,0) mode and its harmonics through a “stick-slip” process, while tapping excites the other modes as well.

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Session 4pNSa

Noise and Architectural Acoustics: Distinguished Lecture “Noise: My 62 Years of It”

Leo L. Beranek, Chair
975 Memorial Drive, Suite 804, Cambridge, Massachusetts 02138-5755

Chair’s Introduction—1:15

Invited Paper

1:20

4pNSa1. Noise: My 62 years of it! Laymon N. Miller (1504 Harbor Court, Fort Myers, FL 33908-1651, laymluce@aol.com)

Imagine getting paid for having fun! Well, in retrospect, it was fun; but there were several tough challenges. Even those are worth remembering. From 1941 to 1982, there were acoustic torpedoes, HVAC acoustics, noise and vibration in auditoriums, aircraft and airport noise, OSHA and industrial noise control, power plants, community noise problems, vibration, railroad and subway vibration control, legal acoustics, noise manuals, and noise courses—and a few other things that don’t fit into those neat categories. Some specific jobs could be named, but that would take away the suspense and the surprise. But 1941 to 1982 is only 41 years. How about the other 20-odd years?

Session 4pNSb

Noise and Psychological and Physiological Acoustics: Noise and Society

Nancy Nadler, Chair
League for the Hard of Hearing, 50 Broadway, New York, New York 10004

Chair’s Introduction—2:30

Invited Papers

2:35

4pNSb1. The nature of noise in society. Leslie Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601)

Noise is unique among pollutants. It is, for example, the only pollutant commonly defined in subjective, psychological terms: “unwanted sound.” Ironically, noise experts work almost entirely in objective measures—sound pressure, sound power, Leq, Ldn, etc. This paper suggests that between the science and engineering of acoustics and the psychology of sound perception, in the societal and social context, lies the true nature of noise—its causes and effects. Enriched by contacts with tens of thousands of individuals over the past 8 years, the Noise Pollution Clearinghouse is working to redefine noise in societal terms. In understanding the nature of noise, the concepts of civility, sovereignty, environmental quality, quality of life, and connectedness to others are much more helpful than “unwanted sound.”

2:55

4pNSb2. Noise pollution: A threat to our mental and physical well-being. Arline L. Bronzaft (Dept. of Psych., Lehman College, Bronx, NY 10468, albtor@aol.com)

While noise may not yet be in the forefront of the environmental movement, it is being recognized worldwide as a major environmental pollutant. In New York City, noise is the number one quality of life complaint, far outweighing other quality of life complaints, and throughout the United States it has been noted as a major reason for people moving from their homes. Although there is a need for additional research to confirm the health/noise link, the World Health Organization has already recognized noise pollution as a serious health issue. There is certainly sufficient research to warrant warnings that noise is injurious to mental and physical health. Yet, despite this growing body of literature attesting to the relationship between noise and health impacts, government bodies have not yet invested the dollars needed to abate noise nor to educate people to the dangers of noise. Organizations such as the League for the Hard of Hearing, the Noise Pollution Clearinghouse, and the United Kingdom Noise Network have assumed the tasks of educating the public to the harmful effects of noise, of advocating anti-noise measures, and of urging public officials to move more assertively in lessening the din of our ever-increasing noisy society.
**3:15**

*4pNSb3. Children’s quality of life in a noisy world.*  
Brigitte Schulte-Fortkamp  
(Inst. of Tech. Acoust., TU-Berlin, Einsteinufer 25, D-10587 Berlin, Germany, brigitte.schulte-fortkamp@tu-berlin.de)

On 30 April 2003 during the International Forum Noise Awareness Day European experts from the fields of medicine, acoustics, sociology, psychology, city planning, and traffic regulation led the current discussion on the risks of noise pollution for children at the Institute of Technical Acoustics at the Berlin Technical University, Germany. Studies probing into the negative effects of noise exposure upon the psychic, cognitive, and emotional functions of children are of a quite recent date. Even the quality of life of children as auto-directed experience and functionality is only now starting to get recognized; of special interest are the changes in self-awareness, performance, and health. The Forum focused upon methodological reflections for the collection of data as well as studies regarding the living conditions of children under sound exposure from different perspectives. In this context the research of sound effects has taken initial tentative steps to break away from the tried-and-true procedures of the last 30 years. Next to applied science, investigating fundamentals will need to be considered. Aiming at a networking process is one of the goals conducting the “Tag gegen Lrm—International Noise Awareness Day.”

**3:35**

*4pNSb4. Training as a critical component of successful noise enforcement programs.*  
Eric Zwerling  
(Rutgers Univ. Noise Tech. Assistance Ctr., 14 College Farm Rd., New Brunswick, NJ 08901)

The point of application of any noise enforcement program is the enforcement officer. The quality of their training is of paramount importance in determining their efficacy in resolving complaints in the field or, failing that, in court. Some of the critical components that must be addressed in a training program are the technology, techniques and strategies of legally valid sound level measurement; documentation of measurement parameters and results; calculation of corrected source sound levels; managing the expectations of complainants; negotiations with alleged violators; and compliance determination methods for nonmetered performance standards. A strong emphasis must be on practical field measurements. The training must assist the enforcement officer to become comfortable with the process, motivating the officer to embrace the new skill, rather than resenting a new task. It is important to take into account the background of the students, professionally, and as individuals, as well as the institutional culture of their agency. The better prepared an officer is to go to court, the less likely is that possibility. A well designed and executed program, represented by its field officers, provides significant deterrence. Thirteen years of training experience at the Rutgers Noise Technical Assistance Center is reviewed.

**3:55**

*4pNSb5. The effects of removing barriers to hearing protector use.*  
Mark Stephenson  
(Ctr. for Disease Control and Prevention/NIOSH, Cincinnati, OH)

When high level noise exposure is unavoidable, wearing hearing protectors is essential to preventing hearing loss and preserving the quality of ones life. Unfortunately, many people fail to wear hearing protectors when they are exposed to loud noise. Focus groups and surveys conducted by the National Institute for Occupational Safety and Health (NIOSH) have identified a set of barriers frequently cited as reasons for not wearing hearing protection. NIOSH has employed these data to develop a field study designed to address barriers to hearing protector use. Interim results have demonstrated that by applying contemporary health communication theory to focus training messages and methods on specific barriers, it is possible to positively influence attitudes and behaviors associated with hearing protector use. This paper will describe the methods used in this study and discuss the results obtained to date.

**4:15**

*4pNSb6. Hearing loss and tinnitus in adolescents and young adults.*  
Alice Holmes  
(Univ. of Florida, Box 100174, Gainesville, FL 32610, aholmes@hp.ufl.edu)

Little attention has been paid to hearing abilities and the effects of noise on the normal adolescent and young adult population. A series of studies will be presented on the prevalence of hearing loss and reported effects of hearing loss and tinnitus in adolescents and young adults from different cultural backgrounds. Adolescents and young adults from different backgrounds may tend to seek or avoid various noise environments that could be detrimental to their hearing and cause tinnitus. Attitudes and exposures to noise environments were evaluated to see if these may be correlated with their hearing losses and/or tinnitus. In addition, these adolescent and young adult subjects reported how often they used hearing protection in various noise environments. Finally, the issues of quality of life and the need for hearing conservation programs with these populations will be presented.
The discussion around the concept of the addiction to noise has evidenced the importance of noise for the human being and explains why in some cases the regulations fail to control the noise in cities. In this presentation the different uses, consciously or unconsciously, of the noise will be analyzed, uses that go from habits to maybe addictions. Also discussed are the implications of establishing regulations against the human nature as well as the importance of education to manage the noise and design acoustically instead of trying to ban the noise in some social circumstances.

The European Directive 2002/49/EC defines useful procedures in order to evaluate, by means of indexes, the acoustic pollution affecting urban areas. To apply those methods correctly, it is necessary to get a properly designed acoustic database. Usually the environmental information managed by the laboratories shows a different kind of problem. Nowadays the Acoustic Laboratory of the University of Cadiz (Spain) is very interested in the developing of new data-processing tools. These tools overcome the errors in the calculation of the indexes due to the use of inappropriate databases. We focus not only on the determination of the possible problems, but on the best solutions. We suggest in this paper a simple data treatment method that permits one to reach more accurate indexes in spite of such common problems.
The investigation of wave propagation and scattering of elastic waves in heterogeneous, anisotropic media is of substantial interest to quantitative nondestructive evaluation and materials characterization. The scattering of elastic waves in polycrystalline media is primarily due to interaction with the grains. Knowledge of wave velocity and attenuation may be used to infer material texture in polycrystalline aggregates. In this presentation, a model for wave propagation and scattering in polycrystalline materials with texture is presented. Attenuations and wave velocities are discussed for a general orthorhombic material made up of cubic crystallites. The attenuations of each wave type are derived as a function of dimensionless frequency and wave propagation direction, respectively, for given orientation distribution coefficients (ODCs). The ODCs are, in essence, the coefficients of an expansion of crystallographic orientation distribution function (ODF) in terms of a series of generalized spherical harmonics. A relationship between the phase velocity and recrystallization variables, such as annealing time, is also investigated for specific examples. Finally, numerical results are presented and discussed in terms of the relevant dependent parameters. The results are anticipated to advance the field of materials characterization of statistically anisotropic media. [Work supported by DOE.]

2:00

4pP5. Scattering of two-dimensional periodic gratings composed of cylindrical cavities in an elastic medium. Sebastien Robert, Hervé Franklin, and Jean-Marc Conoir (LAUE, Université du Havre, place R. Schuman, 76610 Le Havre, France)

A theoretical and a numerical calculation of the scattering by a two-dimensional periodic grating composed of parallel cylindrical cavities embedded in an elastic medium are presented. The major point of the method is that the whole grating can be decomposed as a series of a finite or an infinite number of infinite rows of periodically spaced identical cavities. The scattering by each row is determined by an exact self-consistent multiple scattering calculation developed in the elastic case. The propagation of the waves from row to row can then be determined by an iterative method or from Bloch’s theorem. Compared to previous works on gratings of elastic scatterers in a fluid, this study emphasizes new and interesting results. Among them are the systematic formation of new stopping and passing bands when the scattering is resonant, and the presence of frequency domains of higher coupling between the longitudinal and transverse waves, corresponding to the propagation of guided waves between two reticular planes of the grating. Examples of gratings with only one periodicity direction are also studied.

2:20

4pP6. Backscattering in the vicinity of a mode cutoff. Alan M. Whitman (Villanova Univ., Villanova, PA 19085), Mark J. Beran, and Shimshon Frankenthal (Tel Aviv Univ., Ramat Aviv, Israel)

In a previous paper [Whitman et al., Waves Random Media 13, 269–286 (2003)] a set of coupled equations was derived that describes the intermodal scattering of acoustic radiation in a duct whose speed of sound varies randomly in space and time. In that paper the main interest was in modes that were not near cutoff. Here the solution of these equations in the vicinity of the cutoff is treated. It is found that near cutoff almost all of the energy is reflected back independently of the other duct parameters. In addition to presenting this result the mathematical structure of the equations in these regions is analyzed in order to elucidate the reason for the behavior. Some numerical results are also presented.

2:30

4pP7. Backscattering in space- and time-dependent random media acoustic intensity fluctuations. Shimshon Frankenthal and Mark J. Beran (Faculty of Eng., Tel Aviv Univ., Ramat Aviv 69978, Israel, shim@eng.tau.ac.il)

In two recent publications [(1) S. Frankenthal and M. J. Beran, Waves Random Media 13, 241–268 (2003); (2) A. M. Whitman, M. J. Beran, and S. Frankenthal, ibid. 13, 269–286 (2003)] an incremental slab formulation was employed to derive the equations for the intensity of acoustic signals that undergo scattering in a medium whose sound speed fluctuates randomly both in space and time. These equations were used to track the propagation of a narrow-band pulse and its echoes across a one-dimensionally stratified slab, and to compute the effects of intermodal scattering on the modal intensities along a sinusoidally excited duct. Here, following a brief review of the main assumptions and the major features of the intensity equations, the same formulation is used to derive equations for the mean-square intensity, which is needed to calculate the intensity fluctuations. With sinusoidal excitation, an analytical solution of these equations is obtained for propagation across a one-dimensionally stratified slab, and a system of differential equations, which can be solved by the techniques used in Ref. 2, is derived for the modal mean square intensities in a duct.

2:45

4pP8. Multilayer impedance coating for submerged objects—theory and applications. Ronald Hughes, Jan Niemiec, and Herbert U berall (Naval Surface Warfare Ctr., Carderock Div., Bethesda, MD 20817-5700)

We consider acoustic returns from water-immersed objects with partially absorptive single or multiple coating layers. For sufficiently high absorptivity, the effect of the layers, even when applied to a strongly reflective substructure, can be described by an effective impedance Z, acting at the outer surface of the top layer. Expressions for Z may be obtained following procedures for the corresponding case of electromagnetic scattering by Uberall. In acoustics, the surface impedance of a one-layer coating was obtained by Brekhovskikh and is presented here in a very concise form. For two-layer coatings, Z is obtained analytically by a procedure used by Glegg for sediment layers. Equations by Folds and Loggins or Brekhovskikh can be used for a multilayer case to obtain numerical solutions. With these procedures for obtaining surface impedances, applications have been developed for the study of acoustic reflections from coated submerged structures.

3:00–3:15 Break

3:15


The reflection and transmission coefficients of a slab containing a random distribution of elastic, infinitely long cylindrical scatterers in water are studied. The slab is immersed in water and insonified by a harmonic plane wave. All scatterers are identical, parallel, with axes normal to the incidence plane. A uniform random distribution is supposed. An extension of the Fikioris and Waterman model is developed in order to describe the propagation of the coherent wave in the slab. Its dispersion equation is numerically solved, providing its velocity and its attenuation. When there is only one solution of the dispersion equation that corresponds to an actually propagating wave, the slab is shown to behave as a dissipative equivalent fluid medium (effective medium). The results are compared to...
those obtained from the Waterman and Truex model and to those of Foldy. The reflection and transmission coefficients are then written in a compact form similar to that of a fluid slab, so that the frequency dependence of the density of the effective medium may be studied. Results are shown for different concentrations of scatterers and for two different kinds of cylindrical scatterers.

3:30

4pPA10. Reflection coefficient of a water-loaded monoclinic plate: Obtaining of a factorized expression. Olivier Lenoir and Lionel Guénégou (LAUE UMR CNRS 6068, Univ. Le Havre, Pl. R. Schuman, 76610 Le Havre, France, lenoir@univ-lehavre.fr)

For an isotropic plate, it was shown by Schoch and used by Überall et al. that the reflection coefficient $R$ of an elastic plate immersed in a fluid can be written in terms of the $C_{ij}$ functions and the ratio $\tau$ of the acoustic impedances in the plate and in the fluid. The denominator of $R$ is the product of the $(C_{ij} + j\tau)$ functions. The roots of these functions correspond to the antisymmetric ($A$) and symmetric ($S$) vibration modes of the fluid-loaded plate. In this case, the Lamb-type guided waves are the combinations of dilatational and shear waves polarized in the incident plane. In this study, we consider an orthotropic plate where the incident plane is different from a symmetry plane; therefore, everything happens as if we dealt with a monoclinic plate. In this case, the guided waves are the sum of nonpure longitudinal waves, shear waves, and $SH$ waves. Nevertheless, it can be shown that the expression of the reflection coefficient of the monoclinic plate has formally the same factorized form as the one of the isotropic plate. This expression is numerically compared to the one obtained by the stiffness matrix method developed by Rokhlin et al.

3:45

4pPA11. Self-organizing spatial pattern formation in wedge diffraction. Mitsuhiro Ueda (Dept. of IDE, Tokyo Inst. of Technol., Ookayama, Meguro-ku, Tokyo 152-8552, Japan, ueda@ide.titech.ac.jp)

It is well-known that the amplitude of potential near the apex of a 2D rigid wedge is proportional to the $PAI/WA$th power of the distance between the observation point and the apex, where $PAI = 3.14$ and $WA$ is an apex angle of the wedge measured in the free space. In the case of a semi-infinite plane ($WA = 2\pi$), the square-root dependence is observed. This phenomenon should be regarded as a self-organizing spatial pattern formation, since this pattern arises for any waves incident to the apex. It has been, however, explained in terms of eigenfunctions, and no mechanism for this self-organization has been proposed so far. The boundary value problem for the potential on the boundary of the wedge is formulated using a new principle of diffraction, that is, the virtual discontinuity principle of diffraction (VDPD). By applying this formulation to observation points that are located near the apex, a relation that describes mutual dependence of the potential around the apex is obtained. It can be shown that this relation allows the stable solution only for the specific distance dependence. Thus, the mechanism of the self-organization in wedge diffraction is made clear by the VDPD analysis.

4:00

4pPA12. The phase gradient method (PGM) applied to a monoclinic plate immersed in water. Lionél Guénégou and Olivier Lenoir (LAUE UMR CNRS 6068, Univ. Le Havre, Pl. R. Schuman, 76610 Le Havre, France, lionel.guenegou@univ-lehavre.fr)

For a water-loaded isotropic plate, the PGM deals with the study of the partial derivatives of the phase of its reflection coefficient. The phase derivatives with respect to the frequency $f$, to the bulk phase velocities of the pressure ($c_p$) and shear ($c_s$) waves propagating in the plate, to the phase velocity $c_f$ of the waves in water, and to the incident angle are investigated. The frequency and angular derivatives permit characterization of the frequency and angular resonances of the plate (locations and widths) without calculations in the associated complex planes. The derivatives with respect to $c_{p, f}$ give the prevailing polarization state of the Lamb waves. For a monoclinic plate (orthotropic plate where the incident plane is different from a symmetry plane), at a given incident angle, three phase velocities in the plate are to be considered. The additional one with regard to the isotropic case is the quasi-$SH$ wave phase velocity. It is shown in this study that the frequency phase derivative study is still convenient to obtain the frequency resonance features, and that the phase derivatives with regard to the three velocities indicate whether a guided wave is mainly a quasilongitudinal, quasishear, or quasi-$SH$ mode.

4:15

4pPA13. Flow noise and rapidly distorting turbulence in shear layers. R. Martinez (Cambridge Acoustical Assoc./Anteon Corp., 84 Sherman St., Cambridge, MA 02140)

Rapid distortion theory (RDT) is a linear analytical framework for the formal split of a flow’s hydrodynamic sources from their acoustic effects. The former include time-varying vorticity, turbulence, and unsteady heat injection. Their acoustic effects manifest themselves through a generalized wave-propagation operator in the consistently separated equations of fluid mechanics. The present development reports on progress in extending RDT to include mean shear and associated macro-vorticity in the static but spatially nonuniform carrier flow. The analysis begins by fully recasting standard RDT [Goldstein, JFM (1978)] for rotational backgrounds in tensor-dyadic form for the curvilinear coordinates of the streamlines of the background flow. This complete geometrization of the background offers clues for achieving a similar split in the time-varying acoustical and turbulent variables that perturb the more general sheared freestream. The new development could eventually be applied to turbulent boundary layers, and particularly to spatial discontinuities such as steps and gaps that (rapidly) distort the statistics of the perturbed rotational flow and thereby lead to additional broadband noise via RDT’s generalized wave-equation operator and geometrized source terms.
Session 4pPP

Psychological and Physiological Acoustics and Speech Communication: The Perception of Complex Sounds: Honoring the Contributions of Charles S. Watson

Marjorie R. Leek, Cochair
Army Audiology and Speech Center, Walter Reed Army Medical Center, Washington, DC 20307-5001

Larry E. Humes, Cochair
Department of Speech and Hearing Sciences, Indiana University, Bloomington, Indiana 47405-7002

Invited Papers

1:00

4pP1. “Watson, come here!” Larry Humes (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405-7002)

This famous quote marked a unique historical event in the development of telephony and an early experiment involving hearing. Although our own Watson postdates that work (considerably!), the life and work of Charles S. Watson are true reflections of good science and significant growth in our knowledge of human hearing. On behalf of those speaking at and attending this session in honor of Charles S. Watson, we ask this Watson to “come here!” (or, perhaps as a colleague suggested, to “come hear!”). We ask that he do so to permit us to pay tribute to the many ways he has influenced the fields of psychoacoustics and speech communication, in general, and our careers, in particular. This introduction will provide a very brief overview of Chuck Watson, the person, as well as the scholar. By doing so, this overview will lay a foundation for understanding the impact of this man and his work. Subsequent speakers will build upon this foundation, with special focus on Chuck’s work in the area of the perception of complex sounds.

1:15

4pP2. The role of stimulus uncertainty in speech perception. Diane Kewley-Port (Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405, kewley@indiana.edu)

Among the important experimental factors that affect psychophysical measurements of speech perception is stimulus uncertainty. Charles Watson has defined stimulus uncertainty as variation in stimulus parameters from trial to trial and demonstrated its highly degrading effects on a variety of complex auditory signals. Watson, Kelley, and Wroton showed large (>10) elevation of frequency-discrimination thresholds for “word-length tonal patterns” under high uncertainty conditions [J. Acoust. Soc. Am. 60, 1176–1186 (1976)]. Investigations of speech, such as the perception of VOT (voice onset time) in stops [Kewley-Port, Watson, and Foyle, J. Acoust. Soc. Am. 83, 1113–1145 (1988)] and discrimination of vowel formants [Kewley-Port, J. Acoust. Soc. Am. 110 (2001)], have also demonstrated the systematic and profound effects of higher levels of stimulus uncertainty. This presentation will discuss extensions of the concept of stimulus uncertainty that demonstrate the degrading effects of the variability in more natural speech (versus synthetic speech) and longer phonetic context (including sentences) on vowel formant discrimination. Results from normal-hearing and hearing-impaired listeners demonstrating similar detrimental effects of high stimulus uncertainty will also be presented. [Research supported by NIH-NIDCD.]

1:40

4pP3. Molecular psychophysics and sound-source identification. Robert A. Lutfi (Dept. of Commun. Disord. and Waisman Ctr., Univ. of Wisconsin, Madison, WI 53706, ralutfi@wisc.edu)

Threshold and d-prime measures of performance are ubiquitous in psychophysics. Yet, because these measures require averaging over many responses they can conceal important aspects of the subject’s decision process as it is reflected in the data from trial to trial. Chuck Watson clearly demonstrated this some 4 decades ago in his Ph.D. thesis. He was an early advocate of a “molecular” approach to psychophysics that attends specifically to the relation between individual stimuli and their associated response on each trial. In recent years, a variant of the molecular approach, perturbation analysis, has been applied with great success to the problem of image identification in vision [J. Vision 2(1) (2002), special issue]. This talk reviews the application of this method to a similar longstanding problem in auditory psychophysics—the identification of sound sources. Published and previously unreported studies are presented that use synthesized sounds to investigate listener identification of the material, geometric, and driven properties of simple resonant sources—stretched membranes, clamped bars, and suspended plates. These studies show that when listeners are confronted with a complex identification task, for which there are multiple sources of acoustic information, they regularly adopt different decision strategies that yield the same level of identification accuracy (same d-prime). [Work supported by NIDCD.]
Humans exhibit a remarkable ability to segregate sounds produced by multiple sources that overlap in frequency and time. It is likely that other species that use hearing also have this ability. To investigate the neural mechanisms that contribute to spectral segregation, responses of auditory nerve fibers (ANFs) and inferior colliculus (IC) neurons to harmonic complex tones and to the same tones with a mistuned component were measured. Mistuning leads to the perception of a new sound source and also produces dramatic qualitative changes in the temporal discharge patterns of IC neurons. In contrast, the same stimulus manipulation produces only modest quantitative changes in the responses of ANFs. These results indicate that the processing of complex tones undergoes a major transformation in the lower brainstem that is likely to contribute to perceptual segregation based on harmony. A computational model has been developed to investigate integrative mechanisms that may underlie this transformation. The model reproduces the distinctive discharge patterns of IC neurons, and suggests that these patterns arise as a result of narrow-band envelope extraction, followed by broadband excitatory–inhibitory interactions. Specific predictions of the model have been confirmed in subsequent electrophysiological experiments, providing further support for this interpretation. [Work supported by NIDCD.]

Chuck Watson was among the first in the psychoacoustic community to seriously address the topic of individual differences. At a time when there was little concern with variation among “normal listeners” in psychoacoustic research, Watson began a research program to document the range of human auditory abilities. The primary goals were to determine the number of distinct abilities, to specify the nature of each ability, and to document the distribution of these abilities in the general population. Thanks to Watson’s talent for organizing and directing large-scale projects and his workmanlike approach to science, a large and valuable body of data on human individual differences has been collected. The research program began about 20 years ago with the study of basic auditory abilities, and it has expanded to include other modalities and cognitive/intellectual abilities in adults and children. A somewhat biased view of the importance of this work will be presented by one of Watson’s many colleagues in this endeavor. The talk will provide an overview of this ongoing research program as well as a brief review of some related research by other investigators. New findings from recent extensions of this work will also be discussed.

Charles Watson’s studies of informational masking and the effects of stimulus uncertainty on auditory perception have had a profound impact on auditory research. His series of seminal studies in the mid-1970s on the detection and discrimination of target sounds in sequences of brief tones with uncertain properties addresses the fundamental problem of extracting target signals from background sounds. As conceptualized by Chuck and others, informational masking results from more central (even “cognitive”) processes as a consequence of stimulus uncertainty, and can be distinguished from “energetic” masking, which primarily arises from the auditory periphery. Informational masking techniques are now in common use to study the detection, discrimination, and recognition of complex sounds, the capacity of auditory memory and aspects of auditory selective attention, the often large effects of training to reduce detrimental effects of uncertainty, and the perceptual segregation of target sounds from irrelevant context sounds. This paper will present an overview of past and current research on informational masking, and show how Chuck’s work has been expanded in several directions by other scientists to include the effects of informational masking on speech perception and on perception by listeners with hearing impairment. [Work supported by NIDCD.]

I met and started working with C. S. Watson on or about the Fall of 1958. Even then there was usually a pause and a wind-up before he made a point, usually neither concise nor terse. But once made, a point would be repeated and repeated. These oft-repeated points transform and become “Watson’s Precepts.” Several such precepts, related to the study of speech perception by the hearing-impaired, will be quoted or paraphrased and their significance illustrated with samples from his work or that of others. Included on the list are the following. “Bryan and Harter found that learning to perceive Morse Code continued over many hundreds of hours of practice. Training in many auditory tasks may require similar investments of practice time.” “One must distinguish response proclivities and sensory capabilities and tasks with low- or high-uncertainty stimulus conditions.” “Psychoacoustic measures, other than the audiogram, don’t correlate with speech perception measures.” Finally, it will be revealed how this collection of “precepts” has led to his current views of speech perception by the hearing impaired and to our current collaboration on auditory training for the hearing-aid users which features extensive practice and transition from low- to high-uncertainty stimulus conditions.
4:00


After years of research on laboratory-generated complex sounds, in the early 1990s Chuck Watson and colleagues in the Hearing and Communications Laboratory (HCL) became interested in whether sounds with some meaning to the listener were processed differently by the auditory system. So began in his lab a program of environmental sounds research, in the meticulous, deliberate manner Watson was known for. The first step was developing an addition to the Test of Basic Auditory Capacities (TBAC) which would measure individual differences in the identification of familiar environmental sounds. Next came the psychophysical basics: detection and identification in noise. Then, borrowing a page from early speech researchers, the effects of low-, high-, and bandpass filtering on environmental sounds were investigated, as well as those of processing environmental sounds using vocoder methods. Work has continued outside the HCL on developing a standardized canon of environmental sounds for generalized testing, with an aim to creating diagnostic tests for environmental sounds similar to the SPIN and modified rhyme and reverberation (MRRT).

4:15

4pPP9. Selective auditory attention to features of complex sounds: A comparative approach. Eduardo Mercado III (Dept. of Psych., Univ. at Buffalo, SUNY, Park Hall, Buffalo, NY 14260, emiii@buffalo.edu) and Itzel Orduna (Rutgers Univ., New Jersey, NJ 07102)

When listeners are trained to respond based on one spectrottemporal component of a complex sound, enhanced processing of the behaviorally relevant feature provides an objective correlate of selective attention [I. J. Hirsh and C. S. Watson, Annu. Rev. Psychol. 47, 461–484 (1996)]. To study this issue in a nonhuman species, rats were trained to classify multidimensional acoustic stimuli based on the rate, direction, and range of frequency modulation. Rats successfully learned to classify complex sounds along the dimensions of rate and direction of frequency modulation, but not based on the range of frequency modulation. Rats classified stimuli most accurately when the relevant dimension was rate of frequency modulation. The relative ease with which rats learn to classify complex sounds along a particular dimension can be predicted based on how auditory cortical neurons in rats respond to such sounds. These findings provide new insights into how neural processing may constrain selective auditory attention to features of complex sounds. [Work supported by NIH.]

4:30

4pPP10. Listening weights for signals and maskers with uncertain frequency in normal-hearing and hearing-impaired listeners. Joshua M. Alexander and Robert A. Lutfi (Dept. of Commun. Disord. and Waisman Ctr., Univ. of Wisconsin, 1500 Highland Ave., Madison, WI 53705, jmalexai@wisc.edu)

The effect of signal and masker uncertainty on listener decision weights was measured in 12 normal-hearing (NH) and 6 hearing-impaired (HI) listeners. The signal was a tone of 0.8, 2.0, or 5.0 kHz. The maskers were fixed-frequency tones separated from the signal(s) by two-third octaves and played simultaneously with the signal(s). Each masker tone had an independent probability of occurrence of 0.5 on each trial. In the signal-uncertain (SU) condition one of the three signals was played at random with equal probability on each trial. In the signal-certain (SC) condition the signal frequency was constant on each trial. In each condition the relative influence of each signal and masker frequency on listener decisions was estimated from regression weights relating listener responses to the presence or absence of each tone on every trial. For SC, the results indicate that NH and HI listeners put significantly greater weight on the signal frequencies than the masker frequencies. For SU, however, HI listeners tended to weight a narrower spectral region compared to NH listeners. It is concluded that HI listeners are able to selectively attend to an individual signal better than they can divide their attention among multiple signals. [Work supported by NIDCD.]

4:45

4pPP11. Contributions of internal noise and Bernoulli variance to the variability in multiple estimates of d’. Walt Jesteadt (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, jesteadt@boystown.org)

Multiple estimates of d’ obtained from the same observer will vary as a result of differences in attention and other sources of internal noise, but also as a result of the variances associated with the two proportions that contribute to each d’ estimate. This second source, known as Bernoulli variance, causes the expected variance of d’ to vary a function of the true value of d’. Because estimates of d’ obtained from 2 × 2 matrices are discrete rather than continuous, the expected variance of the estimates cannot be specified by an equation. Miller [Percept. Psychophys. 58, 65 (1996)] has presented a method for computation of the sampling distribution of d’, for any true value of d’ and any given number of trials, and has demonstrated that a well-known approximation greatly exaggerates the variance for large values of d’. In the current paper, the standard approximation and Miller’s exact method are extended from Yes–No to the more commonly used 2IFC procedure and the effects of two standard corrections for zero cells are examined. A comparison of the theoretical results to actual data suggests that Bernoulli variance plays a greater role than internal noise in determining the variability in d’ estimates.

5:00

4pPP12. Informational masking of speech-analog signals: Independence of prosodic-rhythmic and formant trajectory cues. Pierre Divenyi (Speech and Hearing Res., VA Medical Ctr., Martinez, CA 94553, pdivenyi@ebihre.org)

One factor of poor speech understanding in cocktail-party settings is masking of information in a target stream by similar concurrent information in an interfering background stream. To overcome this deficit, the information in the two streams must be different along at least some acoustic dimensions. The present study investigated whether informational differences along various dimensions between a speech-analog target and an interfering distractor stream were processed independently from one another, or whether dividing one’s attention between the dimensions affects performance. In a single-interval, forced-choice experiment, listeners had to discriminate the rhythmic pattern of a burst triplet (AM) or an up–down/up–down–up 150-ms single-formant transition pattern (FM), or both, imposed on a 500-ms harmonic complex carrier, embedded in a distractor stream of random AM bursts, or constant single-formant up–down FM started in a random transition phase, or both. Fundamental frequencies of target and distractor were different. Results indicate that, when compared to discrimination of either the AM or the FM pattern alone, simultaneous discrimination of the AM and FM patterns did not result in a loss, suggesting that informational masking may operate independently on different dimensions. [Research supported by Grant R01-AG07998 from National Institute on Aging and by the VA Medical Research.]
Structural Acoustics and Vibration: Measurements and Transducers

Kenneth D. Frampton, Chair
Department of Mechanical Engineering, Vanderbilt University, Box 1592, Station B, Nashville, Tennessee 37235-1592

Contributed Papers

1:30

4pSA1. Noise analysis of a simplified Michelson interferometer vibrometer hydrophone. Lee E. Estes and Benjamin A. Cray (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841-1708)

The use of a Michelson interferometer to sense acoustic vibrations imposed on a water/plate/air boundary is investigated. To focus on fundamentals, a simplified interferometer is considered rather than the usual heterodyne or dual track homodyne configurations. Based on analytical models, the sensitivity limits due to photoelectron shot noise, laser light amplitude and phase noise, atmospheric turbulence, thermal vibration, and amplifier noise are predicted. Techniques for mitigation of turbulence and laser noise will be discussed.

1:45


Laser Doppler vibrometers (LDV) are designed to measure structural vibration velocity by sensing the phase shift in the laser signal reflected from a vibrating source. It is known that index of refraction modulations resulting from acoustic pressure distributions along a laser light path will also cause a phase shift. Simpson et al. [J. Acoust. Soc. Am. 99(4), 2521(A) (1996)] have investigated this acousto-optic phase modulation as a possible contaminating effect for underwater LDV vibration measurements. This paper will investigate acousto-optic phase modulations measured by a scanning LDV as a method for measuring pressure radiating from underwater vibrating surfaces. This is done by passing the laser beam through the radiating pressure field and measuring the backscattered laser signal which is reflected off a rigid and retroreflective surface (outside the pressure field). It is shown experimentally, using the average pressure measured with an LDV over a plane in the vicinity of a vibrating structure, that the pressure at a far-field location normal to the plane can be determined.

2:00

4pSA3. Visualization of the energy flow for guided forward and backward waves in and around a fluid-loaded elastic cylindrical shell via the Poynting vector field. Cleon E. Dean (Phys. Dept., Georgia Southern Univ., P.O. Box 8031, Statesboro, GA 30460-8031, cdean@GeorgiaSouthern.edu) and James P. Braselton (Georgia Southern Univ., Statesboro, GA 30460-8093)

Color-coded and vector-arrow grid representations of the Poynting vector field are used to show the energy flow in and around a fluid-loaded elastic cylindrical shell for both forward- and backward-propagating waves. The present work uses a method adapted from a simpler technique due to Kaduchak and Marston [G. Kaduchak and P. L. Marston, “Traveling-wave decomposition of surface displacements associated with scattering by a cylindrical shell: Numerical evaluation displaying guided forward and backward wave properties,” J. Acoust. Soc. Am. 98, 3501–3507 (1995)] to isolate unidirectional energy flows.

2:15

4pSA4. Cylindrical transducers to generate and detect axisymmetric waves in a pipe. Jin O. Kim, Kyo-Kwang Hwang, Jung-Goo Lee, and Hyung-Gon Jeong (Soongsil Univ., 1 Sangdo-dong, Dongjak-gu, Seoul 156-743, Korea, jokim@ssu.ac.kr)

This paper presents the radial vibration characteristics of piezoelectric cylindrical transducers and the application of the transducers to the generation and detection of axisymmetric longitudinal waves in a pipe. Dynamic differential equations of piezoelectric radial motion derived in terms of radial displacement and electric potential and mechanical and electrical boundary conditions have yielded a characteristic equation for radial vibration of the radially polarized piezoelectric cylinder. Theoretical calculations of the fundamental natural frequency have been compared with numerical and experimental results for transducers of several sizes, and have shown a good agreement. It has been shown that the piezoelectric natural frequency of the fundamental mode for radial vibration in a cylindrical transducer depends mostly on the radius rather than on the thickness of the cylinder. Experiments have been performed in empty and water-filled pipes equipped with the transducers that were used for transmitting and receiving axisymmetric elastic waves in the pipe wall. The measured wave speeds have been compared with the analytical ones. This work has demonstrated the feasibility of using cylindrical transducers and pipe waves for the determination of the mass density and, eventually, the flow rate of the liquid in a pipe.

2:30

4pSA5. Actuator/sensor placement optimization for vibration control based on finite element model of the car chassis. Bouzid Sebu, Nikola Nedeljkovic, and Boris Lohmann (Inst. of Automation, Bremen Univ., Bremen 28359, Germany)

The finite element method is a convenient approach to model the structures with complex geometry and subjected to complicated boundary conditions. However, the order of the obtained model using FEM strategy is large. Therefore, the control design and implementation based on the FEM model turns out to be complicated (sometimes not feasible). Hence, the order reduction techniques are inevitable to reduce the size of the system obtained with FEM. The reduced model is then used to explore and to optimize the system in terms of collocated actuator/sensor positioning problem. In this paper, we suggest a strategy to optimize the location of the electromechanical actuators in the car chassis subframe for vibration control. The proposed method can be formulated using an optimization problem with constraints which are introduced to consider the spill over effect resulted from the neglected higher modes in the model reduction part (which affects the stability of the closed loop system) and to ensure the minimum controllability of individual modes. The optimization algorithm is used to select three optimal positions of the collocated actuator/sensor among 298 possible positions.

The analysis of sound radiation with steady-state vibration of a steel plate with the system of piezoceramic elements was performed. The system of piezoceramic actuators consists of four pairs PZT4 squared elements with dimensions $10 \times 10 \times 1$ mm$^3$. The actuators were located symmetrically on the plate and driven 180 deg out of phase with the same signal. It was assumed that rectangular plate is fixed along one edge. Other edges have the free boundary conditions case. The structural vibration analysis of a plate with a system of piezoelectric actuators was performed with FEM ANSYS® code for the first three modes. The hybrid [Kozien and Wiciak, Quantum Mol. Acoust. 24, 98–110 (2003)] and fluid structure interactive methods were applied to estimate the sound pressure radiated by a vibrating system plate and actuators. Analysis of the SPL radiated by a vibrating plate with piezoceramic actuators into chosen points in the acoustic field gave similar results with two applied methods. The small differences in results were possibly an effect of the boundary assumptions. The best results in the reduction of sound pressure by plate were obtained for the first mode. [Work supported by SCFIR.]

3:00–3:15 Break

3:15


Previous research has demonstrated the utility of acoustic energy density measurements as a means to gain a greater understanding of acoustic fields. Three spherical energy density probe designs are under development. The first probe design has three orthogonal pairs of surface mounted microphones. The second probe design utilizes a similarly sized sphere with four surface mounted microphones. The four microphones are located at the origin and unit vectors of a Cartesian coordinate system, where the origin and the tips of the three unit vectors all lie on the surface of the sphere. The third probe design consists of a similarly sized sphere, again with four surface microphones, each placed at the vertices of a regular tetrahedron. The sensing elements of all three probes are Panasonic electrolyt microphones. The work presented here will expand on previously reported work, and address bias errors, spherical scattering effects, and practical implementation issues. [Work supported by NASA.]

3:30


Forward wave propagation to the farfield using nearfield pressure measurements has undergone extensive development. Up until now, discrete pressure measurements are typically made using a conformal array of microphones. However, complex acoustic fields require large microphone arrays in order to accomplish accurate field reconstruction. The use of energy density sensors has indicated several advantages over conventional pressure microphones in active noise control applications. This study examines possible benefits of energy-based measurements in free field reconstruction. To determine and quantify the possible advantages of energy density measurements, an analytical model was built to evaluate the performance. The error in acoustic fields reconstructed using energy density sensors is compared with conventional pressure microphone reconstruction accuracy. A number of analytical studies will be presented showing the benefits of using energy-based measurements in free field reconstruction.

4pSA9. Detecting and classifying adhesive flaws between bonded elastic plates. Ricardo Leiderman (Dept. Engenharia Mecânica, Pontificia Universidade Catolica do Rio de Janeiro, PUC-Rio, Brazil) and Paul E. Barbone (Boston Univ., Boston, MA 02215)

Nondestructively evaluating the quality of an adhesive bond is challenging. The reason is that the only way to measure bond strength is by measuring the force required to break a bond, after which the bond is broken. Localized adhesion flaws that diminish bond strength, however, also tend to diminish bond stiffness. For example, pockets of elevated porosity, microcracking, or other damage in the adhesive layer will simultaneously lower local bond strength and local effective bond stiffness. So guided, we formulate and solve an inverse scattering problem to reconstruct the effective stiffness and compliance of an adhesive layer in a layered elastic plate. Our formulation is based on the method of invariant embedding, and applies to isotropic and anisotropic elastic layers. We present two solutions of the inverse problem: the Born approximation and the exact solution. Both solutions are unstable, as is the nature of inverse scattering problems, and require some regularization in the presence of noise. We present computed examples and discuss the role of regularization.

4:00

4pSA10. Assessment of material fatigue damage using nonlinear vibro-modulation technique. Andrei Zagrai (Davidson Lab., Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, azagrai@stevens.edu), Dimitri Donskoy (Intelligent Sensing Technologies, LLC, Fair Haven, NJ 07704-6457), Alexander Chudnovsky, and Hudson Wu (Univ. of Illinois at Chicago, Chicago, IL 60607)

Heavy periodic loads exerted on structural materials often lead to fatigue damage (material degradation at microscale) which may finally trigger irreversible fracture process. Conventional NDT techniques detect only the latter, and there is an increasing need for new tools to assess fatigue damage at the earliest possible stage, i.e., before fracture. This paper presents experimental results of early damage characterization using an innovative nonlinear vibro-modulation technique (VMT) [Donskoy et al., NDT&E Int. 34 (2001)]. In the experiments, fatigue damage was initiated in steel, aluminum, and carbon–carbon composite specimens during strain-controlled three-point bending high-cycling fatigue tests. The damage progress was independently monitored using dataflow from the testing machine and the real-time nonlinear vibro-modulation measurements. The tests demonstrated that the reduction in the specimens' stiffness (direct indication of damage accumulation) correlates well with the increase in the VMT's nonlinear damage index. These results confirm that VMT could offer new opportunities for early damage detection and remaining life prediction. [Work supported by NAVAIR.]

4:15


The author applied modal analysis and energy flow methods for detection of delaminating plate delaminating. New design of hardware for field measurements especially for conservation of monuments of history was developed. The author also tested two new noninvasive measurement methods. The first one is based on measurement of electric field dynamic parameters in the vicinity of subsurface delaminating and the second on the measurement of the frequency response of a structure being excited by the diffusive acoustic near field of the radiator. Main application of developed devices was use in a wide band of practical applications in detection of disassembling of precious wall paintings — frescoes. All laboratory conditions were carried out with designed devices on fresco samples prepared due to old technologies. Results proved satisfactory sen-
sitivity of the devices for practical application to conservator team works. The assessment of a structure technical state is performed on the basis of real-time calculations of the quality measures and damage probability. The proposed device enables the investigator to identify fresco delaminating of 8-cm diameter located at a depth of 5 mm. Discussed methods can be useful in diagnostics of many other multilayer structures.

THURSDAY AFTERNOON, 27 MAY 2004

CONFERENCE ROOM E, 1:40 TO 3:30 P.M.

Session 4pSP

Signal Processing in Acoustics: Sensor and Array Processing

Paul Hursky, Chair

SAIC Ocean Sciences Division, 10260 Campus Point Drive, San Diego, California 92121

Chair’s Introduction—1:40

Contributed Papers

1:45

4pSP1. Model-based broadband towed-array processing. Edmund J. Sullivan (OASIS, Inc., 5 Militia Dr., Lexington, MA 02421) and Geoffrey S. Edelson (BAE SYSTEMS, Nashua, NH 03061-0868)

Conventional broadband towed array processors operate in the frequency domain and exploit the phases of the frequency components from a DFT to extract the bearing estimate. In this process, the fact that the array is moving is not taken into consideration. In fact, to first order, the effect of the motion is negligible. However, it has been shown that, in the narrow-band case, the effect of the motion on the variance of the bearing estimate is not negligible. This is due to the fact that the Doppler incurred by the motion itself carries bearing information. In this work, it is shown that a recursive bearing estimation scheme, based on sequential DFT’s, can produce the so-called synthetic aperture effect, where the Cramer–Rao lower bound on the bearing estimate progressively decreases, as compared to the case where the motion is not explicitly included in the processing scheme. Examples of this effect are shown based on simulated data, and the capabilities and limitations of the technique are discussed.

2:00

4pSP2. Optimal sensor placement in highly variable noise fields to increase detection. Donald R. DelBalzo and Erik R. Rike (Neptune Sci., Inc., 40201 Hwy. 190 E., Slidell, LA 70461, delbalzo@neptunesci.com)

Ambient noise (AN) is highly variable in littoral environments. AN predictions are often smoothed to produce mean estimates in support of signal detection. In a complex AN environment, the optimal placement of sensors for detection also depends on the spatial and temporal variance of AN. The recent development of the Dynamic Ambient Noise Model (DANM) allows accurate, detailed estimation of noise statistics. DANM was used to produce space- and time series of directional noise based on discrete ship tracks. This work describes a fast optimization scheme that exploits nonhomogeneous noise variability to select optimal locations that maximize detection. A standard, nonoptimal approach for choosing locations is sequential and myopic. A better approach is combinatorial; i.e., consider all possible sensor sets and choose the best. This optimal approach is often computationally prohibitive. Brown’s iterative algorithm combines elements of both algorithm types to achieve rapid results. Brown’s algorithm was modified and used to demonstrate the importance of nonhomogeneous noise statistics. The results show a significant improvement in probability of detection when DANM AN statistics are considered while choosing optimal sensor locations. This work has application to design of optimal sonobuoy field patterns. [Work sponsored by ONR under the LADC project.]

2:15

4pSP3. Blind deconvolution for non-stationary acoustical systems. Mark R. Gramann, Josh G. Erling, and Michael J. Roan (Appl. Res. Lab, Penn State Univ., P.O. Box 30, State College, PA 16804, mrg227@psu.edu)

Blind deconvolution is a signal-processing technique that has been shown to be highly effective for removing multipath propagation channel-induced corruption of a signal. These algorithms are usually implemented as adaptive filtering processes that learn finite impulse response inverse filters based on measurements of the corrupted signal. A major limitation of these techniques is that a stationary propagation environment is generally assumed. This assumption is not valid for many real-world applications, such as active or passive sonar operating in a shallow-water environment or for communications systems with moving sources or receivers. For slowly varying systems, standard algorithms with adaptive learning rates that improve convergence speed, such as frequency-domain implementations of the Infomax and Natural Gradient algorithms, may be sufficient. However, in systems where either the input signal or impulse response change rapidly, it becomes necessary to provide a means for tracking the statistical changes that occur in the input signal or impulse response. Approaches to tracking these changes through the use of a tracking algorithm such as Kalman filtering are discussed. [Work supported by Dr. David Drumheller, ONR Code 333, Grant No. N00014-00-G-0058.]

2:30

4pSP4. Free-field equivalent shape of scattering conformal microphone arrays. Philippe Moquin and Stephane Dedieu (Mitel Networks, 350 Legget Dr., Kanata, ON K2K 1X3, Canada)

When an array is placed upon an acoustic scatterer, the interelement spacing of the array appears to be different from free field conditions [J. Meyer, J. Acoust. Soc. Am. 109, 185–193 (2001)]. This manifests itself primarily as a nonlinearity in the interelement phase difference. Ideal scatterers (rigid and nonconducting) can be used to understand these effects. A spherical scatterer with a circular array serves as a first example since it has an analytical solution. To help understand the effect, equivalent shape of a free-field array can be calculated. Numerical simulation and measured results on a telephonelike shape with an elliptical array will serve to illustrate that the results from the sphere can be used as a guide for estimating the scattering effects that can be expected in a more complex shape.
A four-microphone array and signal-processing card have been integrated with a handheld computer such that the integrated device can be used with large-scale sensor networks using a decentralized computing approach. This algorithm, based on a time delay of arrival (TDOA) method, uses information from the minimum number of sensors necessary for an exactly determined solution. Since the algorithm is designed to run on computational devices with limited memory and speed, the complexity of the computations has been intentionally limited. The sensor network consists of an array of battery-operated COTS Ethernet-ready embedded systems with an integrated microphone as a sensor. All solutions are calculated as distinct values, and the same TDOA method used for solution is applied for ranking the accuracy of an individual solution. Repeated for all combinations of sensor nodes, solutions with accuracy equivalent to complex array calculations are obtainable. Effects of sensor placement uncertainty and multipath propagation are quantified and analyzed, and a comparison to results obtained in the field with a large array with a centralized computing capability using a complex, memory intensive algorithm is included.

3:00


A four-microphone array and signal-processing card have been integrated with a handheld computer such that the integrated device can be carried in and operated with one hand. Automatic speech recognition (ASR) was added to the USAMRMC/TATRCs Battlefield Medical Information System (BMIST) software using an approach that does not require modifying the original code, to produce a Speech-Capable Personal Digital Assistant (SCPDA). Noise reduction was added to allow operation in noisier environments, using the previously reported Hybrid Adaptive Beamformer (HAB) algorithm. Tests demonstrated benefits of the array over the HP/COMPAQ-iPAQ built-in shielded microphone for noise reduction and automatic speech recognition. In electroacoustic and human testing including voice control and voice annotation, the array provided substantial benefit over the built-in microphone. The benefit varied from about 5 dB (worst-case scenario, diffuse noise) to about 20 dB (best-case scenario, directional noise). Future work is expected to produce more rugged SCPDA prototypes for user evaluations, revise the design based on user feedback and real-world testing, and possibly to allow hands-free use by using ASR to replace the push-to-talk switch, providing feedback audibly and/or via a head-up display. [Work supported by the U.S. Army Medical Research and Materiel Command (USAMRMC), Contract No. DAMD17-02-C-0112.]

3:15

4pSP7. Plane-wave decomposition by spherical-convolution microphone array. Boaz Rafaely (Elec. and Computer Eng. Dept., Ben-Gurion Univ., Beer-Sheva 84105, Israel, br@ee.bgu.ac.il) and Munhum Park (Univ. of Southampton, Southampton SO17 1BJ, UK)

 Reverberant sound fields are widely studied, as they have a significant influence on the acoustic performance of enclosures in a variety of applications. For example, the intelligibility of speech in lecture rooms, the quality of music in auditoria, the noise level in offices, and the production of 3D sound in living rooms are all affected by the enclosed sound field. These sound fields are typically studied through frequency response measurements or statistical measures such as reverberation time, which do not provide detailed spatial information. The aim of the work presented in this seminar is the detailed analysis of reverberant sound fields. A measurement and analysis system based on acoustic theory and signal processing, designed around a spherical microphone array, is presented. Detailed analysis is achieved by decomposition of the sound field into waves, using spherical Fourier transform and spherical convolution. The presentation will include theoretical review, simulation studies, and initial experimental results.

THURSDAY AFTERNOON, 27 MAY 2004

NEW YORK BALLROOM B, 1:25 TO 5:00 P.M.

Session 4pUW

Underwater Acoustics: Acoustic Propagation and Modeling

Michael Brown, Chair

RSMAS-AMP, University of Miami, 4600 Rickenbacker Causeway, Miami, Florida 33149-1098

Chair’s Introduction—1:25

Contributed Papers

1:30

4pUW1. Focusing at an arbitrary waveguide location using time reversal. Shane C. Walker, Philippe Roux, and W. A. Kuperman (Marine Physical Lab., Scripp’s Inst. of Oceanogr., UCSD, 8820 Shellback Way, La Jolla, CA 92039, shane@physics.ucsd.edu)

A method for producing acoustic time reversal focusing at an arbitrary location is presented. In principle, producing a focus at any location in a waveguide is trivial once the point to point propagation matrix is known. In practice, the propagation matrix over a given range, $G(z_s,z_r,R)$ between sources $z_s$ at $r=0$ and receivers $z_r$ at $r=R$, can be directly measured with a pair of vertical line arrays. Once known, repeated iterations of $G(z_s,z_r,R)$ result in good approximations to the propagation matrices, $G(z_s,z_r,mR)$, for ranges $mR$, where $m=2,3,\ldots$ is the number of iterations applied. By combining this iteration technique with frequency-dependent range shifting, it is possible to produce acoustic time reversal focusing at any range and depth.

1:45

4pUW2. Diffraction of nonuniform sound in the sea. John L. Spiesberger (Dept. of Earth and Environ. Sci., 240 S. 33rd St., Univ. of Pennsylvania, Philadelphia, PA 19104, johnsr@sas.upenn.edu)

A theory of diffraction is applied to nonuniform sound emissions of both long and short wavelengths. The ray approximation is sometimes good and other times not so good. Some theories for the scattering of nonuniform sound emissions of sound assume that propagation occurs within a Fresnel zone
of the ray path. This assumption is incorrect. Indeed, the use of a correctly sized region based on a theory of diffraction appears to be able to help resolve major discrepancies between observations and such theories for acoustic fluctuations. Indeed, the correctly sized region can be orders of magnitude smaller than given by a Fresnel radius.

2:00
4pUW3. Ray and travel time stability in weakly range-dependent sound channels. Francisco J. Beron-Vera and Michael G. Brown (RSMAS-AMP, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149)

Ray path and travel time stability are investigated in environments consisting of a range-independent background on which a weak range-dependent perturbation is superimposed. Theoretical arguments suggest and numerical results confirm that both ray path and travel time stability are strongly influenced by the background sound speed profile. Both ray path and travel time stability are shown to increase with increasing magnitude of \( a(l) = (l/\alpha) \omega \alpha / l \), where \( 2\pi/\omega(l) \) is the range of a ray cycle (double loop in a deep ocean sound channel) and \( I \) is the ray action variable. This behavior is illustrated using internal-wave-induced scattering in deep ocean environments and rough surface scattering in upward refracting environments. [Work supported by ONR.]

2:15
4pUW4. Wavefield stability in weakly range-dependent sound channels. Michael G. Brown, Francisco J. Beron-Vera, Irina Rypina, and Ilya Udovychchenko (RSMAS-AMP, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149)

It is shown that the mode-based “waveguide invariant” \( \beta \) is asymptotically equivalent to the ray-based “stability parameter” \( \alpha \). These parameters are known to control the dispersive properties of the sound channel and various measures of both ray path and travel time stability. This leads to the expectation that finite-frequency wavefield stability is also largely controlled by \( \beta \) (or \( \alpha \)). PE simulations are shown to confirm this expectation. [Work supported by ONR.]

2:30
4pUW5. Three-dimensional wide-angle azimuthal PE solution to modified benchmark problems. Li-Wen Hsieh, Ying-Tsong Lin, and Chi-Fang Chen (Dept. of Eng. Sci. and Ocean Eng., Natl. Taiwan Univ., No. 73 Chou-Shan Rd., Taipei, Taiwan 106, R.O.C., rhs@ms11.url.com.tw)

In predicting wave propagation, the size of the angle of propagation plays an important role; thus, the concept of wide angle is introduced. Most existing acoustic propagation prediction models do have the capability of treating the wide angle but the treatment, in practice, is vertical propagation angle. This is desirable for solving 2D \((r-z)\) problems. Typically, 3D problems are dealt with an \( N \) by 2D approximation. To deal with problems possessing 3D effects, the azimuthal coupling terms have to be considered in PE approximation. Moreover, in extending the 2D treatment to 3D, the wide-angle capability is maintained in most 3D models, but it is still vertical. Hence, the concept of wide angle is introduced in the azimuthal direction to enhance the capability of predicting the azimuthal coupling and thus the 3D effects. A truncated wedge-shaped ocean which is modified from ASA benchmark problems is used in this study. The results show apparent 3D effects and validate the 3D wide-angle azimuthal PE model, a wide-angle version of FOR3D.

2:45

A three-dimensional split-step Pade parabolic-equation approach [J. Comput. Acoust. 9(2), 17–39 (2001)] was developed as an extension to Collins and Chin-Bings algorithm [J. Acoust. Soc. Am. 87, 1104–1109 (1990)] to account for azimuthal acoustic refraction via a narrow-angle approximation. It was shown that high accuracy requires fine azimuthal and range step sizes. Therefore, a couple of Pade terms are sufficient given the required small range step size. The paper also mentions the possible implementation of a variable azimuthal grid size, where the number of azimuthal segments increases proportionally with increasing range, as another approach to boost the models computational speed. In this talk, we investigate the possibility of azimuthal padding as another approach to speed-up the model when the user is only interested in propagating the acoustic field along a wedge of azimuthal angles instead of the entire 360.

3:00
4pUW7. A variable rotated parabolic equation for elastic media. 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)

There are several approaches for improving the accuracy of the parabolic equation method for problems involving sloping interfaces. Energy-conservation and single-scattering corrections have essentially resolved the acoustic problem, but these approaches have been less effective for problems involving elastic layers. Some promising results have recently been obtained using approaches based on mapped [J. Acoust. Soc. Am. 107, 1937–1942 (2000)] and rotated [J. Acoust. Soc. Am. 87, 1035–1037 (1990)] coordinates. The latter of these approaches provides greater accuracy for problems involving relatively steep interfaces. With the extension to variable slopes [J. Acoust. Soc. Am. 114, 2428–2429 (2003)], the rotated parabolic equation became a useful approach for solving a large class of range-dependent problems. This presentation will describe a generalization of the variable rotated parabolic equation to elastic media. At each change in slope, a new rotated coordinate system is introduced and the solution is interpolated onto the new grid. In the elastic case, this process requires special operators for changing variables, the rotation of the dependent variables, and a rotationally invariant quantity. Examples will be presented to illustrate and test the approach. [Work supported by the ONR.]

3:15–3:30 Break

3:30
4pUW8. Broadband normal-mode computations within a multiprocessing environment. Steven A. Stotts and Field G. Van Zee (Appl. Res. Labs., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78758, stotts@arlut.utexas.edu)

A technique to facilitate normal-mode modeling within a multiprocessing environment is presented using the Message-Passing Interface (MPI) communication library. Portability and standardization were the driving factors for using MPI as the multiprocessing implementation. Using only a minimal set of commands, an example algorithm illustrates the approach as a basis to transform existing models to perform computations via multiprocessors. Applications to propagation modeling are given with emphasis on broadband normal-mode computations. A static approach to equalizing processor workload for normal-mode computations utilizes the linear dependence of mode number with frequency. The area under a curve representing the mode number versus frequency is used to determine the loading. An assumption of equal area implying equal loading determines the beginning and ending frequency increments for each processor. Comparisons with other approaches to processor loading for optimizing speedup time are given. A determination of the optimal number of processors for several examples of broadband normal-mode computations using a subroutine version of the Navy Standard NAUTILUS will be presented. Implementing this approach into more advanced models such as geoaoustic inversions will be discussed.
A steady-state 3D finite-element software called FESTA (Finite-Element STructural Acoustics), is being developed at the NATO Undersea Research Centre. The code is geared towards a variety of applications in underwater acoustics, such as multistatic scattering from localized inhomogeneities, scattering across interfaces between fluids and/or solids, and multistatic scattering from single and multiple fluid-loaded elastic targets. One issue of importance to researchers in underwater acoustics is the trade-off between full 3D models and thin-shell theory. To address this issue, FESTA predicts that the importance of the rescattering depends on seabed characteristics of the target via its error bar are estimated for the Continental Shelf region in the South China Sea.