

Session 3aAA

Architectural Acoustics: Integration of Synthesis Techniques and “Acoustical” Music

Richard H. Campbell, Cochair

Bang-Campbell Associates, Box 47, Woods Hole, Massachusetts 02543

K. Anthony Hoover, Cochair

Cavanaugh Tocci Associates, Inc., 327F Boston Post Road, Sudbury, Massachusetts 01776

Chair's Introduction—8:25

Invited Papers

8:30

3aAA1. The real time use of signal processing in brass performance. Thomas J. Plsek (Berklee College of Music, 1140 Boylston St., Boston, MA 02215)

The real time use of signal processing for brass instrument performance (specifically the trombone) provides an opportunity, as well as challenges, to create a type of hyper-instrument quite unlike most conceptions of the term. These include the expansion of the available sound palette, the problem of real time parameter control, microphone techniques, and repertoire considerations. The various types of processing used includes reverb, pitch change, delays—short time and longer looping types, distortions, and combinations of the above. One of the notions that must be accepted is that, unlike instruments whose sound is produced electronically, the processed signal, unless significantly long-delay-based processing is used, is always layered onto an acoustic output. By creatively managing the acoustic instrument and the electronic equipment, a vast array of musical resources become available to the performer enabling him/her to enhance existing performance environments, as well as find and develop new ones.

8:50

3aAA2. New control technologies for virtual orchestra playback. David B. Smith (New York City Tech. College, 186 Jay St., Brooklyn, NY 11201), Kojiro Umezaki (Healthy Boys, L.L.C., New York, NY 10010), Frederick W. Bianchi, and Richard H. Campbell (Worcester Polytechnic Inst., Worcester, MA 01609, rhcamp@ma.ultranet.com)

A new software application running on the Linux operating system has been developed exclusively to support live performance by a virtual orchestra (VO). The core engine, which allows unlimited expansion, has a rich feature set including temporal flexibility, arbitrary performance navigation, printed score compliance, multiple input/output devices, and editing capability. The base configuration supports 64 MIDI channels and utilizes metaevent extensions to the MIDI 1.0 specification. MIDI files from multiple platforms can be imported. Considerable attention has been given to human factors in VO real-time music production with meaningful display screens and specialized keyboard designs resulting in a rapid and intuitive user learning curve. [Work supported by Realtime Music Solutions, L.L.C., New York.]

9:10

3aAA3. Synthesized space in pop music production. Alexander U. Case (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA 01776, acase@cavtocci.com)

The sonic vocabulary of the pop music recording engineer is not constrained by the architectural acoustics of any space. Equipped with racks and RAM full of signal processing tools, the engineer tries to create musical timbres, textures, ambience, and emotions to support whatever feeling the music inspires. Perhaps nowhere is this departure from the physical to the contrived more apparent than in reverb. Acoustic analogies like the chamber, mechanical simulations like the plate, and the wholly invented spaces of the digital reverb device are part of a family of effects which—inspired by the great halls and opera houses of the world—have an identity and freedom all their own. Reverb time, early decay time, bass ratio, initial time delay gap, and so on become independent variables freely manipulated by the engineer to create sonic “spaces” that may not be physically possible outside of the studio. This paper surveys contemporary pop music production trends in reverb and analyzes them through the lens of the architectural acoustician.

9:30

3aAA4. Performances of Ballet Mécanique. Paul D. Lehrman (Tufts Univ., Medford, MA)

George Antheil's 1924 Ballet Mécanique, written for two pianos, three xylophones, four bass drums, tam-tam, electric bells, three airplane propellers, and between 4 and 16 synchronized player pianos, is one of the great “lost” pieces of the 20th-century instrumental repertoire. This paper describes the efforts that were made between 1996 and 2000 to revive this remarkable piece, using modern computers, MIDI, digital samplers, and MIDI-compatible player pianos. It talks about how the project came about; the musical and technical decisions that needed to be made; the problems of transcription, synchronization, and rehearsal; and the practical, artistic, and logistical issues of presenting the piece in three different concert spaces: Durgin Hall at the University of Massachusetts Lowell, Carnegie Hall in New York, and Louise Davies Symphony Hall in San Francisco.

10:05

3aAA5. Quantitative practice for students at the Digital Conducting Laboratory. Teresa Marrin Nakra (Immersion Music and Worcester Polytech. Inst., 381 Farrwood Dr., Ward Hill, MA 01835, teresa@immersionmusic.org) and Gary Hill (Arizona State Univ.)

Students of conducting and conducting pedagogy at Arizona State University's School of Music now have the opportunity to enhance their skills as performers and teachers with the assistance of a unique technology housed in the Digital Conducting Laboratory. Utilizing a Digital Conducting Feedback System designed by Dr. Teresa Marrin Nakra of Immersion Music, Inc., the laboratory has the capability to simulate many of the behaviors of a live orchestra rehearsal setting. The system's unobtrusive sensor interface and interactive program recreates several fundamental ensemble-conductor interactions, particularly by reacting to the tempo, articulation, and dynamic line generated by the conductor. In addition to immediate aural feedback, the system allows conductors to review their performances via sound files, video playback, and analysis of muscle-tension profiles. The musical materials consist of a set of etudes that systematically take a conductor through a review of basic conducting gestures. Compiled and arranged by the laboratory's founder and director, ASU Professor Gary W. Hill, the etudes are principally derived from familiar classical music, freeing the user to focus primarily on his or her conducting and the concomitant response.

10:25

3aAA6. Interactions among acoustics, digital signal processing, and movement. Seth Cluett (64 Beacon St., Boston, MA 02108, sethcluett@yahoo.com)

A team of architects, electronic artists, and acousticians designed a system for manipulating acoustic, haptic, and spatial information based on the motion of dancers through space. The dance structure was designed to inspire physical exploration and interaction, to sense and process resultant sounds based on this dance, and to present an immersive sound field to the dancers and observers which complemented the visual impact of the movement. A microphone system captured the sounds of the dancers' contact with the system. An electronic sensor system tracked the location and action of the dancers. A set of digital signal processing algorithms analyzed, modified, and spatialized in real time the information collected from the environment, integrating sensory cues arriving simultaneously and rapidly via multiple modalities. A multichannel speaker system re-presented the natural and modified sounds to the acoustic environment, which in turn affected the dancers' movements. The nature of movement through physical space, the localization of movement in both vision and audition, and the abstraction, support, and deconstruction of both modalities were studied. The results were used to design a performance by a dance ensemble that explored the synergy between movement, multimodal sensory information, and sound.

10:45

3aAA7. Alternative voices for electronic sound. Dan Trueman (Computer Music Ctr., Columbia Univ., New York, NY 10027), Curtis Bahn (iEAR Studios, Rensselaer Polytechnic Inst., Troy, NY 12180), and Perry Cook (Princeton Univ., Princeton, NJ 08544)

Through the design and construction of unique sound diffusion structures, the nature of electronic sound can be reinvented. When allied with new sensor technologies, these structures offer alternative modes of interaction with techniques of sonic computation. This paper describes several recent applications of Geosonic Speakers (multichannel, outward-radiating geodesic speaker arrays) and Sensor-Speaker-Arrays (SenSAs: combinations of various sensor devices with outward-radiating multichannel speaker arrays). Geosonic Speakers, building on previous studies of the directivity of acoustic instruments (the NBody Project), attempt to reproduce some of the diffusion characteristics of conventional acoustic instruments; they engage the reverberant qualities of performance spaces, and allow electronic and acoustic instruments to blend more readily. In collaboration with the U.S. Enclosure Company, over a dozen Geosonic Speakers were produced, varying in size from 8 to 14 in. Their use in the performance and recording of two major new works by composer Steven Mackey is detailed. The electronic improvisation ensemble "Interface" has integrated a family of these Geosonic Speakers into their standard setup, completely replacing their previous P.A. diffusion model; this has encouraged them to substantially reinvent their approach to the performance of live interactive computer music.

Contributed Paper

11:05

3aAA8. Spectro-temporal control parameters for synthetic orchestra applications. James W. Beauchamp (School of Music and Dept. of Elec. and Computer Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801, j-beauch@uiuc.edu)

Several spectral properties can be exploited for control of individual voices in a synthetic orchestra. Sounds can be synthesized from their spectral representations using time-varying additive synthesis, inverse Fourier transform, or, more efficiently, using multiple wavetable synthesis. Primary parameters to be controlled are amplitude and spectral centroid

versus time envelopes, spectral envelope, spectral irregularity, vibrato, noise content, and inharmonicity. These parameters can be further resolved into subparameters. For example, amplitude versus time curves can be partitioned into attack, steady-state, and decay segments which can be manipulated separately. For sustained sounds, it is also important to consider how these parameters vary with fundamental frequency and intensity. In the case of percussion sounds, the density of partials as well as their spectral evolution are important. In all cases, detailed management of parameters is necessary to maintain timbral clarity, uniqueness, and naturalness. Examples of synthetic orchestral instruments will be played.

Session 3aABa

Animal Bioacoustics: Use of Acoustics for Wild Animal Surveys

David K. Mellinger, Chair

*Cooperative Institute for Marine Resources Studies, Oregon State University, 2030 SE OSU Drive, Newport, Oregon 97365***Chair's Introduction—7:30***Invited Papers*

7:35

3aABa1. Acoustic detection distances of sperm whales in the Gulf of Mexico. David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE OSU Dr., Newport, OR 97365), Aaron M. Thode (MIT, Cambridge, MA 02139), Anthony Martinez, Keith Mullin (Southeast Fisheries Sci. Ctr., Miami, FL 33149), and Sarah Stienessen (Texas A&M Univ., Galveston, TX 77551)

During a cruise in June–July 2000 in the north-central Gulf of Mexico, sperm whales were acoustically detected, located, and tracked using a small-aperture hydrophone array towed at a depth of 20–100 m. Water depth was 800–2000 m. Detection ranges to groups of sperm whales were estimated on several occasions by determining the location of a calling whale or group of whales, then moving away in a straight line until the group was undetectable. The range at which sperm whales could no longer be detected, either aurally with headphones or visually in a spectrogram, was found to be 4–6 km, significantly less than the 11 km that had been estimated previously in the Gulf [Norris *et al.*, Minerals Management Service Report 1996-0027]. To investigate this result, acoustic propagation models were run. Results from the models are shown and are used to explain the short detection distances present. In addition, propagation models are used to evaluate the effects of water depth, sensor depth, and bottom composition on detection range, which will be most useful for future attempts to predict detection ranges of sperm whale sounds. [Work supported by ONR, MMS, and NMFS.]

7:55

3aABa2. Using one or two hydrophones for marine animal surveys. Douglas H. Cato (Defence Sci. and Technol. Organisation, P.O. Box 44, Pyrmont, NSW 2009, Australia) and Robert D. McCauley (Curtin Univ. of Technol., Bentley, Western Australia)

Acoustic surveying of marine animals requires the ability to localize the sources or at least determine their distances, if the density of the animals is to be determined. Precisely placed multiple-element arrays will provide this information, but there are logistically simple techniques using one or two hydrophones that are effective under certain conditions. If the propagation conditions are known, the differences in the times of arrival and the received levels on two hydrophones provide the distance of the source and the source level, for sources significantly closer to one hydrophone than to the other. The positions of the hydrophones do not need to be known, but if they are, the sources can be localized with left–right ambiguity. An example is the use of sonobuoys deployed from a ship in transit. One hydrophone is sufficient if the direct and surface reflected paths can be separated, since the surface image can be used in place of the second hydrophone. Some examples of this applied to movement of fish sources will be presented. Complex propagation causes difficulties but these may also be exploited to improve the estimates.

8:15

3aABa3. Multi-modal surveys of fin whales in the Sea of Cortez, Mexico. Christopher W. Clark, Don A. Croll, Alejandro Acevedo, and Jorge Urban-Ramirez (Cornell Lab. of Ornithology, Bioacoustics Res. Prog., 159 Sapsucker Woods Rd., Ithaca, NY 14850, cwc2@cornell.edu)

A population of fin whales (*Balaenoptera physalus*), resident to the Gulf of California, Mexico, was studied over two seasons using an integrated approach. Systematic vessel-based visual survey and photo-ID efforts were conducted every 5–7 days to independently estimate the number and distribution of whales within a 10×30 mi² study area. Some whales were tagged with time-depth-recorders. Sets of 5–6 distributed autonomous seafloor acoustic recorders, operating continuously during each season's research period, were used to detect, locate, and track vocalizing whales. A 16-element towed array tracked individual vocal whales in real-time concurrently with visual observations, allowing biopsy samples of known vocal animals. Active acoustics was used to collect data on the density and distribution of krill so as to place measured variation in whale numbers, distribution, and behavior within an ecological context. The primary whale activity was feeding. Whale feeding patterns and survey distribution followed prey distribution. Vocal whale distribution followed diel feeding patterns and prey distribution. All biopsied vocal animals were males. Numbers of whales estimated by vessel survey, photo-ID, and passive acoustics were correlated. Results suggest that under certain conditions, vocal activity is a reliable measure of distribution and relative abundance. [Work supported by ONR.]

8:35

3aABa4. Locating and enumerating endangered humpback whales in the eastern Caribbean with directional (DIFAR) sonobuoys. Steven Swartz, Phillip Clapham, Tim Cole, Jay Barlow (U.S. Dept. of Commerce, NOAA, Natl. Marine Fisheries Service, 75 Virginia Beach Dr., Miami, FL 33149, steven.swartz@noaa.gov), Mark McDonald (Whale Acoust., Laramie, WY), Erin Oleson, and John Hildebrand (Scripps Inst. of Oceanogr., San Diego, CA)

Visual surveys for whales are limited by visibility conditions at sea and whale diving behavior. Male humpback whales (*Megaptera novaeangliae*) sing complex “songs” during their winter breeding season, and low-frequency song components lend themselves to acoustic detection. A visual and acoustic survey was conducted during the peak breeding season to evaluate passive acoustic methods to assess endangered humpback whales in the Eastern Caribbean, a region where they were exploited to depletion. Directional (DIFAR) sonobuoys were used to detect singing whales. Bearing angles from the sonobuoys to singing whales were calculated in real-time, and locations determined by crossing two or more bearings. A total of 4331 km of “on-effort” visual survey resulted in 9 whale sightings compared to 74 acoustic detections. Acoustic detections from 350 h of monitoring resulted in an abundance estimate of 116 (95% CI: 72-293) whales in February, and 123 (95% CI: 77-313) whales in March. The paucity of visual sightings is attributed to the prevailing high winds and the brief periods that whales spend at the surface. These results demonstrate the advantage of acoustic survey methods over visual methods for detecting whales in areas with poor visibility conditions such as the Eastern Caribbean.

8:50

3aABa5. Acoustic and visual monitoring for marine mammals at Cortez and Tanner Banks. Erin M. Oleson, John A. Hildebrand (Univ. of California San Diego, Scripps Inst. of Oceanogr., La Jolla, CA 92093), Mark A. McDonald (Whale Acoust., Laramie, WY), and John Calambokidis (Cascadia Res. Collective, Olympia, WA)

We have implemented continuous acoustic monitoring for baleen whales in the region surrounding Cortez and Tanner Banks, offshore of Southern California, complemented by bimonthly ship-based visual and acoustic observations. Our objective is to develop methods for acoustic monitoring of baleen whales using seafloor acoustic recording packages, the results of which will be used to produce abundance estimates. The joint acoustic and visual effort will answer questions regarding the probability for acoustic detections of individual animals, the quantitative association of calls with a particular species, and how well the number of animals in an area can be assessed acoustically. These findings can then be used to generate a detectability curve for each baleen whale species, from which the seafloor recorder data can be translated into absolute abundance estimates. The use of continuous acoustic monitoring with seafloor recording packages offers the potential for efficient and economical monitoring of marine mammals. The continuous acoustic census approach will be discussed, and compared to opportunistic visual and acoustic survey methods.

9:05

3aABa6. Simple methods for locating, counting, and tracking sperm whales underwater in three dimensions. Duncan E. McGehee (BAE Systems, 4669 Murphy Canyon Rd., Ste. 102, San Diego, CA 92123, duncan.mcgehee@baesystems.com) and John A. Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA 92037-0205)

Sperm whales’ clicks can be used to locate source animals in three dimensions using volumetric arrays of four or more sonobuoys. Clicks are detected using algorithms that depend on gender. Males are easier to track: their clicks are loud and distinctive, and there is usually only one male nearby. A simple envelope detector suffices. Females click more quietly, at a higher rate, and they often occur in large groups, resulting in a continuous barrage of clicks. Using a single click as a matched filter detector

enhances the detection of clicks coming from a particular animal. Arrival time differences between the different sonobuoys are used to locate the source in three dimensions. A particular animal can be tracked by determining its location over time. An estimate of the number of phonating animals can be made by locating all the source animals during a particular time interval. During NOAA’s SWAPS97 sperm whale survey, arrays of four sonobuoys each were placed around groups of diving sperm whales. Clicks from the animals were recorded at 48ksamples/s. Analyses of these data will be presented. [Work sponsored by NOAA and ONR.]

9:20

3aABa7. Three-dimensional localization of diving sperm whales using a short-aperture towed horizontal array. Aaron M. Thode (MIT, 77 Massachusetts Ave., Cambridge, MA 02139), David K. Mellinger (Oregon State Univ., Newport, OR 97365), Sarah Stienessen (North Seattle, WA 98103), Anthony Martinez (Natl. Marine Fisheries Service, Key Biscayne, FL 33149), and Keith D. Mullin (Natl. Marine Fisheries Service, Pascagoula, MS 39568)

Three-dimensional sperm whale localizations are typically obtained via recording their “clicks” on widely distributed hydrophones. Here, an alternative 3D localization method is investigated, using data collected by an 8-m aperture, five-element horizontal towed array deployed from the NOAA ship GORDON GUNTER in the Gulf of Mexico, as part of a recent sperm whale pilot study. During the night of 3 July 2000, the GUNTER towed the array at a steady 1-kn speed and 60-m depth through a pod of sperm whales, in 1000-m-deep water. During that time, surface and bottom reflections from a single sperm whale click were often recorded. By measuring the bearings and relative arrival times of the direct arrival and reflections, the whale location, array depth, and array tilt could be computed. The latter results were checked against measurements obtained from a time-depth recorder attached to the array. Assuming that the speed of the ship remained fixed and the prevailing currents did not change, subsequent positions could then be obtained using only a surface reflection. This method may provide a convenient way for observing dive profiles of animals within a few kilometers of a survey ship. [Work supported by Minerals Management Service, National Marine Fisheries Service, and ONR.]

9:35

3aABa8. Comparison of acoustic and visual surveying of humpback whales off East Australia. Michael J. Noad (Dept. of Veterinary Anatomy and Pathol., Univ. of Sydney, NSW 2006, Australia) and Douglas H. Cato (Defence Sci. and Technol. Organisation, Pyrmont, NSW 2009, Australia)

Many species of whales produce intense sounds that are audible to substantial distances and thus may be useful in censusing, especially in conditions where visual methods have limited effectiveness. Acoustic methods of surveying, however, have their own limitations and are relatively untried compared with visual surveying. This paper describes the results of an experiment to test the effectiveness of acoustic surveying, by applying it to the East Australian humpback whale population, which has been well-surveyed visually. At the experimental site in southeast Queensland, most of this stock passes within visual range of the coast during migrations. Whales were tracked visually using a theodolite on a hill near shore and acoustically using three buoy systems which received the sounds and transmitted the signals to shore. Almost all whales within 10 km could be tracked visually, thus providing a benchmark for the acoustic surveying. Although only a proportion of the stock is vocalizing at any time, the result tests how representative this is of the total stock. The results show that acoustic surveying can be useful for censusing.

9:50

3aABa9. Planning for a pilot census of marine life in the Gulf of Maine: The role of acoustics. Kenneth G. Foote (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, kfoote@whoi.edu)

Preparations are being made for a pilot census of marine life in the Gulf of Maine ecosystem. The role of acoustics as a rapid, remote sensing tool is elaborated. Potential target organisms for acoustic surveying range from mesozooplankton and macrozooplankton to fish and cetaceans. A number of methodological problems must be addressed. These are illustrated for the echo integration method as applied to a stock of Atlantic herring (*Clupea harengus*) [J. Acoust. Soc. Am. **105**, 995 (1999)]. Particular problems of determining target strength and compensating for possible behavioral effects are also general to the method. [Work supported by the Alfred P. Sloan Foundation.]

10:05

3aABa10. Localizing marine animals and how marine animals might localize sound. Gerald L. D'Spain and Paul A. Lepper (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA 92093-0704)

We can locate vocalizing marine animals, and the animals themselves might locate sound sources, in one of several ways. Time-of-arrival (phase) differences and amplitude differences of a single arrival between spatially separated "ears" are well-known techniques. However, they become ineffective in multipath environments and as the frequency of the sound and/or spatial separation between ears decreases. Another approach is to sense properties of the acoustic field in addition to acoustic pressure. This approach, based on a simple Taylor series expansion of the field and " $f=ma$," apparently is exploited by fish, making them biological equivalents of DIFAR sonobuoys. However, do they also measure acoustic strain rate? An additional method, used by humans, is to take advantage of

time-of-arrival (phase) differences between multipath arrivals. The information on source location contained in the resulting interference patterns can be understood in terms of waveguide invariants. These speculations will be illustrated with numerical simulations and actual ocean acoustic data. [Work supported by ONR.]

10:20

3aABa11. Acoustic identification of female Steller sea lions. Gregory S. Campbell (Cetacean Behavior Lab., San Diego State Univ., San Diego, CA 92182-4611), Robert Gisner (Code 342, Office of Naval Res., Arlington, VA 22217), and David A. Helweg (Code D351, SPAWARSCEN San Diego, San Diego, CA 92152)

Steller sea lions (*Eumetopias jubatus*) breed and rear their young in coastal rookeries dispersed along the northern tier of the North Pacific. Population densities are high and no allomaternal behavior occurs, providing strong selection pressure for mother-pup recognition processes. Mothers and pups establish and maintain contact with individually distinctive vocalizations. Our objective is to understand the acoustic features that serve to identify individual females, and develop a ruggedized computer system to perform acoustic recognition of females in the field. We have cataloged almost 2000 contact calls from 46 females in 1998 and 25 in 1999. Each female is visually identified by marking patterns, which provides the ground truth for acoustic identification. Acoustic properties of the calls were measured and presented to several statistical classifiers. Representations of the calls had to be robust with respect to acoustical variability introduced by motivational changes as the mother and pup regained proximity. The calls, classifiers, and results of generalization tests will be described, and a concept for the field system will be discussed. [Research supported by ONR Research Opportunities for Program Officers award N00014-00-1-0114 to R. H. DeFran, Cetacean Behavior Laboratory, SDSU.]

WEDNESDAY MORNING, 6 DECEMBER 2000

SCHOONER/SLOOP ROOMS, 10:50 TO 11:55 A.M.

Session 3aABb

Animal Bioacoustics: General Topics in Bioacoustics

Lawrence F. Wolski, Chair

Hubbs-Sea World Research Institute, 2595 Ingraham Street, San Diego, California 92109

Chair's Introduction—10:50

Contributed Papers

10:55

3aABb1. Infrasonic and low-frequency vocalizations from Siberian and Bengal tigers. Elizabeth von Muggenthaler (Fauna Communications Res. Inst.)

Tigers have many vocalizations including chuffing, growling, prusten, gurgling, grunting, and roaring. It has been well documented that the tiger's high-amplitude, low-frequency roars, which are thought to be territorial in nature [C. Packer and A. E. Pusey, *Sci. Am.* **276**, 52–59 (1997)] transmit for miles. It has been suggested that because some tigers inhabit dense jungles with limited visibility, the capacity to hear low frequency may be beneficial for sensing and locating prey [G. T. Huang, J. J. Rosowski, and W. T. Peake, *J. Comp. Physiol. A* (2000)]. In an effort to understand more about these low-frequency vocalizations and to provide data to other researchers testing hearing in anesthetized felids, 22 tigers, both Siberian and Bengal, are being recorded. A portable system can record from 3 Hz to 22 kHz. On-site real-time analysis of vocalizations is performed using a portable computer. Real-time and edited playback of sonic and infrasonic tiger vocalizations is facilitated by car audio speakers

capable of producing frequencies from 10 Hz–22 kHz. Initial findings have documented fundamental frequencies of some roars at 17.50 Hz. Other vocalizations, including chuffing, have fundamental frequencies of 35 Hz \pm 5. Playback of both real-time and edited vocalizations appear to illicit behavioral responses, such as roaring, from male tigers.

11:10

3aABb2. On the sound of snapping shrimp: The collapse of a cavitation bubble. Michel Versluis, Anna von der Heydt,^{a)} Detlef Lohse (Dept. of Appl. Phys. and J. M. Burgers Res. Ctr. for Fluid Dynam., Univ. of Twente, P.O. Box 217, 7500 AE Enschede, The Netherlands), and Barbara Schmitz (TU Munchen, 85747 Garching, Germany)

Snapping shrimp produce a snapping sound by an extremely rapid closure of their snapper claw. They usually occur in large numbers providing a permanent crackling background noise, thereby severely limiting the use of underwater acoustics for active and passive sonar, both in scientific and naval applications. Source levels reported for *Alpheus hetero-*

chaelis is as high as 220 dB (peak-to-peak) *re* 1 μ Pa at 1 m distance. Recent ultra-high-speed imaging of the snapper claw closure [Versluis *et al.*, Science (in press)] revealed that the sound is generated by the collapse of a cavitation bubble formed in a fast flowing jet of water forced out from between the claws during claw closure. In this work, we develop a theoretical model for a bubble under such conditions. The dynamics of the bubble radius and the emitted sound can be described by the Rayleigh–Plesset equation. The calculated results are compared with the experimental data. The model fully reproduces the bubble dynamics and it quantitatively accounts for the time dependence of the bubble radius and for the emitted sound.^{a)} Also at Dept. of Physics, Philipps-Univ. Marburg, Renthof 6, 35032 Marburg, Germany.

11:25

3aABb3. A unique way of sound production in the snapping shrimp (*Alpheus heterochaelis*). Barbara Schmitz (Dept. of Zoology, TU Muenchen, Lichtenbergstr. 4, 85747 Garching, Germany), Michel Versluis, Anna von der Heydt, and Detlef Lohse (Appl. Phys., Univ. of Twente, P.O. Box 217, 7500 AE Enschede, The Netherlands)

Sound production is known in more than 50, mostly stridulating, crustacean genera. These acoustic signals occur in agonistic interactions as well as for mate attraction. The mechanism of sound production in snapping shrimp, which also serves to stun or even kill small prey, is especially interesting. The current assumption was that the sound is produced by cocking and then rapidly closing the enlarged modified snapper claw. Snapping shrimp sounds contribute most to coastal biological noise, may be heard up to 1 mile away, and resemble the crackling of dry twigs in fire or the sizzle of frying fat. Recent hydrophone measurements close to tethered shrimp (*Alpheus heterochaelis*) revealed pulse-like signals of 500-ns duration, comprising frequencies beyond 200 kHz, and showing enormous sound pressure levels of up to 220-dB *re* 1 μ Pa (peak to

peak) at 1-m distance. Such high intensities are very unlikely to be produced by the mechanical contact of two claw surfaces. Ultra-high-speed video recordings and simultaneous hydrophone measurements reveal that claw closure results in a water jet, the high velocity of which (25 m/s) leads to the formation of a cavitation bubble, which emits the extremely loud sound upon its collapse.

11:40

3aABb4. Lack of species-specific vocal recognition in Amazonian manatees: *Trichechus inunguis*. Renata S. Sousa Lima and Vera M. F. da Silva (Laboratório de Mamíferos Aquáticos, INPA, C.P. 478, Manaus, AM, Brasil 69083-000, pboi@inpa.gov.br)

Playback experiments were conducted in order to test for the existence of species-specific vocal recognition in Amazonian manatees. The animals were isolated in pools while acoustic stimuli were played from a tape recorder and transmitted underwater through a loudspeaker. Nine animals were monitored for response to playback vocalizations from eighteen different individuals, nine from each species (*Trichechus inunguis* and *T. manatus manatus*). No significant differences were detected in the response of manatees exposed to the different stimuli. Only the time spent close to the speaker was greater when the animals were exposed to conspecific vocalizations ($\rho=0.375$). This result suggests that Amazonian manatees cannot recognize differences between their own and another manatee species' vocalization. The methodology was also tested and no difference in the response of animals exposed to silence or to tape hiss (blank tape used as control) was found. Testing the response to the presence or absence of vocalizations, significant differences were found in time elapsed between the playback and the response ($\rho<0.001$) and in the time spent next to the speaker ($\rho=0.032$), confirming their great ability to perceive sounds underwater. [Work supported by FBPN, MacArthur Foundation, CI, FINEP, MCT/PPG7 and CNPq.]

WEDNESDAY MORNING, 6 DECEMBER 2000

CALIFORNIA SALON 3, 7:20 A.M. TO 12:05 P.M.

Session 3aAO

Acoustical Oceanography: Special Topics

Thomas K. Berger, Cochair

Scripps Institute of Oceanography, University of California, San Diego, 9500 Gilman Drive, La Jolla, California 92093-0238

Christopher D. Jones, Cochair

Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, Washington 98105

Chair's Introduction—7:20

Contributed Papers

7:25

3aAO1. The F factor: Ocean climatology and internal-wave acoustic effects. Stanley M. Flatté and Kimberly J. Noble (Phys. Dept. and Inst. of Marine Sci., Univ. of California at Santa Cruz, Santa Cruz, CA 95064)

The strength of acoustic fluctuations due to internal waves is affected by the temperature-salinity relation of an oceanic region. We define an acoustic fluctuation strength parameter F as the ratio of the fractional potential-sound-speed change to the fractional potential-density change. F is calculated at three depth levels (275, 550, and 850 m), on a one-degree grid of latitude and longitude, using NODC/OCL's World Ocean Atlas 1994. Representative values of F in upper waters range between 5 and 35. Results for intermediate depths range from 5 to 60. In general, F exhibits higher values in the Atlantic Basin than in the Indian or Pacific, and has a maximum at 550 m. The main use of F will be the prediction of travel-

time fluctuations in acoustic propagation experiments, which will be proportional to the value of F , given a universal strength of internal waves.

7:40

3aAO2. Imaging acoustic fluctuations in shallow water using dislocation theory. D. P. Williams, G. L. D'Spain, W. S. Hodgkiss, and W. A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA 92093-0238)

The distribution of the amplitude of a moderate-frequency sound field in a shallow ocean is considered in relation to the existence of dislocations in the phase front [Nye and Berry (1974)] where the amplitude is close to zero. Phase front dislocations strictly occur when the amplitude is zero; in actual ocean acoustic measurements, only low-field amplitudes at a minima can be distinguished due to the interference of background noise.

The dislocations are sensitive to temporal variability in the ocean environment [Kuz'kin, Ogurtsov, and Petnikov (1998)] and their space-frequency fluctuations lend qualitative insight into environmental characteristics. A model is developed to predict the patterns of the sound-field minima resulting, in particular, from internal wave-induced fluctuations in the environment. The model is then applied to the interpretation of the results of 2–7-kHz broadband transmissions collected during a noisy shallow-water acoustics experiment off the coast of San Diego. [Work supported by ONR.]

7:55

3aAO3. Energetics of dislocations in ocean acoustic fields. Gerald L. D'Spain, Duncan P. Williams, and William A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA 92093-0704)

Dislocations are places in the acoustic field where the amplitude goes to zero and the total field phase becomes undetermined [Nye and Berry (1974)]. The positions of these dislocations are sensitive indicators of changes in the ocean environment. Expressions for the acoustic potential and kinetic energy densities and active and reactive vector acoustic intensities are derived for a single-tone sound field. These expressions are then applied to infrasonic ocean acoustic data collected by the Marine Physical Lab's set of freely drifting Swallow floats to help identify dislocations in the underwater acoustic pressure and the component particle velocity fields. Likely dislocation positions were identified during cw source tows at 7, 10, and 16 Hz during the 1990 NATIVE 1 experiment near the Blake Plateau in the northwest Atlantic Ocean. The neighborhoods of dislocations, as well as near other amplitude minima, are regions where a significant acoustic energy flow occurs. The relationship between dislocations and polarization of the acoustic particle motion also is discussed. [Work supported by ONR.]

8:10

3aAO4. Fourier component fluctuations of wideband ocean tomography signals: Empirical temporal statistics from AST96. Rex K. Andrew, Bruce M. Howe, and James A. Mercer (Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105, randrew@apl.washington.edu)

Current theories for wave propagation in random media can predict fluctuations for narrow-band (single-frequency) signals, but do not explicitly address wideband signals. Recent ocean acoustic tomography experiments, however, have employed wideband "m-sequence" signals. Processing here assumes a periodic signal, and hence, involves a unique set of Fourier series components. A technique is presented for decomposing such wideband signals into their Fourier components and then analyzing the space-time fluctuations of these "narrow-band" components. The technique is applied to AST96 data transmitted over 150-km and 1-Mm paths for single-point and two-point (temporal separation) statistics. [Work supported by ONR.]

8:25

3aAO5. Hilbert spectrum of internal waves. Tao Wang (California Inst. of Technol., Pasadena, CA 91125) and Tong Su (Marine Systems Eng. Res. Inst., China State Shipbuilding Corp., Beijing 100036, PROC)

The newly developed Hilbert spectrum offers a powerful tool for nonlinear, nonstationary data analysis, and has found wide applications to Stokes waves, random ocean wind waves, and turbulence [N. E. Huang, Z. Shen, and S. R. Long, *Annu. Rev. Fluid Mech.* **31**, 417–457 (1999)]. This paper presents the Hilbert spectrum of internal waves in the Yellow Sea and East China Sea. According to their intrinsic characteristic scales, the internal wave data are first decomposed into a number of modes of different time scales, designated as intrinsic mode function (IMF) components. The Hilbert transform is then applied to the IMF components to construct the frequency-time distribution of energy density, the Hilbert spectrum. It is shown that the Hilbert spectral analysis provides a new physical insight in the underlying dynamic processes of these shallow-water internal wave phenomena.

8:40

3aAO6. Matched field inversion with a moving source in a shallow-water environment. John Viechnicki and Ross Chapman (Ocean Acoust. Group, School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada, jayvee@ducks.seos.uvic.ca)

Geoacoustic inversion based on matched field processing (MFP) is examined for shallow-water, low-frequency environments. Specific interest lies with resolving geoacoustic parameters from cw tones projected from a moving source. Data obtained using vertical line arrays (VLA) are available from both the 1996 Haro Strait PRIMER Experiment (HSX) and the Santa Barbara Channel Experiment (SBCX) of 1998. The environmental complexity associated with these experiments, namely, strong range-dependent bathymetry and current flow, is typical to littoral environments in general and must be appropriately addressed. Parabolic equation modeling is used as it provides range-dependent results. The VLA receiver configuration is described as a catenary which is typical of bottom-anchored arrays drifting in uniform current flow. Both SBCX and HSX are useful for benchmarking geoacoustic inversion techniques since results from other techniques are available. Estimation of bottom properties is discussed as a function of propagation range, ship track with respect to receiver position, and general bathymetric features. Results for both tangential and radial tracks are presented. [Work supported by ONR.]

8:55

3aAO7. The effect of internal wave time dependence on ray chaos. Michael Vera and Stanley M. Flatté (Phys. Dept. and Inst. of Marine Sci., Univ. of California at Santa Cruz, Santa Cruz, CA 95064)

A range-dependent field of sound speed in the ocean can give rise to chaotic instabilities in acoustic ray paths. A model of internal waves is used as a range-dependent effect on the speed of sound in numerical simulations. These simulations, performed over a 1000-km range, demonstrate the sensitivity of ray paths to slight changes in the sound-speed field or the initial conditions of their launch. Previous work [Simmen *et al.* (1999)] has shown the effect on arrival depth of slightly shifting the launch angle of each ray. The focus of this work is the effect of internal wave time dependence. Comparisons are made between simulations using a time-dependent medium and those using a time-independent ("frozen") medium, and between simulations with different geophysical starting times. The resulting arrival depth differences are compared to those induced by launch angle shifts. The structure and magnitude of these differences are considered in the case of two different sound-speed profiles: a Munk canonical profile and a profile from the Slice89 experiment.

9:10

3aAO8. Characterization of the internal waves and its effect on signal propagation in the Adventure Bank area. T. C. Yang, Kwang Yoo (Naval Res. Lab., Washington, DC 20375), and Martin Siderius (SACLANTCEN, La Spazia, Italy)

During the Advent-99 experiment, which took place in the Sicily Strait in May, 1999, multitone (in the band 200–1500 Hz) narrow-band signals were transmitted from a bottom-mounted tower and received on a 64 phone line array (VLA) covering the water column; the water depth was 80 m. Data at a source-receiver range of 10 km will be presented and modeled using an internal wave model. A CTD chain containing 48 CTDs on a vertical string was towed continuously during the acoustic experiments between the source and VLA, yielding a time-evolving sound speed profile between the source and receiver. From that the average buoyancy frequency and sound speed profile were determined as a function of range. Internal-wave-mode depth functions were calculated using the measured buoyancy profiles. Internal-waves-mode amplitudes were determined from the eigenvalues of the sound speed covariance matrix for the lowest 5 modes. Internal-wave-frequency spectrum has the typical 2-power dependence as in the Garrett-Munk model. The modeled results of mean transmission loss and amplitude fluctuations with and without internal waves will be presented and compared with data. [Work supported by the Office of Naval Research and SACLANTCEN.]

3aAO9. Sound in a time-dependent ocean: Implications on acoustic tomography of currents. Oleg A. Godin (NOAA/Environ. Technol. Lab., Boulder, CO 80305)

Ocean is a nonstationary acoustic environment. Surface and internal waves, tides, and mesoscale eddies contribute to sound-speed and ocean surface variability on different temporal scales. Acoustic effects of the nonstationarity include violation of the reciprocity principle and signal frequency variation along the propagation path. Mathematical models of underwater sound propagation often either ignore ocean nonstationarity or account for it within the frozen-medium approximation. In this paper, accuracy of different approaches to model sound propagation in time-dependent ocean is analyzed within the frameworks of the ray theory and the adiabatic mode approximation. The theoretical approach is based on the space-time geometrical acoustics and a space-time version of the "vertical modes–horizontal rays" technique. While the frozen-medium approximation proves to be generally rather crude and inadequate in modeling nonreciprocity due to sound-speed time dependence, another simple technique, the quasistationary approximation, is shown to be a sufficiently accurate and efficient approach to modeling low-frequency underwater sound propagation. Contributions to ray travel time and mode-phase nonreciprocity due to medium motion and time dependence are compared for several typical current tomography scenarios. The feasibility of distinguishing between acoustic nonreciprocities due to currents and due to medium nonstationarity is discussed. [Work supported by NRC.]

9:40–9:50 Break

9:50

3aAO10. Frequency dependent transmission loss related to modal properties in layered nonlinear internal wave environments. Scott Frank, William L. Siegmund (Dept. of Math. Sci., Rensselaer Polytechnic Inst., Troy, NY 12180), Mohsen Badiye, Yongke Mu (Univ. of Delaware, Newark, DE 19716), and James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Broadband data were taken during the SWARM 95 experiment from airgun and linear frequency modulated sources while nonlinear internal waves were crossing two different propagation tracks. In a previous study, the geotime behavior of the depth-averaged received intensity was examined, and amplitude modulations were associated with the passage of internal waves. Additionally, a time-frequency analysis of the data exhibited periodic changes in the modal structure of the signals. Though some discrepancies occurred, both of these effects were observed in parabolic equation simulations of the SWARM environment. To address the discrepancies in received intensity levels, various physical mechanisms will be investigated, for example geoacoustic attenuation and ocean thermocline variability. Effects on the time-frequency representations of signals passing through internal wave packets will be documented and categorized. To understand features of frequency dependent transmission loss, relationships between the eigenvalues and the variable layer structure of the waveguide will be developed and applied to both measurements and simulations. Such relationships will aid in extracting the amplitude, width, and azimuthal propagation angle of nonlinear internal waves from broadband acoustic data. [Work supported by ONR.]

10:05

3aAO11. One-path scintillation method for transverse current monitoring. Iosif Fuks, Mikhail Charnotskii, and Konstantin Naugolnykh (Univ. of Colorado, CIRES/NOAA, Environ. Technol. Lab., 325 Broadway, Boulder, CO 80305)

A sound wave passing through the inhomogeneous fluid is modulated producing the intensity fluctuations or scintillations at the receiving plane. These fluctuations evolve as the intervening medium changes. If the fluctuations in the medium are produced by the advection of frozen fine-structure field, then the evolution of the scintillation pattern can be linked to the motion of the ocean medium [S. Clifford and D. Farmer, *J. Acoust.*

Soc. Am. **74**, 1826–1832 (1983)]. This method of the ocean current registration can be realized by the measurement of correlation of the signals transmitted from the source to two separated in space receivers and can be respectively considered as space-domain scintillation method. However, it is possible to register the transverse flow by sounding ocean along the one path using the multifrequency signal [J. Fuks, *Radiotechn. Electron.* (in Russian) **3**, 515–524 (1975)]. The coherence of frequency-spaced signal fluctuation is related to the advection of the inhomogeneous medium through the sound path, thus providing a basis for the current velocity measurement. The sensitivity of this method in application to flows in straits and narrow channels is considered and the accuracy of the current velocity measurement is estimated.

10:20

3aAO12. Beamforming through an anisotropic, time-varying internal wave field. Steven Finette and Roger M. Oba (Naval Res. Lab., Washington, DC 20375-5350)

Shallow-water environments commonly exhibit complex spatial and temporal variations in temperature and salinity caused by internal wave activity. These quantities map into a sound-speed distribution that can significantly distort both the amplitude and phase of an acoustic field propagating through the environment. This presentation describes computer simulation results on the effect of an anisotropic, time-varying internal wave field on plane wave beamforming of a point source within a shallow-water waveguide. The (azimuthal) anisotropy is described by a solitary wave packet propagating through the region. A 3D parabolic equation code was used to propagate a 400-Hz signal through this space and time-evolving environment. Beamformed power, parametrized by receiver depth and an azimuthal angle, is illustrated for horizontal arrays and presented in terms of beam space, time, and range from the source. When the packet enters the region between source and receiving array, significant periodic beam wander and beam splitting are seen; such effects are shown to be related to horizontal refraction of energy from individual internal wave depressions forming the solitary wave packet. [Work supported by the Office of Naval Research.]

10:35

3aAO13. Acoustic detection and classification of hurricanes. Joshua D. Wilson and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

Hurricanes are some of the most destructive natural disasters known to man. Currently, it is difficult to detect, track, and classify hurricanes in areas of the world that lack satellite coverage and specialized aircraft. The purpose of this research is to investigate the feasibility of using underwater hydrophone arrays to detect and classify a tropical cyclone. A wave theoretic model is developed for the spatial covariance of the ambient noise field generated by a hurricane. Simulations are performed to determine whether ambient noise due to a hurricane is detectable over the more typical wind-wave noise accumulated from the rest of the ocean. Scenarios where a hydrophone array is placed directly beneath the storm and at a remote distance from the storm are investigated.

10:50

3aAO14. Acoustic observation of hydrothermal flows. Christopher D. Jones, Darrell R. Jackson (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105), Peter A. Rona, and Karen G. Bemis (Inst. of Marine and Coastal Sci., Rutgers Univ., New Brunswick, NJ 08901-8521)

Acoustic backscatter has been used to observe hydrothermal flows at several sites on the Juan de Fuca Ridge at frequencies of 200 and 330 kHz. In the case of plume discharges ("smokers"), backscatter from particulates is used to obtain three-dimensional images and to measure flow via Doppler shift. For diffuse flows, acoustic scintillation of the signal backscattered from the seafloor is detected using ping-to-ping correlation. Mapping of diffuse flows using both fixed and moving platforms will be

discussed. The magnitude and time lag dependence of ping-to-ping correlation have been linked to flow parameters using the Rytov scattering approximation. [Work supported by NSF Grant OCE 98-18841.]

11:05

3aAO15. Low-frequency acoustics of bubble plumes formed in fresh water and salt water. Thomas K. Berger and Michael J. Buckingham (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, tberger@ucsd.edu)

Experiments aimed at comparing the acoustics of bubble plumes formed in salt water and fresh water were performed in a laboratory tank and natural bodies of water. The plumes were generated as a vertical stream of water, with a diameter of millimeter order and velocity ranging to about 10 m/s, penetrated a free water surface. The acoustic signals were measured with hydrophones in different positions, amplified, and digitally sampled. The measured power spectra revealed well-defined, nonuniformly spaced resonance peaks at frequencies below about 1 kHz. A theory based on a conical bubble plume, with a nonuniform sound speed profile acting as a resonant cavity, predicts the frequencies of the resonance peaks and shows remarkable agreement with the experimentally measured values for both salt water and fresh water bubble plumes. Though the bubbles contained in the salt water bubble plumes are much smaller than those in the fresh water plumes, the resonance frequencies are largely unaffected, consistent with the observation that the plumes in both cases are nearly identical in size and geometry for the same incoming velocity and air-entrainment rate. The acoustic levels in salt water were lower, indicating possible differences in attenuation or driving mechanisms. [Work supported by ONR.]

11:20

3aAO16. Acoustic propagation measurements in the surf. John S. Stroud, Kerry W. Commander (Coastal Systems Station, Code R21, 6703 W. Hwy. 98, Panama City, FL 32407-7001, stroudjs@ncsc.navy.mil), Robert J. McDonald, and Jo Ellen Wilbur (Systems Station, Panama City, FL 32407-7001)

As a follow-on experiment to the 1999 Near Shore Acoustic Network Experiment, the 2000 Surf Zone Acoustic Test Experiment (SZATE) was conducted off of the Scripps Institute of Oceanography pier in La Jolla, CA. In this experiment, measurements were conducted in a very shallow water/surf zone (VSW/SZ) region to investigate acoustic propagation and sound channel stability in the coastal area. These measurements utilized two receivers and a broad band source. The data to be presented were acquired by a computer controlled system designed and operated by Grant Deane of Scripps Institution of Oceanography. This system provided for 12 18-min data acquisition periods per day. The probative transmissions consisted of a variety of linear frequency modulated (LFM) and binary phase shift keyed (BPSK), as well as 4-ary frequency shift keyed (FSK) signals. Each of the types of signals was used during each data acquisition period, providing a record of propagation throughout the tidal cycle. Results of this acoustic measurement for select pulse types will be reported. [Work supported by ONR Code 321OE.]

11:35

3aAO17. Monostatic and bistatic scattering by a single bubble near a pressure release interface: Laboratory measurements and modeling. George Kapodistrias and Peter H. Dahl (Appl. Phys. Lab. and Dept. of Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, georgek@apl.washington.edu)

Scattering by a single bubble near a flat pressure release interface is investigated theoretically and experimentally. A ray-acoustic interpretation is used to describe the four scattering paths, from source to bubble to receiver. Multiple scattering effects are accounted for using a closed-form solution derived from the multiple scattering series. A bubble with radius $a \approx 425 \mu\text{m}$ is placed on a fine nylon thread, which is practically transparent to sound, at a distance d_B from the interface. The primary variable is d_B and it ranges from $1a$ to $100a$. Experiments are performed at a frequency of 120 kHz with the transducers arranged in both bistatic and monostatic configuration. Theory and experiment are in excellent agreement, verifying the dominant effect of the four paths in the response of the bubble, with multiple scattering playing a role for $kd_B < 1$, where k is the wave number of the medium. In the long-range limit our simulations agree with those of Ye and Feuillade [J. Acoust. Soc. Am. **102**, 798–805 (1997)] including the shifting of the bubble's resonant frequency. The dependence of scattering on transducer arrangement, range to bubble, grazing angle, and phase relation among the four paths, *vis-à-vis* monostatic and bistatic scattering, are discussed.

11:50

3aAO18. Space-time sound fluctuations due to internal solitons in shallow water. Boris G. Katsnelson, Serguey A. Pereselkov (Voronezh Univ., 1 Universitetskaya Sq., Voronezh 394693, Russia), Valery G. Petnikov (General Phys. Inst., Moscow 117333, Russia), Konstantin D. Sabinin, and Andrey N. Serebryany (N. N. Andreev Acoust. Inst., Moscow 117333, Russia)

In this paper results of numerical modeling of sound propagation in shallow-water waveguide with internal waves are presented on the basis of experimental data about internal solitons (IS) registered in the Japan sea [A. N. Serebryany, Sov. Phys. Oceanol. **27**, 225–226 (1987)]. These IS constitute some sequences of oscillations with amplitude up to 10 m moving toward coastal line with the velocity about 0.25 m/s. The wavelength of separate oscillations in this group is about 325 m; total length of the packet is about 4 km. Three-dimensional sound-speed distribution is constructed using the mentioned data. On the basis of this model, space-time fluctuations of intensity of the sound propagating in such medium are calculated. The modeling is carried out within the framework of the parabolic equation method in the horizontal plane and the modal approach in the vertical one. These fluctuations can be interpreted as the forming of a sound waveguide in the horizontal plane; space-time scales of fluctuations correlate with space-time scales of IS. Different numerical examples are presented; possible experimental setup is discussed. [Work supported by RFBR, Grant 00-05-64752.]

3a WED. AM

Session 3aBB

Biomedical Ultrasound/Bioresponse to Vibration: Therapeutic and Diagnostic Ultrasound

Mark R. Prausnitz, Chair

School of Chemical Engineering, Georgia Institute of Technology, Atlanta, Georgia 30332-0100

Chair's Introduction—7:30

Contributed Papers

7:35

3aBB1. Real-time observation of inception and growth of HIFU-induced tissue lesions. Cyril Lafon, Michael R. Bailey, Lisa N. Couret, Peter J. Kaczkowski, Andrew A. Brayman, Lawrence A. Crum (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105), and Oleg A. Sapozhnikov (Moscow State Univ., Moscow, Russia)

To study the biological effects of high-intensity focused ultrasound (HIFU), experiments are usually performed on isolated or perfused tissues. Indeed, the complex phenomena occurring in tissue during HIFU-induced coagulation necrosis is difficult to mimic with synthetic phantoms. A good phantom should first match the acoustical and thermal properties of tissues. Furthermore, heating above a thermal threshold should induce a permanent, localized and observable change corresponding to protein denaturing in tissue. Lastly, the choice of a transparent material makes possible real-time examination of the development of coagulation necroses. We have used bovine eye lenses in this aim. The density, sound speed, attenuation, and thermal threshold for irreversible damage to the bovine lens were measured and found to be similar to those for liver or muscle, common tissues for HIFU experiments, although acoustic attenuation is slightly higher in the lens. Transparency of the lens allowed us to observe HIFU-induced lesion evolution in real time. The shape and size of the lesions obtained in the lens agreed well with results obtained in liver. In conclusion, the transparent bovine eye lens is a useful model for visualization of thermal lesions. [Work supported by ONR.]

7:50

3aBB2. Theoretical predictions of ultrasonic fields, temperature response, and lesion dynamics in biological tissue for the purpose of noninvasive disease treatment. Francesco P. Curra, Pierre D. Mourad, Steven G. Kargl, Lawrence A. Crum (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105), and Vera A. Khokhlova (Moscow State Univ., Moscow, Russia)

Ultrasound has been used for decades as a means for noninvasive treatment of diseases. Low-intensity ultrasound is routinely applied in physical therapy for muscular and neurological related illnesses. In contrast, high-intensity focused ultrasound (HIFU) is used to induce coagulative necrosis of tissue for cancer treatment or hemostasis. Our efforts concern the latter. Predictions of ultrasound fields, temperature response, and lesion dynamics are obtained by a model which accounts for nonlinear sound propagation in inhomogeneous media, an arbitrary frequency power law for acoustic attenuation, and temperature time history [J. Acoust. Soc. Am. **107**, No. 5, Pt. 2 (2000)]. The model is expanded from its previous version to include attenuation and sound speed dependence on temperature levels and also to consider generation of gas bubbles within the tissue. Results are presented in terms of treatment strategies that provide maximum energy transfer for coagulating the targeted tissue while minimizing damage to the surrounding area.

8:05

3aBB3. Two-dimensional ultrasound phased array for treatment of benign prostatic hyperplasia. Janelle L. Helser, Victor W. Sparrow, and Nadine Smith (Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804, nbs@engr.psu.edu)

In the prostate, focused ultrasound offers an attractive means of non-invasive tissue ablation for treatment of Benign Prostatic Hyperplasia (BPH). Intracavitary two-dimensional, ultrasound arrays allow for deep localized heating and are capable of generating sufficient power for tissue ablation. A two-dimensional array design is capable of focusing electronically in three dimensions, by controlling the power and phase, instead of having to physically adjust or reposition the transducer. The goal of this project was to design a 1.5 MHz, two-dimensional, linear, PZT transducer that could steer a focus throughout the region of the prostate. Pressure fields for the array were theoretically determined, using amplitude shading and varying elemental sizes to reduce grating lobe and side lobe levels. The amplitude shading was altered each time the transducer was refocused so that the peak amplitude was located in the element normal to the focus. Based on this 100-element theoretical array, a 1.5 cm × 8 cm transducer array was constructed to evaluate the feasibility of delivering thermal therapies from a two-dimensional array.

8:20

3aBB4. High-intensity focused ultrasound hyperthermia in nonuniform tissue phantoms. Jinlan Huang, R. Glynn Holt, and Ronald A. Roy (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215, jinlan@bu.edu)

It has been demonstrated that high-intensity focused ultrasound (HIFU) shows promising potential for stopping bleeding, both from individual blood vessels as well as from gross damage to the capillary bed. Thermal effects have been suggested to play a major role in occlusion of small vessels and also appear to contribute to the hemostasis of major blood vessels. The goal of the present work is to understand the heating process in nonuniform flow-through tissue phantoms. The experiments have been focused on a single-vessel flow-through phantom, and the studies include the prediction and measurement of temperature fields as a function of acoustic pressure, insonation time and flow rate, and investigating different insonation geometries including on-axis and off-axis insonations. The experimental data are compared to simulation results using the Westervelt equation for calculating the nonlinear acoustic field and two coupled bioheat equations for the tissue domain and the blood domain for the temperature field calculation. [Work supported by DARPA.]

8:35

3aBB5. Platelet activity as a result of exposure to high-intensity focused ultrasound. Sandra L. Poliachik, Ryan J. Ollos, Pierre D. Mourad, Lawrence A. Crum (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, poliachi@u.washington.edu), and Wayne L. Chandler (Univ. of Washington, Seattle, WA 98195)

Using platelet-rich plasma, we investigated the capability of 1.1-MHz cw high-intensity focused ultrasound (HIFU) to produce "acoustic primary hemostasis," including platelet activation, aggregation, and adhesion to a collagen-coated surface. Platelet activity was evaluated for exposure

durations of 100–500 s at intensities of 0–2250 W/cm². In order to avoid heating effects, temperatures in platelet trials were maintained below 42 °C through use of a tank cooling system and control of exposure parameters. Flow cytometry, laser aggregometry, conventional microscopy, environmental scanning electron microscopy, and passive cavitation detection were used to quantify platelet activation, aggregation, adhesion, and associated cavitation. HIFU can activate platelets and cause them to adhere to a collagen-coated surface. Cavitation was monitored during aggregation trials and was quantified to provide a relative measure of the amount of cavitation that occurred in each aggregation trial. Regression analysis shows weak correlation between aggregation and intensity, and a strong correlation between aggregation and cavitation occurrence. [Research supported by DARPA.]

8:50

3aBB6. Influence of chemical composition of membrane-disrupting polymers on relative cavitation activity and hemolysis. Tyrone M. Porter, Josh Nickerson, Lawrence A. Crum (Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105, tporter@apl.washington.edu), Fiona E. Black, Niren Murthy, Patrick S. Stayton, and Allan S. Hoffman (Univ. of Washington, Seattle, WA 98195)

In previous studies, we have shown that membrane-disrupting polymers may also act as cavitation promoters. These polymers are designed to efficiently disrupt red blood cells in a pH-dependent fashion (Murthy *et al.*). When combined with high-intensity focused ultrasound (HIFU), there is a noted increase in relative cavitation activity and a corresponding increase in red blood cell lysis. Varying the chemical composition of the polymer (length of hydrocarbon chains, molecular weight, etc.) modifies the hemolytic and cavitation-promoting activity of the polymer. For example, the hemolytic and cavitation promoting activity of poly(ethyl acrylic acid) (PEAAc) rises as the pH of the solvent decreases from 7.4 to 6.1. However, the polymer poly(propyl acrylic acid) (PPAAc), which has a more hydrophobic pendant alkyl group, promotes cavitation and, therefore, hemolysis at a pH of 5.0, 6.1, and 7.4. Variations in the polymer molecular weight change the number of hydrophobic regions which also alters the hemolytic and cavitation activity. From these results, polymers may be designed which, when combined with ultrasound, optimize drug transport across cell membranes in a pH-dependent or -independent manner.

9:05

3aBB7. Behavior of ultrasound contrast agents near the fragmentation threshold. Wen-Shiang Chen, Thomas J. Matula, and Lawrence A. Crum (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, wschen@u.washington.edu)

Understanding the destruction process of ultrasound contrast agents is important in therapeutic ultrasound applications (such as ultrasound-enhanced drug delivery), as well as in certain imaging applications (such as “flash echo” imaging of the myocardium). In the destruction of Optison™ microbubbles, our observations suggest that there are two pressure thresholds: a lower threshold which leads to shell rupture and the production of daughter (derivative) bubbles, and a higher threshold leading to the onset of the inertial cavitation (IC) activity. Slightly above the shell-disruption (SD) threshold, the acoustic scattering signal decreases, presumably due in part to the partial dissolution of the derivative bubbles. The extent of the quiescent region, the IC pressure threshold, and the strength of the subsequent cavitation activity are all highly dependent on the acoustic pulse parameters. It is found that for pressure amplitudes in excess of the IC threshold, a longer pulse length or a higher pressure level will decrease the duration of this quiescent period. By controlling the bubbles’ response through fine-tuning the acoustic parameters near the SD and IC threshold, one could design an ultrasound system which would optimize the desirable effects of contrast agents for imaging and therapy applications. [Work supported in part by ONR and WTC.]

9:20

3aBB8. Analysis of deformation process for cell model including a gas bubble by shock waves. Masaaki Tamagawa, Ichiroh Yamanoi, and Atsushi Matsumoto (Grad. School of Energy Sci., Kyoto Univ., Kyoto, 606-8501 Japan)

This paper describes the analysis of deformation process for cell model including a gas bubble by shock waves and flow induced by propagating underwater shock waves, in order to study the mechanism and high efficiency of destruction of cells and microcapsules by shock waves and bubbles, such as for drug delivery system (DDS) and for bioprocesses to make energy and for environmental protection. For this analysis, the cells are modeled as liquid droplet including a gas bubble in water. By computation, using the ALE (arbitrary Lagrangian–Eulerian) method, the coupling oscillation is analyzed for particle velocity and pressure wave through the cell. The results show that the effects of the acoustic impedance of the fluid media in water is large for the deformation process and the effects of deformation of droplet boundary is also large for deformation process for a gas bubble inside the cell and for microjet flow by a collapsing bubble.

9:35

3aBB9. Study of the mechanism of fragmentation of a microbubble exposed to ultrasound using a high-speed observation system. Nobuki Kudo, Takehiro Miyaoka, Kaori Kuribayashi, Katsuyuki Yamamoto (Grad. School of Eng., Hokkaido Univ. N13W8, kita-ku, Sapporo, 060-8628 Japan, kudo@bme.eng.hokudai.ac.jp), and Michiya Natori (Natl. Ohkura Hospital, Tokyo, Japan)

In this study, we observed the dynamic behaviors of microbubbles exposed to ultrasound using a high-speed camera (Ultranac, Nac, Japan). Although the maximum frame rate of the high-speed camera used in this study is 20 million frames/s, the possible frame rate was limited to 6 million frames/s because the high optical magnification of the observation system resulted in a shortage of light exposure to the camera. Behaviors of albumin-shelled microbubbles of 10–60 microns in diameter were clearly observed as in 24-frame sequential high-speed photographs at the maximum optical magnification of 85×. Several typical phenomena of bubble behavior, such as deformation and rupture of the bubble shell, anisotropic contraction of a microbubble and generation of a small stream, and fragmentation of a bubble by the small stream, were observed. From those observation results, it was concluded that the small stream generated by anisotropic contraction of a microbubble exposed to ultrasound causes fragmentation of the bubble in the contraction phase. [This research was partially supported by a grant-in-aid for scientific research from the Ministry of Education, Science, Sports and Culture, Japan.]

9:50

3aBB10. An attempt to explain turbulence occurring when yeast cells suspended in water-rich ethanol mixtures are irradiated in a quasistanding ultrasonic plane wave at 2.2 MHz. Stefan Radel, Aiden J. McLoughlin (Dept. of Industrial Microbiology, Univ. College Dublin, Belfield, Dublin 4, Ireland, radel@iap.twinn.ac.at), Martin Groeschl, and Ewald Benes (Inst. of General Phys., Vienna Univ. of Technol., Vienna, Austria)

Ultrasonic resonators are successfully used for the separation of suspended particles or cells, e.g., as cells filters in biotechnology. These systems are based on the radiation forces spatially rearranging the particles in the pressure nodal planes of the quasistanding ultrasonic wave field. When yeast—as this technology is tested to be applied in brewing—is suspended in water this separation/immobilization takes place as predicted. However, suspensions of yeast in water-rich ethanol mixtures show a breakdown of the spatial distribution above certain concentrations of ethanol and the cells are turbulently mixed in the separation system. This has earlier been reported to go along with an impaired viability of yeast in a 12% (v/v) ethanol/water mixture. The scope of this paper is to attempt an explanation based on the acoustic contrast between the cells and the liquid as water-rich ethanol mixtures have unexpected acoustical and physical properties. Data of the spectral behavior of the system at *in situ* levels of true elec-

trical power input are shown as well as the results of examinations, if (sub) harmonic and/or transient cavitation were taking place. [Work supported in part by the European Commission's TMR Programme, Contract No. ERBFMRXCT97-0156, EuroUltraSonoSep.]

10:05–10:15 Break

10:15

3aBB11. Potential errors in contrast-agent-mediated estimates of regional myocardial perfusion arising from myocardial anisotropy.

James G. Miller, David E. Sosnovik, Steven L. Baldwin, and Mark R. Holland (Dept. of Phys., Washington Univ., St. Louis, MO 63130)

Recently developed imaging modalities based on the nonlinear response of contrast agents are dependent upon the local acoustic pressure. Thus, regional variations of acoustic pressure arising from myocardial anisotropy can result in nonuniform "apparent perfusion" in the absence of a perfusion defect. Measurements of regional variation of myocardial attenuation in the septum and lateral walls of excised sheep hearts are reported. Studies were carried out on hearts obtained immediately after slaughter from 8 healthy adult sheep. These measurements of attenuation were carried out in orientations analogous to those encountered in the apical four-chamber view, which is the preferred view for clinical studies of myocardial perfusion using contrast-agent-enhanced echocardiography. The video signal analysis method introduced by the University of Wisconsin group was used to analyze data obtained with a commercially available (ATL) clinical scanner using a linear array (L 7-4) operating in fundamental mode. Tissue mimicking phantoms were imaged under identical conditions. Normalizing sample data by reference data from the phantoms yields quantitative estimates of slope of attenuation. Results of these experimental measurements document significant transmural variations in attenuation that could lead to apparent perfusion anomalies in contrast-agent-mediated estimates of regional myocardial perfusion. [Work supported by NIHHL40302.]

10:30

3aBB12. Optimal transmit beam synthesis for coded-excitation pulse-echo imaging.

Emad Ebbini and Mok-kun Jeong (Dept. of Elec. and Computer Eng., Univ. of Minnesota, 200 Union St., Minneapolis, MN 55455, emad@ece.umn.edu)

The use of code-fed arrays for pulse-echo ultrasonic imaging is gaining increased attention in medical applications. One of the most important applications is true real-time 3-D cardiac imaging where parallel processing of multiple-image lines is required to meet the stringent real-time constraints. The key to this approach is transmitting multiple codes to illuminate the region of interest (ROI), producing independent echo waveforms from different directions. We have examined several different scenarios in which the transmit aperture is divided into focused subapertures. Each subaperture is excited by one code and focused appropriately based on the definition of the ROI. For example, a 128-element array can be driven by eight independent codes on eight 16-element subapertures to illuminate a 6° ROI with f number of 2. The optimal solution is a sparse array pattern for each transmit subaperture (i.e., the subapertures overlap). Significant improvement in both lateral and range sidelobes of the system point-spread function (PSF) were achieved (compared to random transmit code selection). In this paper, we describe a new optimal synthesis of coded excitation transmit patterns. Examples of pulse-echo PSFs for the imaging system along with image reconstructions from a speckle-generating tissue-mimicking phantom will be given.

10:45

3aBB13. Simulation of tissue displacement using ultrasonic speckle tracking.

Tjundewo Lawu and Mitsuhiro Ueda (Dept. of Intl. Development Eng., Tokyo Inst. of Technol., Tokyo, 152-8552 Japan)

The quantitative evaluation of ultrasonic displacement in tissue under externally applied forces is a necessary step in the field of ultrasonic diagnostics. In this study, the speckle tracking method is used to investi-

gate the motion artifact produced by a rotating tissue. An analytic prediction of this motion artifact in relation to system characteristics (i.e., beam width, pulse duration, frequency, and phase of the signal) is presented. The tissue is modeled as a random array of numerous point scatterers, and rf signals are computed based on the convolutions between the transmission pulse and the impulse response of each scanning line. These analytical predictions are then compared to the computer simulation of the model. The results show that the artifact resulting from the speckle tracking method depends on the ultrasonic beam parameters. The method is quite general and can be extended to study the effects of other tissue motion, in particular, the deformation of tissue.

11:00

3aBB14. Statistical characterization of high-frequency ultrasonic scattering from human skin *in vivo*.

Balasundar Raju and Mandayam Srinivasan (MIT, Rm. 36-788, 50 Vassar St., Cambridge, MA 02139, raju@mit.edu)

High-frequency ultrasound backscatter signals from forearm and fingertip skin *in vivo* of 20 normal human volunteers were analyzed in terms of their envelope statistics. Data obtained from several focal zones by axially translating a focused transducer were gated (100 ns) and Fourier transformed to compute magnitude of envelope at the center frequency of 30 MHz. It was found that the envelope histograms and SNR were significantly affected by the presence of occasional peaks that were unlikely to be from the background tissue. Hence a new parameter, the ratio of median to interquartile range (MQR) was used to evaluate its potential for tissue characterization. For all focal zone depths (up to 3 mm for forearm and 1.95 mm for fingertip), the MSR value was less than the Rayleigh limit of 1.30, indicating the presence of non-Rayleigh statistics. Significant differences in MSR were found between tissues close to the surface (0.3 mm) at the fingertip and forearm (0.67 ± 0.16 vs 0.97 ± 0.13 at 30 MHz, mean \pm SD values are among subjects), which could correspond to anatomical differences at these locations, namely the increased thickness of the stratum corneum at the fingertip. These results indicate that non-Rayleigh statistics might aid in discriminating different skin tissues.

11:15

3aBB15. Detection of the malpositioned endotracheal tube via acoustic reflectometry: An *in vitro* study.

David T. Raphael (Dept. of Anesthesiology, LAC-Univ. of Southern California Medical Ctr., Los Angeles, CA 90033 and UMDNJ-Robert Wood Johnson Med. School, NB, NJ)

Time-domain acoustic reflectometry can generate a one-dimensional area-distance profile of a cavity, such as the lung or esophagus. Can acoustic reflectometry be used to detect an endobronchial intubation? In an Institutional Review Board approved study, an endobronchial intubation in the right mainstem bronchus in an anesthetized patient was detected first by acoustic reflectometry, and the malposition confirmed by fiberoptic bronchoscopy. A study was initiated to determine the area-distance profile characteristics of an endobronchial intubation in an *in vitro* branching glass model. A symmetrical glass model (Witeg Scientific, Anaheim, CA), open at the proximal end, with two orders of bifurcation terminating in four closed distal branches, was studied. Area-distance (A-D) profiles were obtained with a customized computer-based acoustic reflectometer (Hood Labs., Pembroke, MA). With an endotracheal tube (ETT) attached to the reflectometer, area-distance profiles were obtained with the distal ETT in the following positions: the mid-"trachea," "carina," "mainstem bronchus," the second bifurcation, and the secondary "bronchus." The area-distance profile features of an endobronchial intubation, as determined by this model study, are similar to those observed clinically in a patient.

11:30

3aBB16. Surface acoustic wave velocity in thin biotissue. Chiaki Miyasaka and Bernhard Tittmann (Penn State Univ., 212 Earth & Eng. Sci. Bldg., University Park, PA 16802)

We report a technique to characterize small portions on the order of a few microns of a biological tissue. The heart muscle was selected for specimen as an example of the soft material. The heart muscle was cut by a microtome and coated on a substrate. The thickness of the specimen was about 3 μm . Fused quartz was used as a substrate because its elastic properties are known and stable. A spherical lens was used to determine the position for the measurement. A cylindrical lens was used to measure velocities of the SAW waves within the specimen. 400 MHz was used for both imaging and measurement. The generation of the SAW waves was simulated by numerical calculations based on the wave propagation theory for layered media. As a result, the variation of the velocities representing the anisotropy of elastic properties was obtained and found to be about 7%.

11:45

3aBB17. Acoustic reflectometry for estimation of lung gas-phase volume during partial liquid ventilation in sheep. David T. Raphael (Dept. of Anesthesiol., LAC-Univ. of Southern California Medical Ctr., Los Angeles, CA 90033)

Acoustic reflectometry was studied as a possible tool to measure the lung gas-phase volume during partial liquid ventilation (PLVSM). With IRB approval, healthy sheep (weight 10–22 kg) ($n = 3$) were anesthetized, intubated, mechanically ventilated in the supine position, and underwent a low-lying tracheostomy. A base-line functional residual capacity (FRC) via a N₂ washout technique was measured. Perflubron (LiquiVent[®], Alliance Pharmaceutical Corp., San Diego, CA) was instilled through the tracheostomy tube in successive aliquots, each consisting of 20% of measured FRC, up to 100% FRC. A Hood Labs acoustic reflectometer was used to obtain lung gas volume determinations for the FRC measurement

and after perflubron instillations. The acoustic reflectometer was used to generate a cross-sectional area versus axial length curve. The subglottic lung gas-phase volume was calculated by integrating area over length. The reduction in the subglottic lung gas-phase volumes for each animal exhibited a strong linear correlation with the amount of perflubron instilled. Linear least-squares regression models run on the individual sheep data were all significant, with the largest p value being 0.0067. Acoustic reflectometry may be of use in the monitoring of lung gas-phase volume in PLV and respiratory disease.

12:00

3aBB18. Doppler ultrasound detection of shear waves remotely induced in tissue phantoms by focused ultrasound. Evgen A. Barannik, Sergii A. Girnyk, Volodymyr V. Tovstiyak (Kharkiv Natl. Univ., Sq. 4 Svobody, Kharkiv, 61077 Ukraine), and Armen P. Sarvazyan (Artann Labs., North Brunswick, NJ 08902)

Remote generation of shear waves in tissues by radiation force of focused ultrasound is the basis of shear wave elasticity imaging (SWEI), a new acoustic method of medical diagnostics. Feasibility of SWEI was previously demonstrated in the experiments with optical and NMR detection of ultrasonically induced shear waves [Sarvazyan *et al.*, *Ultrasound Med. Biol.* **24**, 1419 (1998)]. In the present study the SWEI system with ultrasonic pulsed Doppler detection of shear waves was designed and tested using a range of gelatin-based phantoms and muscle tissue. Radiation force was generated by a focusing transducer of 8 cm diameter and 7 cm focal length. The carrier frequency was 1 MHz with intensity 1.85 W/cm² at the surface of the transducer. The Doppler detection transducer operating at 3.5 MHz was installed in the center of the pumping transducer. The developed procedures of tissue motion detection and Doppler signal-processing algorithms based on the autocorrelation method allowed estimation of the velocity with accuracy better than 0.2 mm/s. The waveforms of the shear wave at various distances from the pumping ultrasonic beam axis were measured and the shear wave velocities were evaluated. The oscillation character of relaxation process for some phantoms and muscle tissue was shown experimentally.

WEDNESDAY MORNING, 6 DECEMBER 2000 CALIFORNIA SALONS 1 AND 2, 8:20 TO 11:40 A.M.

Session 3aEA

Engineering Acoustics: International Comparison of Calibration and Measurements

George S. K. Wong, Chair

Institute for National Measurement Standards, National Research Council, Montreal Road, Ottawa, Ontario K1A 0R6, Canada

Invited Papers

8:20

3aEA1. International calibration comparisons: Who benefits? George S. K. Wong (Inst. for Natl. Measurement Standards, Natl. Res. Council, Ottawa, ON K1A 0R6, Canada)

The mutual acceptance of acoustical calibrations and measurements between industrial countries is essential for international trade and the removal of no-tariff trade barriers. For example, the sound level or sound power emitted by a machine measured at the country of manufacture with certified instruments and methods in accordance with international standards, should be acceptable by the importing country without the requirement to duplicate the measurements. To achieve this mutual recognition, it is necessary for the exporting country to have proven capabilities via international comparisons and an unbroken chain of traceability from their national metrology institute to the machine shop level. Under the umbrella of the Bureau International des Poids et Mesures (BIPM), the Consultative Committee on Acoustics, Ultrasound and Vibration (CCAUV) has arranged international calibration comparisons, involving over 15 countries. The above comparisons require a lot of effort from each participating country. One may ask the question: Who is the beneficiary of international comparisons? The detailed answer is rather complex. In general, the results of International Calibration Comparisons provide confidence in the measurement capabilities of the participants. In the long term, the consumer is the ultimate beneficiary.

8:40

3aEA2. Key elements of mutual recognition arrangements. Klaus Brinkmann (Physikalisch-Technische Bundesanstalt Braunschweig, Bundesallee 100, 38116 Braunschweig, Germany, klaus.brinkmann@ptb.de)

Mutual recognition arrangements in the field of metrology are concluded between equal partners to promote the acceptance of each other's calibration certificates and test reports in order to facilitate global trade ("one-step-testing"). Acoustics is affected by this development both on the level of national metrology institutes, cooperating in organizations of the Metre Convention, and on the level of accreditation bodies and their accredited calibration and testing laboratories, cooperating in the International Laboratory Accreditation Cooperation (ILAC). Both arrangements are equivalent in the sense that they are based on mutual confidence between the partners, established mainly by the results of interlaboratory comparisons, uniform implementation of International Standards and peer assessments or accreditations.

9:00

3aEA3. Reference values for the sensitivity of standard accelerometers used in intercomparisons. David J. Evans, Stefan D. Leigh, and Beverly F. Payne (Natl. Inst. of Standards and Technol., 100 Bureau Dr., Stop 8221, Gaithersburg, MD 20899-8221, dje@nist.gov)

The National Metrology Institutes (NMIs) of five countries in North America and South America participated in an interlaboratory comparison involving the calibration of the magnitude of the sensitivity of three standard accelerometers. This comparison was performed by laboratories within the framework of the Interamerican Metrology System (SIM). One of the key values to be obtained in any interlaboratory comparison is an estimate of the reference values for the artifact being calibrated. Three statistical methods have been used to obtain candidate reference values and associated uncertainties from the SIM intercomparison data: an average of means method; a method based on the *ISO Guide to the Expression of Uncertainty in Measurement*; and a maximum likelihood method. Reference values and associated uncertainties obtained using the three methods are presented and compared, as well as useful graphical displays with resulting qualitative conclusions.

9:20

3aEA4. International key comparison of ultrasonic power measurements. K. Beissner (Physikalisch-Technische Bundesanstalt, 38116 Braunschweig, Germany, Klaus.Beissner@ptb.de)

One of the worldwide key comparisons performed at present under the auspices of the International Committee for Weights and Measures (CIPM) covers the time-averaged ultrasonic power emitted by an ultrasonic transducer into water. The ultrasonic power value is particularly important in characterizing the output of diagnostic and therapeutic medical equipment. Nine national metrology institutes participate in the comparison. The time schedule is from July 1999 to November 2001. The aim is to demonstrate the international consistency of calibration and measurement certificates. An ultrasonic standard transducer is circulated and power measurements are to be made at frequencies of 2, 6 and 10 MHz and at five power levels in the milliwatt and watt ranges. The measurement procedures applied and the main details of the technical protocol will be discussed.

9:40

3aEA5. International comparisons of hydrophone calibrations. Roy C. Preston, Stephen R. Robinson (Ctr. for Mech. and Acoust. Metrology, Natl. Physical Lab., Teddington, Middlesex TW11 0LW, UK, rcp@npl.co.uk), and Arnie L. Van Buren (Underwater Sound Ref. Div., Naval Undersea Warfare Ctr. Div., Newport, RI 02841)

In the global markets which are increasingly dominating industrial trade, international comparison of hydrophone calibrations is the primary mechanism of ensuring harmonization of underwater acoustics measurement standards across national borders. This paper describes the results of several recent comparison exercises initiated in Europe and the USA. These include both informal comparisons of hydrophone calibrations of a bilateral nature and formal comparisons involving a number of countries organized under the auspices of regional or international metrology organizations. Completed projects which will be described include a Euromet/European Commission project involving a total of 12 participants from seven European countries and a comparison between the Russian Federation and China. In these cases, results will be given and the important conclusions summarized. Finally, a relatively new comparison project will be described which has been set up under the auspices of the Consultative Committee for Acoustics, Ultrasound and Vibration (CCAUV) of the Bureau International des Poids et Mesures (BIPM) in Paris. This key comparison has now been agreed to for the kilohertz frequency range 1–500 kHz and will involve the circulation of three hydrophones. Countries participating in the comparison include the UK, China, Germany, Russian Federation, and USA, with Canada expected to join soon.

10:00–10:15 Break

10:15

3aEA6. Importance of the free-field calibration of microphones. Victor Nedzelnitsky (Natl. Inst. of Standards and Technol. [NIST], Sound Bldg. [233], Rm. A147, 100 Bureau Dr., Stop 8221, Gaithersburg, MD 20899-8221, Victor.Nedzelnitsky@nist.gov)

Numerous regulatory requirements, standards, and product characterization and quality control procedures important for industry, commerce, health, and safety rely on practical measuring instruments including sound level meters, personal sound exposure meters (noise dosimeters), and standardized measuring microphones. Such instruments are usually large enough that the effects of diffraction must be considered throughout a significant portion of the operating frequency range. These effects cause differences between the free-field, the pressure, and the diffuse-field sensitivities of instruments, particularly at high frequencies. Differences can also result from other phenomena, such as the effects of a microphone's static (barometric) pressure equalization vent, especially at low frequencies. Different calibration procedures have evolved, so that an appropriate calibration is available for each of the field types:

free, pressure, and diffuse. Critically important free-field procedures and some of the most accurate and widely used diffuse-field procedures should be, but are not always, traceable to primary free-field calibrations of microphones by the reciprocity method. Future international key comparisons at the highest level of accuracy should include determinations of the free-field sensitivities of laboratory standard microphones such as IEC type LS2 by the reciprocity method.

10:35

3aEA7. Reference sound source calibration at various temperatures and site altitudes. Angelo J. Campanella (Campanella Assoc. & Acculab, 3201 Ridgewood Dr., Columbus, OH 43026)

Adjustments are made for atmospheric temperature and barometric pressure when calibrating a sound power reference sound source (RSS) and when applying an RSS in the substitution method to evaluate an unknown sound power. Experience with Hemi-anechoic calibration of a dozen RSS units outdoors at various temperatures near sea level and a special test series on one RSS unit at three higher sites up to 9927 feet and applying a pistonphone reference to verify microphone and sound-level meter stability are described. According to these varied temperature results and the theoretical work of others [Allan Powell, *J. Acoust. Soc. Am.* **32**, 982–990 (1960), Eqs. (41)–(42), after Lighthill], it appears that the mechanical fan-type RSS is a multipole—chiefly quadrupole and octupole—type sound source. It was apparent, according to these varied altitude results, that emitted sound pressure and sound power are both linear functions of barometric pressure. Experimental results will be presented. Practical applications to determine actual sound power, to report normalized sound power, and to predict actual sound pressure and sound power expected at a remote site under nonstandard conditions are described.

Contributed Papers

10:55

3aEA8. Model-based method for improvements in the calibration of ultrasonic power measurements at low-power levels. Lixue Wu (Acoust. Standards, Inst. for Natl. Measurement Standards, Natl. Res. Council Canada, Ottawa, ON K1A 0R6, Canada)

The measurement uncertainty in the calibration of ultrasonic power measurements at low-power levels was reduced by a model-based signal-processing method. The ultrasonic power measurement system is based on the radiation force balance method according to IEC 61161. Outputs of an electronic balance and the excitation voltage were simultaneously sampled and recorded. A computer-controlled coaxial switch that utilizes data lines of the parallel port of a desktop computer was used for power-on and power-off control. Power-on and power-off intervals were finely controlled by the desktop computer. The Electroacoustic radiation conductance, defined as the ratio of the ultrasonic power of a transducer and the square of the rf voltage at its input, was calculated from the recorded data. During data processing, a simple model for drifting and switching was proposed and a least-mean-squares algorithm was used to estimate parameters of the model. Good measurement reproducibility was also observed that resulted in the reduction of total measurement uncertainty.

11:10

3aEA9. Boundary-element simulation used in the design of a sound intensity calibrator. Johan Gramtorp, Vicente Cutanda, Anders Eriksen, and Erling Sandermann Olsen (Briel & Kjaer, Skodsborgvej 307, 2850 Naerum, Denmark)

The recent introduction of hand-held intensity measurement equipment has created a growing need for verification of the measurement equipment in the field. The key feature for easy field use is the possibility of calibrating the two-microphone intensity probe without dismounting the spacer. Working models were built, but none of them worked to more than approximately 3 kHz. Compliance with IEC 61043 requires the calibrator to

work up to 7.1 kHz. A lot of modifications on these models were tried, but none of them worked. The development project was close to being stopped. A boundary-element model of the sound intensity calibrator was built. It verified the measured result from the working models. Based on the boundary-element model the best type and position of the sound source and the optimum dimensions of the calibrator cavity were found. Measurements on a final model verified the simulation results. The calibrations could now be made without dismounting the spacer.

11:25

3aEA10. Calibration of electret microphone cartridges for application in a multichannel acoustic array. Allan J. Zuckerwar, William M. Humphreys, and Scott M. Bartram (NASA Langley Res. Ctr., M.S. 493, Hampton, VA 23681)

Electret condenser microphone cartridges, having the advantages of small size, low cost, few lead wires, low output impedance, and acoustic specifications comparable to those of air condenser microphones, are well suited for application in multichannel acoustic arrays. Amplitude and phase calibration of electret cartridges by two methods is described. In the first, a test cartridge, reference microphone, and driver are inserted into a coupler, whereby the calibration takes place by a comparison method. In the second method, the calibration is performed in a commercial acoustic calibrator (B&K type 4226), which was modified by the manufacturer to permit the phase calibration. Further, a special adapter was fabricated to seat the cartridge at the proper position in the calibrator coupler. Amplitude and phase calibrations of 80 electret microphones were performed. The calibrated microphone cartridges were flush-mounted in an 80-channel array to measure dynamic surface pressures. The array was installed in a backward-facing step model and tested in a low-speed wind tunnel. An associated data acquisition system performed conditioning and digitization of the signals from each of the microphones. Preliminary results using the array to characterize the pressure distribution in the separated flow region downstream of the step will be reported.

3a WED. AM

Session 3aMU

Musical Acoustics: Asian Musical Instruments and Traditions I

James P. Cottingham, Chair

Department of Physics, Coe College, Cedar Rapids, Iowa 52402

Chair's Introduction—8:25

Invited Papers

8:30

3aMU1. The acoustics of the Asian free reed mouth organs. James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402, jcotting@coe.edu)

Mouth-blown instruments using a free reed coupled to a pipe resonator have a long history in China, Japan, and throughout Southeast Asia. The sheng, khaen, and bawu have been studied experimentally and theoretically as typical representatives of this family of instruments. Acoustical measurements made include studies of reed vibration and impedance measurements of the pipes. Particular attention has been paid to the coupling of the reed vibration with the pipe resonator. The sheng employs a free reed at one end of a closed tube with a conical-cylindrical cross section. The khaen employs an open tube of effective length L , with the reed located at approximately $L/4$. The bawu is a closed cylindrical pipe with the free reed at one end, in which the effective acoustical length is varied by the use of tone holes. The playing frequency of each pipe of the sheng or the khaen is typically slightly above both the resonant frequency of the pipe and the natural frequency of the reed. In the bawu, on the other hand, both the pipe resonance and sounding frequency are normally well above the natural reed frequency, resulting in a striking difference in tone quality.

9:00

3aMU2. Some acoustics of the shakuhachi—and of the shakuhachi player's face. Joe Wolfe and John Smith (School of Phys., Univ. of New South Wales, Sydney NSW 2052, Australia, J.Wolfe@unsw.edu.au)

The shakuhachi is an end-blown Japanese flute. Like other flutes, it is open at the place of excitation, so it operates at minima of the acoustic impedance. The jet interacts with both the bore impedance and the radiation impedance, which is baffled by the player's face. Large changes in the relative geometry of the instrument and the face contribute to the flexibility in pitch and timbre that are important elements of the traditional playing style, and which gives the instrument much of the expressiveness for which it is renowned. We report measurements of the impedance spectra $Z(f)$ of the bore, and of the baffled radiation load imposed on the jet. $Z(f)$ of the instrument differs from that of the Western flute family because the shakuhachi has neither the narrow chimney of the Western flute, nor the short resonator upstream of the jet. These influence the overall form of $Z(f)$ for the instrument. Other features of $Z(f)$ are explained by the tone hole position and geometry. The $Z(f)$ of the radiation field varies for different playing positions, which has important effects on the tuning of the minima and on the spectral response.

9:30

3aMU3. The Klais organ in Kyoto: A comparison of the acoustics of the Japanese organ stops to the traditional Japanese instruments. Jonas Braasch (Institut für Kommunikationsakustik, Ruhr-Universität, Bochum, Germany) and Christian Ahrens (Ruhr-Universität, Bochum, Germany)

The Klais organ in Kyoto is unique among organs of the world in having a section with Japanese organ stops. Although common organ stops are sometimes named *shakuhachi* in Japan, the Klais organ is the only one that has, besides a French and a German section, a Japanese section containing four specially designed stops imitating the Japanese instruments: *shakuhachi*, *shinobue*, *sho*, and *hichiriki*. While the *shakuhachi* and *shinobue* stops use flue pipes, free-reed pipes are used for the *sho* and the *hichiriki*. Sound recordings of single tones were made of these Japanese stops and analyzed regarding the sound spectra and the attack transients, as well as the variation of the fundamental frequency during the attack transient. For comparison, sound recordings of traditional instruments were made at the Geijutsu Deigaku University of Tokyo. The results show that the sound characteristics of the traditional instruments could be imitated quite well. The free-reed pipes work well not only to imitate the free-reed instrument *sho*, but also to imitate the double-reed *hichiriki*. This organ is a good example in which the long-neglected free-reed stops are more suitable than the commonly used beating reed stops. [Work supported by DFG.]

10:00

3aMU4. Chinese string instruments 400 B.C.—200 A.D. Bo Lawergren (Hunter College of the City Univ. of New York, 695 Park Ave., New York, NY 10021, bo.lawergren@hunter.cuny.edu)

When the tomb of Marquis Yi of Zeng (deceased 422 B.C.) was opened in 1978, a large number of musical instruments came to light. Many of those have since been well studied—in particular the bronze and stone chimes—but the string instruments (various types of zithers) have not. This year the author published the first Western language study of the zithers from Zeng and contemporary Chinese sites (B. Lawergren, "Strings," in *Music in the Age of Confucius*, edited by J. F. So, Washington, DC), and it forms the basis

of this survey. Some types of Zeng zithers were precursors of instruments (*qin* and *zheng*) that became important during the last two millennia of Chinese music. Other types (*se* and the *zhu*) died out after some centuries. The most interesting precursor is that of the classical *qin*. The latter has been held in highest esteem since at least 200 A.D., imbued with deep philosophical meaning, and considered an indigenous invention *par excellence*. But several features of the precursor appear to be foreign, probably from central or western Asia. A reproduction will be used to demonstrate sounds, tuning systems, and concepts of the Zeng *qin*.

10:30

3aMU5. The Burmese harp: Its scale and tunings. Robert M. Williamson^{a)} (Emeritus, Dept. of Phys., Oakland Univ., Rochester, MI 48309)

The 16 strings of the Burmese horizontal, arched harp are tuned in pentatonic octaves over a range of two-and-a-half octaves. A brief description of the harp, its history, and examples of its classical music will precede a discussion of the results of pitch measurements of harp tones recorded in Mandalay in the mid-20th century of the late Muriel C. Williamson [*The Burmese Harp* (Southeast Asia Press, DeKalb, IL, 2000)]. Prior to about 1960, the seven-tone Burmese harp scale could be accurately described by the notes CDE*(E)FGA*(A)B*, where the starred notes were about a quarter tone (50 cents) lower than the corresponding Western notes. The pitches of the notes E* and A* could be varied when they occurred as stopped tones of the harp. The distinctive Burmese notes became Western within a few decades. The remarkably precise tunings of master harpists sometimes showed octaves stretched by about 15 cents. Previous measurements of Burmese fixed-pitch instruments show them to have had somewhat tempered version of the harp scale. ^{a)}Present address: 1189 Pond Cypress Drive, Virginia Beach, VA 23455.

11:00

3aMU6. How are two tones produced simultaneously in throat-singing? Seiji Adachi (ATR Info. Sci. Dept., 2-2 Hikaridai Seika, Kyoto, 619-0288 Japan) and Masashi Yamada (Osaka Univ. of Arts, Higashiyama Kanan, Osaka, 585-8555 Japan)

The most remarkable characteristic of throat-singing is that one singer simultaneously produces two distinct tones—drone and melody. The drone has a low and sustained pitch, and is sung with a pressed voice. The melody has a whistle-like tone, whose pitches are much higher than the drone. It was conjectured that the production of the melody was caused by the resonance of the vocal tract, but has never been proved. This study clarifies that the high melody pitch is produced by the pipe resonance of the rear cavity in the vocal tract. This is derived from acoustic investigations on a throat-singer's vocal tract measured by magnetic resonance imaging. Four different shapes of the vocal tract are examined, with which the melody pitches of F6, G6, A6, and C7 are sung along with the F3 drone of a specific pressed voice. The second formant frequency calculated from each tract shape is close to the melody pitch within an error of 36 cents. Sounds are synthesized by convolving a glottal source waveform with transfer functions calculated from the vocal tract shapes. Two pitches are found to be successfully perceived when the synthesized sounds are listened to.

WEDNESDAY MORNING, 6 DECEMBER 2000

CALIFORNIA SALON 4, 7:55 TO 11:45 A.M.

Session 3aPA

Physical Acoustics: Thermoacoustics I

Gregory W. Swift, Chair

Los Alamos National Laboratory, K764, Los Alamos, New Mexico 87545

Chair's Introduction—7:55

Contributed Papers

8:00

3aPA1. Frequency response of a nonlinear acoustical resonator of arbitrary shape. Mark F. Hamilton, Yurii A. Ilinskii, and Evgenia A. Zabolotskaya (Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712-1063, zhenia@mail.utexas.edu)

The frequency response of a nonlinear acoustical resonator is investigated analytically and numerically. First, natural frequencies of a resonator whose shape is close to cylindrical are studied analytically by perturbation. An asymptotic formula is derived for the natural frequencies as a function of resonator shape. The solution at first order in the perturbation shows that each natural frequency can be shifted independently via appropriate spatial modulation of the resonator wall. Numerical calculations for resonators of different shapes establish the limits of the asymptotic formula. Second, the nonlinear interactions of modes in a nearly cylindrical resonator are investigated with Hamiltonian formalism as a basis for studying wave interaction in a resonator of arbitrary shape. A nonlinear frequency response function for the fundamental mode is derived by as-

suming that nonlinear interaction occurs only with the second harmonic. Bent tuning curves are predicted which lean up or down depending on whether the second-harmonic frequency is above or below the corresponding natural frequency, respectively. A fully nonlinear 1-D numerical code is used to verify the analytical result for the frequency response. [Work supported by ONR.]

8:15

3aPA2. Two-dimensional instability study of a thermoacoustic prime-mover. Ivan Delbende (Univ. of Paris 6, LIMSI, BP 133, 91403 Orsay Cedex, France)

A numerical model aimed at investigating the development of instability in a thermoacoustic prime-mover is presented. It is based on a discretization of the two-dimensional compressible Navier–Stokes equations, and does not rest on the plane pressure wave assumption. The computational domain is made of a single stack interplate space extended up to both resonator ends. The working gas is assumed to be initially at rest, and

a longitudinal temperature gradient is enforced. The linear instability properties of the system are then given by the eigenmodes of the discretized evolution operator linearized in the vicinity of the above basic state. Temporal growth rate and oscillation frequency are thus obtained for all thermoacoustic modes, and the prime-mover instability onset is determined as a function of the various parameters (mainly temperature gradient magnitude and mean pressure). As far as *linear* instability is concerned, this formulation is attractive, since it avoids any direct numerical simulation and any problem involved by the presence of two very different time scales.

8:30

3aPA3. Experimental instability study in a thermoacoustic prime-mover. Emmanuel Bretagne, Ivan Delbende, and Maurice-Xavier François (Univ. of Paris 6, LIMSI, BP 133, 91403 Orsay Cedex, France)

An experimental linear stability study of a thermoacoustic prime-mover is performed for different values of the mean pressure between 0.5 and 10 bars. The damping rate is carefully obtained as a function of the temperature gradient $|\nabla T|$ enforced along the stack, up to the instability onset at $|\nabla T|_c$. These results are then confronted with the predictions of a numerical model based on Rott's theory used with complex frequencies, in which the prime-mover is seen as a feedback loop in the electrical analogy. The experimental results are found to reasonably comply with Rott's theory as soon as the mean pressure exceeds 2 bars. Below this value substantial discrepancies are found, as confirmed by the work of Yazaki. Above the instability onset, the saturated wave amplitude is measured as a function of $|\nabla T| - |\nabla T|_c$ for fixed pressure. The bifurcation to the nonlinear saturated wave has been tentatively determined as subcritical, although thermal inertia effects make it look supercritical.

8:45

3aPA4. Fluctuation before onset of oscillations in a thermoacoustic device. Young Sang Kwon, Orest G. Symko, and Karin Durrant (Phys. Dept., Univ. of Utah, 115 S. 1400 E., Rm. 201, Salt Lake City, UT 84112-0830)

The onset of oscillations in a resonant device where a temperature gradient produces a sound is a fundamentally interesting problem as it consists of a transition where random gas motion, when biased, produces an almost pure sound. This was studied in small 1/4-wave resonant tubes containing a fibrous stack across which a temperature gradient was maintained by means of heat exchangers at each end. The temperature gradient biases the random motion of the gas to trigger a standing wave in the resonator. The acoustic spectrum was measured as the temperature gradient approached the onset for oscillations. Factors affecting the transition to oscillation consist of stack gain, quality factor of the resonator, magnitude of the temperature gradient, and acoustic load. The directed diffusion of the gas along the stack leads to a series of sharp pressure pulses whose stochastic behavior triggers resonant oscillations in the tube. Onset temperature differences less than 50 °C have been observed and this could further be reduced by a suitable choice of parameters affecting the onset. This is an example of a thermal ratchet engine biased by a temperature gradient.

9:00–9:15 Break

9:15

3aPA5. A graphical software application for design and simulation in thermoacoustic research (DSTAR). Thomas J. Hofler (Dept. of Phys., Naval Postgrad. School, Monterey, CA 93943, tjhofler@nps.navy.mil), Eric Purdy, and Scott Curtis (U.S. Navy)

A graphical application called DSTAR has been written for the 32-bit Windows environment, for the purpose of thermoacoustic heat engine analysis. Written in C++ and MFC, the application uses pull-down menus and tabbed dialog boxes, and has additional graphical features such as automatic plotting of thermoacoustic state variables and geometry out-

put in CAD file format. DSTAR supports a variety of variable units including dimensionless units, which are extremely convenient in the design phase of an engine. Current versions of the software are freely available at <http://cooler.physics.nps.navy.mil/hofler/>.

9:30

3aPA6. Development of a miniature thermoacoustic refrigerator. Thomas J. Hofler and Jay A. Adeff (Dept. of Phys., Naval Postgrad. School, Code PH/HF, Monterey, CA 93943, tjhofler@nps.navy.mil)

A miniature thermoacoustic refrigerator is being developed for the purpose of cooling integrated circuits below their failure temperature in hot environments. Work has been done on a piezoelectric acoustic driver operating at 4 kHz. A simple refrigerator has been built and tested that uses one atmosphere of air as a working medium, is less than 3 in long, and has produced 12 °C of cooling. A more advanced pressurized refrigerator is under development. [Work supported by DARPA and Rockwell Science Center.]

9:45

3aPA7. Redesign of a small thermoacoustic refrigerator. Tamra S. Underwood, Bryan S. Morley, Benjamin L. Kuritz, and Ralph T. Muehleisen (Dept. of Civil, Environ., and Architectural Eng., Univ. of Colorado, Boulder, CO 80309)

At the Fall 1999 meeting of the Acoustical Society of America, the design and initial performance of a small, electrically powered thermoacoustic refrigerator was presented. The refrigerator has nominal dimensions of 5 in. long and 1 in. diameter with a target of 10 W of cooling over a 25 °C temperature span using 2-bar helium as the working gas. Because of limitations in the acoustic transducer, performance was limited. A new design with a more powerful acoustic transducer is presented. The performance of the actual device will be compared to the predicted values. [Work supported by the Office of Naval Research.]

10:00

3aPA8. A large solar/heat-driven thermoacoustic cooler. Reh-lin Chen and Steven L. Garrett (Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804, rxc132@psu.edu)

Based on the success of an earlier solar-powered thermoacoustics prime mover which used a direct-illumination stack and no hot-side heat exchanger [Chen and Garrett, Proc. 16th Int. Cong. Acoust., Vol. II, 813–814 (1998)], a large solar/heat-driven thermoacoustic cooler was designed and fabricated. Target cooling powers of 10 to 60 W, over a 25-deg temperature span, were based on a thermal input power of 150 to 600 W. To concentrate the required amount of solar power on an 11-cm-diameter ceramic stack, a 10-ft diameter fiberglass parabolic dish, used for satellite TV, has been converted by gluing aluminized Mylar™ on its surface over a 2-m diameter. A two-axis coordinated solar tracking system, driven by two computer-controlled motors, has produced the required 600 W of solar power to illuminate the hot side of the stack for a maximum of 3 h. Measured performance of the solar refrigerator will be compared to DELTA-E models. [Work supported by the Office of Naval Research.]

10:15

3aPA9. Performance improvements for the solar powered, thermoacoustically driven, thermoacoustic refrigerator. Jay A. Adeff and Thomas J. Hofler (Naval Postgrad. School, Phys. Dept. Code PH, Monterey, CA 93943)

The world's first completely solar powered, thermoacoustically driven, thermoacoustic refrigerator has been improved with increased cooling power and higher efficiency. A larger 24-in.-diam Fresnel lens provides increased heat power to focus sunlight at 550 °C onto the hot end of a 1-in. reticulated vitreous carbon prime mover stack. The high-intensity

sound waves produced by the prime mover are used to power a thermoacoustic refrigerator. The cooling power has been substantially improved and the temperature span has been increased from 18 to 30 °C with the cold end at 0 °C.

10:30–10:45 Break

10:45

3aPA10. A two-microphone thermoacoustic impedance tube. Timothy Simmons, Richard Raspet, and Robert Hiller (Dept. of Phys. and Astron., and Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677, tgsimmon@olemiss.edu)

An experimental arrangement has been developed to explore the thermoacoustic properties of stack materials. The apparatus may be viewed either as a thermoacoustic engine with two microphones mounted in the tube wall, or as a two-microphone impedance tube that accommodates thermoacoustic elements. A sample stack material is placed between two heat exchangers and specific acoustic impedance measurements are taken as a function of frequency and at low-drive amplitude (to avoid nonlinear effects). The design is such that nearly all physical elements and operational parameters can be varied fairly easily. The “tube” is composed of three (flanged) pipe sections of different length, which independently contain a loudspeaker, stack and heat exchangers, and a termination. The test is performed with and without a temperature gradient imposed across the stack. The tube may be pressurized—allowing for different gases held at various ambient pressures—while the relative positioning of the elements within the tube may be varied. Hence a very wide range of testing conditions may be achieved. [Work sponsored by ONR.]

11:00

3aPA11. Thermoviscous functions of wire mesh and RVC stacks. Ralph T. Muehleisen and C. Walter Beamer (Dept. of Civil, Environ., and Architectural Eng., Univ. of Colorado, Boulder, CO 80309)

Thermoacoustic stacks made of stacked wire-mesh elements or reticulated vitreous carbon (RVC) are becoming popular because of their performance, cost, and ease of construction. The thermoviscous Rott functions f_v and f_k [related to the porous media function $F(\lambda)$] have been measured for wire mesh and RVC stacks. The thermoviscous functions were determined from measurements of the characteristic impedance and propagating wave number of the stacks. The measurements were then used to develop empirical models for the stacks. The measurement technique, measurement results, and empirical models are presented. [Work supported by the Office of Naval Research.]

11:15

3aPA12. High-amplitude viscous effects in single pores. Andi Petculescu and Larry Wilen (Dept. of Phys. and Astron., Ohio Univ., Clipping 251, Athens, OH 45701, apetcule@helios.phy.ohiou.edu)

Recently, we reported on a new technique to measure viscous effects of single pores [A. Petculescu and L. Wilen, *J. Acoust. Soc. Am.* **107**, 2819 (2000)]. The method involves a simple lumped-element analysis of a compliant region in parallel with the pore. The pore impedance is determined by subtracting off the measured compliance. With two lengths of the same pore and a second subtraction, one can determine the impedance of the uniform middle part of the pore as well as the impedance of the pore ends. We have extended our measurements to higher amplitudes to look at nonlinearities in the pore. We will discuss how we can determine that the nonlinear effects are principally due to the pore ends, and how we measure these effects. Specifically, results will be presented for the amplitude dependence of the impedance as well for as the generation of higher harmonics. Different end geometries will be considered (such as rounded versus sharp tube ends) and the implication for jet pumps will be addressed. [Work supported by ONR.]

11:30

3aPA13. Thermoacoustic oscillation around a plate in a nonlinear standing wave field. Huang Dongtao, Guo Qing, and Zhu Zhichi (Dept. of Eng. Mech., Tsinghua Univ., Beijing 100084, PROC, dongtaohuang@263.net)

A stack of plates is a most basic and important component of thermoacoustic engines and thermoacoustic refrigerators. Analysis of thermoacoustic oscillation around a plate is a basis of analysis of the mechanism of these machines, but only the works in a linear standing wave field were published. On the other hand, it is shown from some experiments that the thermoacoustic efficiencies of these machines will be increased in the nonlinear standing wave field, but any quantitative result has not been found yet. In this paper a numerical simulation of thermoacoustic oscillation around a plate in a standing wave tube where the sound-pressure level of the sources is from 90 to 170 dB has been presented in order to discuss the nonlinear effects quantitatively. Two conclusions have been obtained: (1) thermoacoustic efficiencies are increased with the increase of the source slowly below 150 dB, moderately from 150 to 160 dB, and rapidly above 160 dB. (2) There exists a rule of multiplying growth of high-order harmonics of oscillation temperature level, which is very similar to the rule of multiplying growth of high-order harmonics of oscillation sound-pressure level in a nonlinear standing wave tube.

Session 3aSA

Structural Acoustics and Vibration: Vibrations of Structures

Courtney B. Burroughs, Chair

Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16802

Contributed Papers

9:00

3aSA1. A unified method for free vibrations of elastically restrained plates arbitrarily loaded with springs and masses. W. L. Li and M. Daniels (United Technologies Carrier Corp., A+R Bldg., Carrier Pkwy., Syracuse, NY 13221)

A simple and unified Fourier series method is presented for the vibration analysis of a class of plates that are simply supported along a pair of opposite edges and elastically restrained along the others. The plates can also be arbitrarily loaded with springs and masses. The transverse displacements of the plates are expressed as the superposition of a Fourier sine series and a simple cubic polynomial. The polynomial is introduced to take care of all the discontinuities with the original displacement and its relevant derivatives along the edges so that the Fourier series now simply represents a residual (or conditioned) displacement which is continuous and has at least three continuous derivatives. The excellent accuracy and convergence of the current method are demonstrated through various numerical examples.

9:15

3aSA2. Weak and strong couplings between a master oscillator and a set of satellite oscillators. G. Maidanik (Carderock Div., Naval Surface Warfare Ctr., Code 7030, 9500 MacArthur Blvd., West Bethesda, MD 20817-5700)

The distribution of resonance frequencies of the satellite oscillators *in situ* is imposed. This imposition demands that the stiffness control term in the satellite oscillator's impedance and the stiffness control term in the coupling, of this oscillator to the master oscillator, be supplemental. Moreover, the distribution is aligned in ascending order; namely, $\omega r \leq \omega q$; $q = (r+1)$; $1 \leq r \leq (R-1)$, where (ωr) and (ωq) are the resonance frequencies of the (r)th and (q)th satellite oscillators, respectively, and (R) is their number. In addition, the numbers of resonance frequencies of the satellite oscillators, on either side of the resonance frequency (ωo) of the master oscillator, are rendered equal. The couplings are assigned mass, stiffness, and gyroscopic control terms. Examples of the influence that the satellite oscillators collectively have on the response of the master oscillator, under various coupling forms and strengths, will be cited.

9:30

3aSA3. On the use of Ritz series as an alternative to field equations for modal analysis of continuous systems. Jerry H. Ginsberg (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

Mathematical analysis of a continuous system's transient response commonly relies on modal properties obtained as an eigensolution of the field equations. Such solutions become cumbersome if the system has attached springs, masses, or substructures, in which case the eigenvalue problem requires mathematically connecting subdomains of the continuum with continuity conditions and Robin boundary conditions. Derivation of the modal orthogonality conditions, knowledge of which is required for implementing the expansion theorem, is quite intricate in such cases. The Rayleigh-Ritz method is a widely used alternative for identifying modes. Representing the displacement field with a Ritz series in conjunction with Hamilton's principle, which is commonly known as the method of as-

sumed modes, offers a generalization of the Rayleigh-Ritz method that can also be employed to analyze forced response. The present paper will demonstrate that if one appropriately alters their viewpoint, and performs a few minor intermediate steps, it is possible to use the Ritz series formalism to obtain the same results (mode functions, orthogonality properties, and the expansion theorem) as those obtained by solving field equations. Such a derivation solely requires knowledge of the kinetic and potential energy functionals, and therefore is much easier to implement for multiply connected systems.

9:45

3aSA4. Natural frequencies of nonaxisymmetric vibration of prolate spheroidal shells. Sabih I. Hayek (Dept. of Eng. Sci. and Mech., Penn State Univ., University Park, PA 16802) and Jeffrey E. Boisvert (Naval Undersea Warfare Ctr. Div. Newport, Newport, RI 02841)

The equations of motion for nonaxisymmetric vibration of prolate spheroidal shells of constant thickness were derived using Hamilton's principle. The thin shell theory used in this derivation includes three displacements and two changes of curvature. The effects of membrane, bending, shear deformations, and rotatory inertias are included in this theory. The resulting five partial differential equations are self-adjoint and positive definite. The nonaxisymmetric modal solutions are expanded in a doubly infinite series of comparison functions. These include associated Legendre functions in terms of the prolate spheroidal angular coordinate, and circular functions of the circumferential coordinate. The natural frequencies and the mode shapes were obtained by the Galerkin method for each circumferential mode. Numerical results were obtained for several shell thickness-to-length ratios ranging from 0.005 to 0.1, and for various diameter-to-length ratios, including the limiting case of a spherical shell. [Work supported by Office of Naval Research and the Navy/ASEE Summer Faculty Program.]

10:00-10:15 Break

10:15

3aSA5. Predicted and measured structural intensities in the lexan and aluminum INCE standard ribbed panels. Stephen Hambric (ARL/Penn State, P.O. Box 30, State College, PA 16804), Richard Szwerz, and Henry Chang (NSWCCD, West Bethesda, MD 20817-5700)

Comparisons of predicted and measured structural intensities in vibrating lexan and aluminum ribbed panels are made. Structural intensities are predicted using finite-element models composed of plates and beams and the McPOW (mechanical power) software. Wave decomposition methods were used to estimate bending wave intensities by postprocessing velocity fields on the lexan and aluminum ribbed panels measured using a scanning laser Doppler vibrometer (SLDV). Each panel was driven at one rib with a point drive and suspended with soft springs to simulate free boundary conditions. Intensity fields are compared for each panel's (7,2) mode. The intensity fields are normalized by the power input to the panels by the point drive. The predicted intensity fields for both panels, when normalized by the power input, are nearly identical. The measured intensity fields

are similar to each other and to the predicted fields, but show some differences. The discrepancies are due to measurement uncertainties and manufacturing differences. The results help mutually validate the prediction and measurement approaches.

10:30

3aSA6. On the use of the Sturm sequence to evaluate modal density.

Philip J. Shorter (Vibro-Acoust. Science, Inc., 12555 High Bluff Dr., Ste. 310, San Diego, CA 92130, pj.shorter@vasci.com)

The modal density of a structural-acoustic subsystem is usually obtained analytically by considering the dispersion of various propagating wave types. Closed form expressions are available for the modal densities of simple beams, plates and shells (with curvature in one or two directions). However, one is often interested in subsystems with complex geometry which may possess inhomogeneous material and physical properties. The classical asymptotic formulations are not appropriate for such subsystems and numerical methods are often adopted. The most straightforward approach is to perform a finite-element-based modal analysis and count the number of eigenvalues that fall within various frequency bands. However, the computational expense associated with solving the full eigenproblem is often prohibitive. Significant computational savings can be made by employing the Sturm sequence property to evaluate the modal density. This paper describes the approach in more detail and provides a numerical example.

10:45

3aSA7. Test stand for measuring the vibration of impact-type hand-held tools.

Douglas Reynolds and Jeff Markle (Ctr. for Mech. and Environ. Systems Technol., Univ. of Nevada, Las Vegas, 4505 Maryland Pkwy., Las Vegas, NV 89154)

Vibration from hand-held impact tools is a major cause of hand-arm vibration syndrome (HAVS) in industrial workers. Evaluating the effectiveness of engineering methods for reducing tool vibration in impact tools has been hampered because of the inability to measure the impact vibration of the working end of the tools. A test fixture has been developed that

can be used to measure the impact vibration at the working end of a hand-held impact tool. These measured values can be used to determine whether or not the claimed vibration reduction of a tool is related to an effective engineering modification to the tool without reducing the productivity of the tool. Measurements were made on a rivet hammer to determine the effectiveness of a vibration-reducing mechanism applied to the tool attachment section at the working end of the tool. This mechanism reduced the vibration at the handle of the tool by 80%.

11:00

3aSA8. Cable strum self-noise cancellation for sonar towed arrays.

Vincent E. Premus (MIT Lincoln Lab., 244 Wood St., Lexington, MA 02420)

Nonacoustic self-noise observed on marine seismic streamers and towed sonar arrays represents a serious problem for acoustic source detection at low frequency. Towed array self-noise, also known as cable strum, consists of mechanical vibrations induced by vortex shedding. Transverse vibrations in the array body subject each hydrophone pressure head to local accelerations. The resultant acoustic response can be several orders of magnitude stronger than the water-borne acoustic signals of interest. In this paper, a beamspace, time-domain adaptive signal processing architecture for the coherent rejection of broadband nonacoustic self-noise is presented. The approach is based on the recognition that most vibrational modes of a towed array propagate at phase speeds substantially less than those of acoustic signals in the water column. This property supports the formation of a signal-free, strum reference using the same sensor that samples the acoustic data. The approach removes the need for additional measurement channels, such as accelerometers or strain gauges, to independently sense the undesirable distortions introduced by cable strumming. The phenomenology underlying flow-induced self-noise for towed arrays is discussed and characterized using k-w analysis. An overview of the method is presented and cancellation performance is demonstrated using snapshots of passive sonar towed array data. [Work sponsored by the Department of the Navy, under Air Force Contract F19628-95-C-0002. Opinions, interpretations, conclusions, and recommendations are those of the author and are not necessarily endorsed by the U.S. Air Force.]

Session 3aSC**Speech Communication: Alvin M. Liberman and the Development of Scientists**

Doug H. Whalen, Chair

*Haskins Laboratories, 270 Crown Street, New Haven, Connecticut 06511-6695****Invited Papers*****9:00****3aSC1. Alvin M. Liberman and the development of scientists: Chairman's introduction.** D. H. Whalen (Haskins Labs., 270 Crown St., New Haven, CT 06511)

Alvin M. Liberman contributed to the theoretical field of speech for half a century, and was still contributing at the time of his death in January of 2000. The papers in this session will celebrate some of those contributions, but will also highlight another aspect of his career that made him a powerful force: He brought out the best scientific effort of all those around him—his students, his colleagues, his children, and (even perhaps especially) his critics. Liberman's influence will continue for the lifetimes of those who knew him, and no doubt longer through his impressive body of work.

9:05**3aSC2. Alvin Liberman, infant research, and the speech module.** Peter D. Eimas (Dept. of Cognit. and Linguistic Sci., Brown Univ., Providence, RI 02912) and Joanne L. Miller (Northeastern Univ., Boston, MA 02115)

In 1957, Alvin Liberman offered a highly controversial statement based on the experimental work at Haskins Laboratories, known as "the motor theory of speech perception." Over time, the theory evolved to presume that the mechanisms underlying speech processing were biologically determined and specialized for the perception and production of speech. Speech perception was in effect not an auditory process, at least beyond the initial reception of speech, but rather part of a species-specific processing system that made language possible. We consider this view in light of selected aspects of our research on infant speech perception, which was driven in large part by the work of Liberman and his associates. We believe that the evidence in its entirety supports Liberman's idea of a species-specific processor or module that permits the acquisition and use of language, including much of the relatively early processing of speech.

9:20**3aSC3. Alvin M. Liberman and the development of Midwestern speech scientists.** James J. Jenkins (Dept. of Psych., Univ. of South Florida, Tampa, FL 33620-8200)

In the summer of 1968, Alvin Liberman came to the University of Minnesota to teach a special seminar in speech perception for the Center for Research in Human Learning. His efforts with the gifted students of the Center not only produced a piece of original research on the perception of speech, but also initiated a productive relationship between the University and Haskins Laboratories. Faculty and students from three different academic departments trekked back to New Haven to make stimuli and profit from the intensely interacting pattern of collaborative work at Haskins Laboratories. This largess was possible because of the special relationship between Haskins and NICHD that supported our use of the Haskins facilities. This relationship was typical of AI's outreach to interested scientists everywhere. At the University of Minnesota alone, over the next decade the relationship produced numerous masters theses and doctoral dissertations. More importantly, it facilitated the training and development of almost a dozen speech scientists who are continuing to advance the field of research to which AI devoted his professional life.

9:35**3aSC4. Functional role of efferent feedback to the inner ear.** M. Charles Liberman (Eaton-Peabody Lab., Mass. Eye and Ear Infirmary and Dept. of Otolaryngology, Harvard Med. School, Boston, MA 02115)

With Alvin Liberman as a father, a career in scientific research was the only one I seriously considered. However, I did stray as far as the study of hearing rather than speech. One of the major interests of my laboratory is the functional role of the neuronal system of efferent feedback to the inner ear, known as the olivocochlear pathway. One part of this olivocochlear pathway projects to the outer hair cells and constitutes a sound-evoked reflex which, when activated, reduces the sensitivity of the cochlea and raises thresholds to sound. In a series of neurophysiological studies in cat and guinea pig, we have shown that, paradoxically, this efferent feedback can act to improve the detection of transient signals in steady background noise and that the presence of an intact efferent pathway also protects the ear from acoustic injury due to overly intense sound.

9:50

3aSC5. Categorical perception and the convergence of social learning. Mark Y. Liberman (Dept. of Linguist., Univ. of Pennsylvania, 619 Williams Hall, 36th and Spruce St., Philadelphia, PA 19104-6305)

About 50 years ago, Alvin Liberman and his colleagues at Haskins Laboratories discovered “categorical perception” of certain phonetic dimensions, where discrimination is hardly better than identification. They proposed this as one piece of converging evidence for a special perceptual mode for speech, part of a species-specific evolutionary adaptation for articulate language. This special sensory-motor module, they argued, makes human speech so uniquely efficient at transmitting discrete symbol sequences in sound. Lively arguments about all aspects of this reasoning continue to this day. This paper broadens the discussion beyond perception in individuals, by suggesting a role for categorical perception in enabling communities of speakers to form and maintain consensus about the pronunciation of tens of thousands of morphemes. Through computer simulations, simple and plausible assumptions will be shown under which the “pronouncing dictionaries” of the members of a speech community will converge rapidly from random starting points. Among these assumptions, a form of categorical perception plays a key role. With it, simulations converge to a consensus from which deviations occur rarely and hardly ever spread. Without it, individual pronunciation beliefs wander chaotically across time in the phonetic space, and at a given time, differences in belief increase rapidly with social distance. The role of partly categorical perception, as in so-called “magnet effects,” will also be addressed.

10:05–10:15 Break

10:15

3aSC6. A biological basis for writing and reading. Michael Studdert-Kennedy (Haskins Labs., 270 Crown St., New Haven, CT 96511)

As Alvin Liberman taught us many years ago, speech is a complex acoustic code on the phonological units of its message, but writing is a simple cipher on that message. Yet we learn to speak more readily than we learn to read and write—a fact central to Liberman’s biological view of speed. Must there not, however, also be a biology of writing and reading? Why is writing possible at all? What property enables language, alone among systems of animal communication, to be transduced into an alternative, but no less efficient, perceptuomotor modality? The outline of an answer emerges when we recall that writing systems, like speech itself, represent not meaning, but the intrinsically meaningless units, whether syllables or phonemes, on which the hierarchy of language is raised. Such units lend themselves to transduction not only because they have no meaning, but also because, as vehicles of communicative “parity” between speakers and listeners, they are neither sensory nor motor but cognitive and abstract. The nature and origin of these particulate entities at the base of language was the topic of a paper that Liberman and I were planning at the time of his death. The paper was to have been called “A Biological Basis for Writing and Reading.”

10:30

3aSC7. The importance of parity; its implications for understanding speech perception. Carol A. Fowler (Haskins Labs., 270 Crown St., New Haven, CT 06511, University of Connecticut, and Yale Univ.)

Early in his career, Alvin Liberman obtained experimental findings that he describes in his book, *Speech: A Special Code*, as “an epiphany.” The findings revealed that listeners track speakers’ articulations. From these findings and subsequent others, Liberman developed his motor theory of speech perception. In addition, however, he set out to understand why listeners track articulation. His explanation was that humans evolved a phonetic system jointly responsible for producing coarticulated speech and for recovering intended phonetic segments from coarticulated signals. Underlying the claim that production and perception are linked in this way is the idea of parity. In Liberman’s interpretation, parity is the requirement that, for language to serve as a major component of human communication systems, listeners and talkers must agree on what set of perceivable human actions can count as components of a linguistic message. In addition, and more concretely, in their conversational interactions, listeners and talkers must typically achieve a relation of parity between messages sent and received. This paper will focus on how this parity requirement rationalizes the body of findings that listeners track articulatory gestures.

10:45

3aSC8. Articulatory gestures in the motor theory. Ignatius G. Mattingly (Haskins Labs., 270 Crown St., New Haven, CT 06511 and Dept. of Linguist., Univ. of Connecticut, Storrs, CT 06269)

According to Alvin Liberman’s motor theory, the objects of speech perception are “articulatory gestures.” But, what is an articulatory gesture? In early statements of the theory, this phrase seems to mean simply the movements of a single articulator, as idealized in traditional impressionistic-phonetic terms such as “tongue raising.” But, the notion of the articulatory gesture was greatly enriched over the years as Liberman and his colleagues took account of evidence from experiments in both speech perception and speech production. In the most recent statements of the motor theory, the articulatory gesture has become a temporary modification of the geometry or the excitation of the vocal tract, typically involving more than one articulator, often global rather than local. To Liberman and his colleagues, the human ability to distinguish and identify various such gestures from the available acoustical and optical data, despite extensive temporal overlap, appeared more remarkable than ever.

3a WED. AM

11:00

3aSC9. Perceiving speech: A developmental perspective. Janet F. Werker (Dept. of Psych., Univ. of British Columbia, Vancouver, BC V6T 1Z4, Canada)

The question of how we so instantaneously and effortlessly perceive the rapidly changing and highly encoded speech signal was one of the guiding questions in Alvin Liberman's research career. From the original presentation of the "motor theory" through the rich body of research it generated and its subsequent revisions, he posed for the field a challenge to view speech in a radically different way from what he called the "dominant" view. On my first trip to Haskins Laboratories as a young graduate student, Alvin Liberman listened carefully to and took seriously the (naive) research questions I was attempting to ask about young infants, and challenged me to consider whether my approach was sufficiently rich to capture the essence of speech perception. That challenge continues to guide my work. In this talk I will present research that I (and others) have conducted with infants and young children in an attempt to understand the nature of speech perception in the young infant, and how that changes as a function of experience—both heard and seen—with the native language.

11:15

3aSC10. Alvin M. Liberman's legacy: Haskins Laboratories. D. H. Whalen (Haskins Labs., 270 Crown St., New Haven, CT 06511)

Alvin M. Liberman came to Haskins Laboratories in June 1944 to help create a reading machine for the blind. Under Franklin S. Cooper, his group did create the first such system but, more importantly, Liberman led the way in creating an environment where the basic issues of speech and reading could be addressed by an assemblage of specialists which could not reasonably be supported by any one academic institution. The technological tools that were first available only at Haskins—controlled speech synthesis on the Pattern Playback, EMG recording, and dichotic presentation in particular—were the primary attractors, but the collaborations continued long after the equipment became less than unique. Haskins embodied the scientific ideal of the free exchange of information, across disciplinary boundaries, to a greater extent than most other institutions. A large force behind it all was Liberman's insistence on the highest technical expertise backing up theoretically interesting experiments. He was rightfully proud of the fact that Haskins results always replicated, whether the interpretation was accepted or not. The atmosphere of scientific rigor continues at Haskins, and will remain so as long as we can continue to uphold the high standards that Liberman set for us.

WEDNESDAY MORNING, 6 DECEMBER 2000

BRIGANTINE ROOM, 10:25 TO 11:45 A.M.

Session 3aSP

Signal Processing in Acoustics: Acoustical Imaging

David H. Chambers, Chair

Lawrence Livermore National Laboratory, P.O. Box 808, L-154, Livermore, California 94551

Chair's Introduction—10:25

Contributed Papers

10:30

3aSP1. Inversion methods for ultrasonic travel-time tomography.

James G. Berryman (Lawrence Livermore Lab, P.O. Box 808 L-200, Livermore, CA 94551)

Ultrasound travel-time tomography in a laboratory setting, where data can be collected without limited views in 2-D and/or 3-D imaging, has many advantages over seismic travel-time tomography in the earth. When samples to be imaged have relatively high contrast, some new methods that might be called "velocity spectroscopy" and "smart backprojection" become viable for fast and accurate imaging of any number of inclusions, as long as they are not too small and not too dense in the imaging plane. For low contrast, some simple methods have also been developed for ray bending analyses that are relatively quick and more accurate than straight ray methods, such as filtered backprojection. Applications of these methods to synthetic and real data will be discussed.

10:45

3aSP2. Recent developments in 3-D SAFT reconstruction applied to laser-ultrasonics.

Daniel Levesque, Alain Blouin, Christian Neron, and Jean-Pierre Monchalin (Industrial Mater. Inst., Natl. Res. Council of Canada, 75 de Mortagne Blvd., Boucherville, QC J4B 6Y4, Canada, daniel.levesque@nrc.ca)

The Synthetic Aperture Focusing Technique (SAFT) has been used to improve both the detectability and the lateral resolution of conventional ultrasonics. This method has recently been applied to laser-ultrasonics, an ultrasonic technique based on the generation and detection of ultrasound with lasers. Actual 3-D SAFT reconstruction is still considered computationally intensive and the pseudo-3-D SAFT, stacking line SAFT images together, is more commonly used. A recent development in 3-D SAFT reconstruction is toward improvements of frequency domain SAFT. In addition to increasing speed, the improved method (F-SAFT) includes temporal deconvolution to enhance both axial and lateral resolutions, control of the aperture to improve signal-to-noise ratio, as well as spatial interpolation in subsurface planes. These last two features also allow a reduction in the sampling requirements to further decrease both data ac-

quisition time and processing time, therefore making the technique more attractive for industrial use. All the above operations are well adapted to the frequency domain calculations and embedded in the F-SAFT processing. The performance of F-SAFT reconstruction will be illustrated using laser-ultrasonic data taken from test samples with flat-bottom holes.

11:00

3aSP3. Acoustic coherence imaging through the atmosphere.

Jonathan W. Benson (SAIC / Demaco, 100 Trade Ctr. Dr., Ste. 303, Champaign, IL 61820, jonathan.w.benson@saic.com)

The process of making images of acoustic scenes through the atmosphere has been investigated. Representations of the acoustic power with angle and the power spectrum with angle can be formed. The performance of interferometric imaging methods based on the Van Cittert-Zernike theorem has been evaluated using data collected in a series of field experiments. An array of microphones received the far-field emissions from a pair of large loudspeakers that reproduced broadband random noise signals. Both the loudspeakers and the array are located near the ground. All imaging is done in the horizontal plane. Pairs of microphones in the array are correlated to form measurements of the complex coherence. An image is then formed via an inverse Fourier transform operation. The field data show that making accurate acoustic images outdoors is difficult due to the effects of turbulence. The image distortion, signal-to-noise ratio, and variance associated with the coherence measurements were calculated as a function of the average wind speed. Even at moderate wind speeds (6 m/s) turbulence can cause an order of magnitude increase in the variance of coherence measurements. Guidelines were drawn that define the expected performance based upon the observed wind speed. Methods for image enhancement are discussed.

11:15

3aSP4. Optimal source distribution for virtual acoustic imaging.

Takashi Takeuchi and Philip A. Nelson (ISVR, Univ. of Southampton, Highfield, Southampton SO17 1BJ, UK, tt@isvr.soton.ac.uk)

When binaural sound signals are presented with loudspeakers, the system inversion involved gives rise to a number of problems such as loss of dynamic range and a lack of robustness to small errors in control performance. Regularization, often used to design practical filters, also results in poor control performance around ill-conditioned frequencies. These prob-

lems for such systems are investigated and this has resulted in the proposal of a new system concept. The system overcomes these fundamental problems by means of a conceptual pair of monopole transducers whose span varies continuously as a function of frequency. The underlying theoretical principle is described in detail. The significance is that all of the above problems that are associated with the multi-channel system inversion are solved by using this principle. The limitations with this principle are also made clear in terms of the operational frequency range. Several examples of practical solutions that can realize a variable transducer span by discretization are also described. The discretization expands the operational frequency region to be used with only a little decrease in performance. This principle is extremely useful and practical because a single transducer which can cover the whole audible frequency range is not currently available.

11:30

3aSP5. Ultrasonic fault detection by processing of signals from fixed transceiver system.

Khan Mohammad Mahmud (Div. of Appl. Phys., Grad. School of Eng., Hokkaido Univ., Sapporo, 060-8628 Japan, mah@eng.hokudai.ac.jp) and Ryoji Ohba (Hokkaido Univ., Sapporo, 060-8628 Japan)

A nonscanning method for an ultrasonic flaw detection technique is described. The system employs an M -sequence modulated ultrasonic wave as the excitation signal. Irrespective of the number of faults to be detected, the data acquisition system predeterminedly consists of a transmitter and a few receivers to be fixed at convenient locations. The cross-correlation function (CCF) between the original M -sequence and the demodulated received sequence indicates the presence of faults in the form of sharp peaks. In order to clearly detect the expected peaks corresponding to the faults in the CCF, new signal processing techniques are proposed. Experimental results (on metal plate) demonstrate the capability of the system and confirm the feasibility of the approach under heavy ambient noise condition. An algorithm, only on the basis of travel time of the signal, to determine the exact location of a fault is described. Synchronous moving average and coincidence multiplication process are found to be very effective in eliminating the false peaks from the CCF while increasing the SNR significantly. The former can be easily modified for the purpose of next generation online signal processing of an ultrasonic NDT signal. [The authors would like to express their gratitude to the Ministry of Education, Japan, for support with the scholarship as well as the funds needed for this research.]

Session 3aUW

Underwater Acoustics: Modeling

Chris T. Tindle, Chair

Applied Research Laboratory, Pennsylvania State University, Box 30, University Park, Pennsylvania 16804-0030

Chair's Introduction—7:55

Contributed Papers

8:00

3aUW1. On the use of the sonar equation approximation when computing the response of a target in shallow water. Angie Sarkissian and Louis R. Dragonette (Naval Res. Lab., Washington, DC 20375-5350, angie@aquanrl.navy.mil)

The sonar equation may be used to compute the scattering response of a target in shallow water by approximating the target to be a point scatterer. The approximation significantly simplifies the interaction of the scatterer with the medium. In the more general case, since the field produced by a source arrives at the scatterer through multipaths, it is incident on the target at various directions. The response of the scatterer must be computed at all of the incident angles to obtain the correct scattered field in shallow water. Scattering results computed using the sonar equation are compared to the more correct solution for a ribbed cylindrical shell placed in shallow water for various monostatic and bistatic geometries to examine the validity of the use of the sonar equation at various frequencies and geometries. [Work supported by ONR.]

8:15

3aUW2. Sonar performance in turbid and bubbly environments. Simon D. Richards (Underwater Sensors & Oceanogr. Dept., Defence Evaluation and Res. Agency, DERA Winfrith, Dorchester, Dorset DT2 8XJ, UK) and Timothy G. Leighton (Univ. of Southampton Highfield, Southampton SO17 1BJ, UK)

The presence of both solid particles and gas bubbles in coastal waters may have a significant effect on sound propagation, particularly at high frequencies, and may therefore be partially responsible for the observed variability in high-frequency sonar performance in shallow waters. Suspended particles increase volume attenuation through the processes of thermo-viscous absorption and scattering. Microbubbles also attenuate sound through viscous and thermal dissipation and scattering. The presence of microbubbles in the water column may also modify the sound speed and result in a dispersive medium. The effects of dilute suspensions of solid particles on the sound speed may generally be neglected for practical sonar applications. Algorithms for estimating the acoustic absorption coefficient and the speed of sound in water containing suspensions of fine mineral particles and populations of microbubbles have been developed. These have been incorporated into a sonar performance prediction model based on the ray method. Results are presented from this model which show that, for populations of suspended mineral particles and microbubbles which are commonly encountered in shallow-water environments, the predicted sonar performance is significantly different from that predicted when the additional effects are not taken into account. [© British Crown copyright 2000/DERA. Published with the permission of the Defence Evaluation & Research Agency on behalf of HMSO.]

8:30

3aUW3. High-frequency propagation modeling with the parabolic equation. David M. Fromm, Michael D. Collins (Naval Res. Lab., Washington, DC 20375), and Guy V. Norton (Stennis Space Center, MS 39529)

It is generally believed that the parabolic equation method rapidly becomes impractical above a few hundred Hertz for ocean acoustics problems. Results will be presented to illustrate that the split-step Padé algorithm [J. Acoust. Soc. Am. **100**, 178–182 (1996)] is practical out to ranges of tens of kilometers for frequencies up to at least on the order of 10 kHz. The presentation will include a discussion of modifications to the parabolic equation model to handle effects that are usually neglected at lower frequencies, such as rough surfaces, internal waves, and volume attenuation in the water column. The examples will involve propagation in surface ducts and interaction with the seafloor. [Work supported by the Office of Naval Research.]

8:45

3aUW4. A wide-angle parabolic equation for advected acoustic waves. Joseph F. Lingeitch, Michael D. Collins, Dalcio K. Dacol (Naval Res. Lab., Washington, DC 20375), Joel C. W. Rogers (Polytechnic Univ., Brooklyn, NY 11201), and William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180)

Parabolic equations are derived by factoring operators in elliptic equations. For acoustics problems involving an ambient flow, the elliptic equation contains two depth operators that do not commute with each other. Due to this difficulty in factoring the operator, the parabolic equations that have been derived for this problem are limited to small grazing angles (i.e., energy that propagates nearly horizontally). A wide-angle parabolic equation for advected waves will be described and examples will be presented to demonstrate its accuracy. As in previous investigations of this problem, the Mach number is assumed to be small. The derivation is based on the introduction of a new dependent variable that is related to the acoustic pressure by an amplitude operator that can be implemented using either of two approximations. A zeroth-order approximation (in Mach number) is useful in some cases since the error is local in range. Greater accuracy can be achieved using a first-order approximation, which can be expressed in terms of the derivative of an operator with respect to a parameter and implemented using a difference formula. [Work supported by the Office of Naval Research.]

9:00

3aUW5. A new stable formulation of an elastic parabolic equation. Wayne Jerzak, William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180-3590), Michael D. Collins, and Joseph F. Lingeitch (Naval Res. Lab., Washington, DC 20375-5330)

The parabolic equation method can be used to model acoustic-wave propagation in elastic media. Current implementations do not accurately match the shear stress between two elastic layers, for which the expression involves a second-order depth derivative. This inaccuracy could be large

in range-dependent problems. A reformulation of the elastic equations in terms of new variables permits all interface conditions to be handled accurately since no expressions contain second- or higher-order depth derivatives. However, the implementation in the new variables is found to be unstable because some evanescent modes are allowed to grow. Higher-order evanescent modes can be suppressed using a rational approximation to the depth operator with a polynomial in the denominator having a higher degree than the numerator. The lower-order evanescent modes can be controlled by imposing constraints on the choice of coefficients in the approximation. Examples are presented to show the accuracy and stability of the new parabolic equation in shallow-water environments. [Work supported by ONR.]

9:15

3aUW6. Computation of the acoustical wave-fronts in a phase space. Nick E. Maltsev (MVM Intl., 10678D Maplewood Rd., Cupertino, CA 95014, Maltsev_Nick@msn.com)

The computation of acoustical wave-fronts in 2-D and 3-D coordinate space can be performed by the computation of rays and approximation of the fronts between the rays by some interpolation formula. The amount of necessary rays increases with distance and rapidly exhausts memory and performance of the computer. It happens due to the extremely sophisticated and unsmooth nature of the ray picture. Due to the fact that trajectories of a dynamic system do not intersect in phase space, appropriate rays and fronts surface in phase space is smooth and can be easily approximated by a smaller number of rays, and this number increases very slow with distance. This number can be reduced by involving equations in variations, which generate tangent vectors to this surface. Formally, dimension of phase space is a doubled dimension of coordinate space. Using some properties of ray equations, the dimension of the phase space in the 2-D coordinate case can be reduced from 4-D to 3-D and in 3-D coordinate case from 6-D to 5-D. This approach opens new ways of fast global and local estimates of sound field and for inverse problems in underwater acoustics. This report produces a set of numerical and analytical examples.

9:30

3aUW7. Wave-front modeling. Chris T. Tindle (Appl. Res. Labs., Penn State Univ., Box 30, University Park, PA 16804-0030)

Wave-front modeling offers a direct method of finding pulse wave-forms in both deep and shallow water. The wave-front is found by tracing rays to a fixed range and finding the ray depth as a function of starting angle. Interpolation on the wave-front provides the amplitude and arrival time at the receiver depth. Phase is determined by the ray history. The expressions for the phase and amplitude are derived directly from the wave equation using WKB approximations for solutions to the depth-separated equation. Caustics are handled by treating rays as pairs and involve Airy functions which describe the smooth transition between insonified and shadow regions. On the shadow side of a caustic the rays have complex angles and travel times. Ray-tracing for complex angles can be avoided and waveforms in the shadow can be found approximately by fitting the phase function near the caustic.

9:45–10:00 Break

10:00

3aUW8. Evaluation of schemes for interpolating eigenray properties between updates in acoustic propagation modeling. Joseph A. Clark (CDNSWC, Code 734, Bethesda, MD 20084, clarkja@nswccd.navy.mil)

Acoustic simulators can model the propagation of sound in the ocean by computing a set of eigenrays between sources and receivers. A matrix of features characterizing each eigenray is computed using environmental data. The eigenray matrices are then used to modify the source signals received by each receiver. Temporal variations, introduced into the eigenrays by motions of the sources and receivers, require that the eigenray matrices be updated. Some form of interpolation between updates is needed in order to avoid the introduction of artifacts into the simulated

signals. Doppler effects must also be appropriately reproduced in the data. This talk describes a method for evaluating interpolation schemes by performing transient analyses of the simulated signals. Graphical displays used to examine the data streams will be shown. Visual and aural indications of artifacts produced by poor interpolation schemes and difficult modeling conditions will be identified. Evaluations of some improved interpolation schemes will be presented.

10:15

3aUW9. A locally conformal method for modeling rigid boundaries in the FDTD method. John B. Schneider and Julius G. Tolan (School of Elec. Eng. and Computer Sci., Washington State Univ., Pullman, WA 99164-2752)

The finite-difference time-domain (FDTD) method is a numerical technique that can be used to solve a wide variety of problems via time-domain simulations. A low-cost method has previously been presented for accurately modeling continuously varying pressure-release surfaces [Schneider *et al.*, J. Acoust. Soc. Am. **104**, 3219–3226 (1998)]. The overall simplicity of the FDTD method was preserved by requiring modifications only in the vicinity of the pressure-release boundary. Thus, the implementation of such things as sources or absorbing boundary conditions can be left unchanged. This paper describes a technique, which is similar in spirit but different in detail, that permits accurate modeling of continuously varying rigid boundaries. The technique is simple to implement, has low computational cost, and provides substantial improvements over a traditional “staircased” representation of the boundary. Accuracy is established by comparing results to those for canonical scatterers. [Work supported by ONR.]

10:30

3aUW10. The use of complex images in acousto-elastic propagation problems. John A. Fawcett (Defence Res. Establishment Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada)

In many propagation and scattering problems it is necessary to compute a half-space (for example, water over a seabed) Green’s function for a variety of source/receiver positions. For example, in the boundary integral equation method (BIEM), the kernel of the integral equation involves the Green’s function, and numerically the Green’s function must be evaluated at a large number of source/receiver positions. The distances involved may range from near-field, where the Green’s function will have a singular behavior, to the far-field. In this paper a method from the electromagnetics literature [e.g., Vitebskiy *et al.*, IEEE Trans. Antennas Propag. **44**, 143–151 (1996)] is described which reduces the usual wave number integral representation to a small number of image sources with complex positions. This representation is valid over a large range of source/receiver distances and hence is appropriate for efficiently evaluating the Green’s function for many source/receiver positions. Numerical examples are presented illustrating the accuracy of the method for both fluid and elastic seabeds.

10:45

3aUW11. Monte Carlo model for underwater ambient noise fields. Daniel L. Hutt, Andrew L. Rosenfeld, and Paul C. Hines (Defence Res. Establishment Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 2Z7, Canada, daniel.hutt@drea.dnd.ca)

Three-dimensional ambient noise fields possessing realistic cross-spectral properties have been simulated using the Monte Carlo method. In the model, weighted monochromatic sources are distributed about an array of sensors to create the desired directional characteristics of the noise field. The hydrophones can be positioned arbitrarily to form any desired array structure and orientation. Noise signals are represented in the frequency domain where phase shifts due to propagation are most naturally applied. Examples of simulated cross spectra for a variety of noise fields will be presented. Good agreement with theory is found for cases in which analytical solutions are possible.

11:00

3aUW12. Simulation of underwater intensity vector measurement: Effect of errors and noise. Daniel L. Hutt, Paul C. Hines, and Victor W. Young (Defence Res. Establishment Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, daniel.hutt@drea.dnd.ca)

Pressure and pressure-gradient signals from an array of hydrophones can be processed to yield the three-dimensional sound-intensity vector of an acoustic field. In underwater applications, the technique may prove useful for near-field measurements of acoustic intensity and characterization of ambient noise fields. Unfortunately, the accuracy of the magnitude and direction of the calculated intensity vector is degraded by many factors including: fluctuations in the received signal, the presence of ambient noise, system noise, phase error, and errors in channel sensitivity. To investigate the effect of these errors, a numerical simulation of the pressure field due to signal sources and ambient noise was developed. The model is used to calculate the variance of intensity measurements under different noise conditions and to simulate the effects of channel gain and phase imbalance. The results are found to be in good agreement with theoretical expressions. The simulation was also used to compare the performance of a six-element orthogonal intensity array and a four-element tetrahedral intensity array in terms of gain against noise and variance in estimated direction to a sound source.

11:15

3aUW13. Numerical results for an approximate form of the nonlocal small slope approximation scattering strength. Shira L. Broschat (School of Elec. Eng. and Computer Sci., Washington State Univ., P.O. Box 642752, Pullman, WA 99164-2752)

Voronovich introduced the nonlocal small slope approximation (NLSSA) as a generalization of the small slope approximation to explicitly include nonlocal interactions. He showed that the NLSSA generally accounts for double scattering in the high-frequency limit. Broschat and Thorsos presented numerical results for the lowest-order NLSSA scattering strength for two-dimensional pressure-release surfaces. Their results

agreed well with Monte Carlo integral equation results and, in particular, were better than the higher-order SSA results at low forward grazing angles. However, the computational cost was extremely high, and results were unobtainable at low grazing angles. In this paper, we discuss the results obtained by making an *ad hoc* approximation to the lowest-order NLSSA scattering cross section that reduces the computational complexity of the integration substantially. In addition, we discuss the difficulties of the numerics and how they are handled. Numerical results for the scattering strength are presented for 2-D pressure-release surfaces. For the cases presented, the NLSSA and its approximate form give virtually the same results. Also, results at low grazing angles are obtainable and accurate with the approximate form. [Work supported by ONR.]

11:30

3aUW14. Spectral integral representations of multistatic scattering from sediment volume inhomogeneities. Henrik Schmidt (Dept. of Ocean Eng., MIT, 77 Massachusetts Ave., Cambridge, MA 02139) and Kevin D. LePage (SACLANT Undersea Res. Ctr., La Spezia, Italy)

A perturbation approximation for multistatic scattering from sediment volume inhomogeneities has been developed using a Fourier–Bessel spectral integral representation of the source and receiver Green’s functions. This approach is particularly efficient for modeling scattering from horizontally isotropic, three-dimensional distributions of scatterers in waveguides with arbitrary horizontal layering, and is accurate for scatterers in fast bottoms near the critical angle and for scatterers in layers with background sound-speed gradients. The theory has been implemented in the OASES code and the model has been exercised over broad bandwidths in order to explore the temporal and angular evolution of bistatic scattering for a variety of environmental and experimental scenarios. Examples show that the scattered field received on a bistatic vertical line array shows a distinct evolution in time and angle which can be simply related to the source–receiver geometry and the location of the contributing scatterers at any given time. Similarly, the horizontal angular evolution of bistatic scattering shows a time-angle trajectory which may be easily interpreted using similar geometrical arguments. [Work partially supported by ONR.]

WEDNESDAY MORNING, 6 DECEMBER 2000 BIG CANYON I/II ROOM, 9:00 A.M. TO 12:00 NOON

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

to be held jointly with the

U.S. Technical Advisory Group (TAG) Meeting for ISO/TC 108 Mechanical Vibration and Shock (and Subcommittees ISO/TC 108/SC1, SC2, SC3, SC5, and SC6)

R. J. Peppin, Chair S2

5012 Macon Road, Rockville, Maryland 20852

D. J. Evans, Vice Chair S2 and Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108,
Mechanical Vibration and Shock

National Institute of Standards and Technology (NIST), 100 Bureau Drive, Stop 8221, Gaithersburg, Maryland 20899-8221

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will present reports of their recent progress on writing and processing various shock and vibration standards. There will be a report on the interface of S2 activities with those of ISO/TC 108 (the Technical Advisory Group for ISO/TC 108 consists of members of S2, S3, and other persons not necessarily members of those Committees), including plans for future meetings of ISO/TC 108 and/or its Subcommittees.

Scope of S2: Standards, specifications, methods of measurement and test, and terminology in the fields of mechanical vibration and shock, and condition monitoring and diagnostics of machines, but excluding those aspects which pertain to biological safety, tolerance and comfort.

Session 3pAA

Architectural Acoustics: Amphitheater Acoustic Design and Sound Control for Nearby Communities

Dana S. Houglund, Cochair

ACODA Acoustics, LLC, 9603 Orchard Drive, Englewood, Colorado 80111-3503

Leslie D. Blomberg, Cochair

Noise Pollution Clearinghouse, P.O. Box 1137, Montpelier, Vermont 05602-1137

Chair's Introduction—1:00

Invited Papers

1:05

3pAA1. Amphitheater noise, a community perspective. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601, les@nonoise.org)

Amphitheaters and amplified music often provoke strong community reaction. Leaders of approximately 20 community organizations near existing or proposed amphitheaters were surveyed, and three were selected for case studies. Their experience provides insight into the affected public's perception of the concert industry, acoustical consultants, ordinances, and regulations.

1:25

3pAA2. Outdoor concert sounds—Searching for acceptability criteria for communities adjacent to amphitheaters. William J. Cavanaugh (Cavanaugh Tocci Assoc., Inc., 327 F Boston Post Rd., Sudbury, MA 01776)

Most municipal, state, and federal codes and regulations are inadequate for protecting residential communities adjacent to concert amphitheaters from annoyance due to sound of high-level amplified music, as well as from other concert-related sounds. From numerous consultations involving siting of new and evaluating existing amphitheaters, the author has evolved a criterion which appears to predict with sufficient accuracy the likely response of residential occupants to amphitheater sounds. This experience suggests that if the intrusions represented by the 1-percentile A-weighted levels are not more than 5 dB above the pre-existing (i.e., nonconcert) background represented by the 90-percentile levels, the intrusions are sufficiently immersed in and masked by the normal background and most communities are satisfied. However, some neighbors find the mere audibility or detectability of amphitheater sounds objectionable and measurement is at best difficult where the sound levels are of the same order of magnitude as the normal community background sound. The experience in developing and applying this signal-to-noise based criterion to several amphitheater projects, one of which has been in operation for over 15 years, is reviewed.

1:45

3pAA3. Evolution of metrics used in “real world” concert sound level management systems. Richard G. Cann (Grozier Technical Systems, Inc., 157 Salisbury Rd., Brookline, MA 02145, rcann@grozier.com)

With the ever increasing popularity and proliferation of outdoor amphitheaters and the advent of more powerful amplification systems serving high-energy performing groups, nearby residents are inevitably threatened by higher sound levels. To address these concerns, acoustical consultants have specified monitoring systems that must use simple metrics for the management and control of music facility sound emissions. These metrics must be pragmatic; the visiting artist and sound engineer, the concert promoter, the concert facility management personnel, and even local regulatory personnel must be able to easily understand and interpret their meaning. Furthermore, the management system must accommodate rapid real-time implementation of necessary management control measures and store accurate, permanent, electronic records for future reference. This paper describes the evolution of the Grozier sound level management systems that use simple, accurate metrics developed for several amphitheater venues with diverse local conditions and requirements.

2:05–2:15 Break

2:15

3pAA4. Using time-variant electronic architecture to improve outdoor acoustics. Steve Barbar (LARES Assoc., 9170 Rte. 108, Ste. 404, Columbia, MD 21045)

The outdoor environment offers little of the acoustical support provided by the traditional architecture and surface treatments found in good concert halls or opera houses. While these treatments can be used as part of a shell or enclosure where the musicians perform, incorporating architectural treatments that improve listening enjoyment for the audience, or return energy for the musicians, is much more difficult. Sound reinforcement systems used for classical music or opera outdoors can assist in improving the impact of the direct sound when used carefully. Such systems, however, can do little to improve the sense of envelopment and warmth that are

an important part of the listening experience for this type of performance. Recent developments in technology have produced a system that uses time variance to reduce the effects of acoustic feedback. These systems have a significant advantage in gain before coloration over previous methods (typically 12 dB or more). When used outdoors, this system provides reflected and reverberant energy for both musicians and the audience. Additionally, this energy can help to mask noise intrusion from sounds outside the designated seating area. The following is what we have determined to be important parameters for the successful implementation of such a system.

2:35

3pAA5. Community noise reduction through sound system renovation, a case study. David E. Marsh (Pelton Marsh Kinsella, 1420 W. Mockingbird Ln., Ste. 400, Dallas, TX 75247) and Jack E. Randorff (Randorff and Assoc., Inc., Ransom Canyon, TX 79366)

Noise intrusions from the Miller Outdoor Theater in Houston, Texas caused recurring complaints from the surrounding upscale community. Measurements confirmed that events at this performing arts venue regularly produced levels of 75 dBA and above at residents homes approximately 1,300 ft from the proscenium stage. The project goals were to provide the theater with a high-quality sound system, capable of producing nominal levels of 95 dBA at the audio mixing console and 85 dBA at the farthest lawn seats (340 ft from the stage) while not exceeding levels of around 55 dBA in the nearby neighborhood. These goals were met through loudspeaker selection (including some customization) along with careful loudspeaker zoning, elevation, and aiming. Two different computer-based programs were used for acoustical modeling—one to design and predict the sound systems coverage throughout the seating area and another to predict noise propagation from the theater into the community. Both programs were used to produce maps indicating sound levels as color contours which simplified presentation of these data to the client.

2:55

3pAA6. The acoustics of the outdoor Shakespeare theater—Rutland, UK. Peter A. Mapp (Peter Mapp Assoc., Colchester C03 4JZ, UK)

The outdoor Shakespeare Amphitheatre at Tolethorpe (Rutland, UK) is an unusual form of theater in that the actors perform in the open air while the audience is enclosed in a lightweight Teflon fabric auditorium. This unique format produces some unusual acoustic properties and problems. The early reflection patterns from and around the stage are quite different from a normal enclosed proscenium theater. The Teflon walls and roof, while effectively acoustically transparent at low and lower midfrequencies, become highly reflective at high frequencies. This not only impacts the transmission of sound through the structure but also speech transmission and intelligibility within the auditorium. The hyperbolic curved-roof surfaces and flat parallel walls were found to cause undesirable sound focusing and reflections detrimental to speech intelligibility. The acoustics of the space was investigated by means of both directional and binaural time-domain spectrometry techniques as well as more traditional acoustic measures. The paper presents the results of these measurements and outlines the remedial treatments adopted.

WEDNESDAY AFTERNOON, 6 DECEMBER 2000

PACIFIC SALONS A–F, 1:55 TO 3:00 P.M.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Ralph T. Muehleisen, Chair

*Civil, Environmental and Architectural Engineering, University of Colorado, Campus Box 428,
Boulder, Colorado 80309-0428*

Chair's Introduction—1:55

Invited Papers

2:00

3pID1. Hot topics in acoustical oceanography. James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Acoustics continues to play an ever-increasing role as a means of exploring the ocean. In every area of oceanography, acoustic techniques are being used to make measurements that would have been considered science fiction only a few scant years ago. In this lecture, we will show some of the recent, sometimes startling advances that have been made in oceanography using acoustics. We will also endeavor to put the acoustics measurements in the context of other types of measurements that are made. Starting with physical oceanography, we will look at the advances that have been made, going from the largest (gyre) scales to the smallest (microscopic turbulence) oceanographic scales. We will then turn to marine geology, and examine how various acoustic techniques both map the structure of the marine sediments and also help us understand the formation of sedimentary structures. Finally, we will examine marine biology, where acoustic techniques are helping us quantify the abundance of marine life and, at a more basic level, understand marine ecosystems.

2:20

3pID2. Recent developments in biomedical ultrasound. Lawrence A. Crum (Appl. Phys. Lab., 1013 NE 40th St., Univ. of Washington, Seattle, WA 98105)

Biomedical ultrasound has seen remarkable advances in recent years. By utilizing the properties of nonlinear acoustics, diagnostic ultrasound has shown increased applicability to a wide number of clinical conditions and pathologies. Techniques such as harmonic imaging and the use of ultrasound contrast agents (stabilized microbubbles) have enabled such long-sought goals as noninvasive determination of myocardial perfusion to be clearly within our grasp. Advancements in semiconductor miniaturization have led to the construction of ultrasonic scanners that are now hand-held, and together with telemedicine techniques, it is now reasonable to expect that diagnostic ultrasound will soon be the doctor's stethoscope. An even more promising future is seen for therapeutic ultrasound. Although the mechanism is not yet clearly understood, ultrasound can transiently permeabilize cell membranes, thus permitting the delivery of therapy to specific sites within the body; indeed, together with drug-carrying ultrasound contrast agents, "site-specific drug delivery" is now in clinical trials. Finally, the application of High Intensity Focused Ultrasound can induce coagulative necrosis at well-controlled sites within tissue. When imaging and therapy are combined, "image-guided, transcatheter, bloodless surgery" devices are now under development. With acoustics, "Star Trek medicine" is just around the corner.

2:40

3pID3. Recent studies of the lip reed: Low-tech experiments and stroboscopic observations. R. Dean Ayers (Dept. of Phys. and Astron., California State Univ., Long Beach, 1250 Bellflower Blvd., Long Beach, CA 90840, rdayers@csulb.edu)

There are currently three basic models for the brass player's vibrating lips: (1) Helmholtz's outward-striking or swinging door model, (2) a sliding door model, with motion that is transverse to the air stream, and (3) a hybrid, two-dimensional model that combines those two motions. Time-domain computer simulations of these models have been carried out by S. Adachi and M. Sato [J. Acoust. Soc. Am. **97**, 3850–3861 (1995); **99**, 1200–1209 (1996)]. The experience of brass players agrees qualitatively with the behavior of the hybrid model. Feedback is optimized near each peak of the input impedance curve, and from there it is easy to drop the playing frequency but not to push it upward. (Buzzing on an isolated mouthpiece yields the same behavior, even though there is no regime of oscillation.) A graph of playing frequency versus an operationally defined "lip frequency" is obtained by using a side hole in the backbore. Agreement with results from the hybrid model is good for the first few modes but not the higher ones. Stroboscopic images of Rayleigh waves on the upper lip suggest that a more advanced model should have at least two masses performing orbital motion. Demonstrations will be included. [Work supported in part by the Scholarly and Creative Activities Committee at CSULB.]

WEDNESDAY AFTERNOON, 6 DECEMBER 2000

YAWL ROOM, 1:00 TO 2:35 P.M.

Session 3pMU

Musical Acoustics: Asian Musical Instruments and Traditions II

Thomas D. Rossing, Chair

Physics Department, Northern Illinois University, DeKalb, Illinois 60115

Chair's Introduction—1:00

Invited Papers

1:05

3pMU1. Acoustics of Asian drums. Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115)

Drums have played an important role in nearly all musical cultures, including those of Asia. The tabla and mrdanga of India are specially tuned to have harmonic partials so that they convey a strong sense of pitch. The very important drums in Balinese and Javanese gamelans, on the other hand, do not have a strong pitch. The large o-daiko of Japan have modes of vibration that are somewhat similar to those in the bass drum. Smaller drums, such as the tsuzumi, have spectra that vary considerably with playing style.

1:35

3pMU2. Tuned sets of bells from ancient China. Robert W. Bagley (Dept. of Art & Archaeology, Princeton Univ., 105 McCormick Hall, Princeton, NJ 08544-1018, rwbagley@princeton.edu)

Tuned sets of bronze bells appeared very early in China, sometime around 1200 B.C., and over the next millennium underwent an extraordinary development. The most spectacular set known so far consists of 65 bells found in 1978 in the tomb of Marquis Yi of Zeng (deceased 433 B.C.), ruler of a small and previously unknown state in central China. The set has a range of five octaves, within which is a three-octave chromatic stretch; the largest bell weighs 200 kg. Despite much study of this and other sets, it remains unclear

how tuned sets were designed and cast; it has not been possible to deduce a scaling rule from measurements of bells. Invoking both archaeological evidence and musical considerations, a hypothesis will be presented to suggest how tuned sets might have been invented and elaborated.

2:05

3pMU3. Sounds of ancient Korean bells. Yang-Hann Kim (Ctr. for Noise and Vib. Control (NOVIC), Dept. of Mech. Eng., KAIST, 373-1 Science Town, Taejon-shi, Korea)

This paper reviews the sounds of bells cast during the Shilla Dynasty. One unique characteristic of Korean bells is that they create beats. The beats of various ancient bells are summarized and analyzed with regard to mass, and other geometrical dimensions. Other interesting and unique features of the old Korean bells are that they have sound pipes on top and hollows below the bells. The types of pipes and hollows are also studied and reported. In fact, the pipe is a high-pass filter so that it dissipates high-frequency sound as fast as it can, and the hollow is a resonator that can somehow sustain a beating sound. Bells cast during the Shilla and Korea Dynasty and their sound characteristics are investigated and summarized.

WEDNESDAY AFTERNOON, 6 DECEMBER 2000

CALIFORNIA SALON 3, 2:00 TO 3:05 P.M.

Session 3pNS

Noise and Archives and History: Acoustical Society's Role in Noise Control

Tor S. D. Kihlman, Chair

Applied Acoustics, Chalmers University of Technology, Goteborg S-412 96, Sweden

Chair's Introduction—2:00

Invited Paper

2:05

3pNSa1. The Acoustical Society of America's historical contributions to noise and its control. Leo L. Beranek (975 Memorial Dr., Ste. 804, Cambridge, MA 02138-5755, beranekleo@mediaone.net) and William W. Lang (29 Hornbeck Ridge, Poughkeepsie, NY 12603-4205)

The Society's contributions to noise and its control, a subfield of the broad subject of acoustics, have been significant and constant over the years since the first noise paper was presented at the Society's first meeting in 1929. The major contributions are considered in an historical context—during the period before World War II when the Society first produced national standards for instrumentation used for noise measurement, during the War and the Postwar Era when there was considerable government support for airborne acoustics research, during the 1950s when *Noise Control* magazine was published, followed by the 1960s and the later years of the 20th Century. Of the ASA members expressing interest in the 13 different subfields of acoustics, the number concerned with noise is near the top of the list. But ASA is no longer alone. Nearly a dozen other professional organizations in the U.S.A. have groups that are involved with noise and its control. Over the past three-quarters of a century, ASA has played a leadership role in developing the subfield of acoustics focused on noise control, and in producing a broad range of American National Standards on noise and vibration. In the future, ASA must take concrete, positive steps to retain its position of leadership.

A NOTE ABOUT THE ASA HISTORY LECTURE SERIES

In 1997, the ASA Committee on Archives and History conceived a plan for a series of invited lectures on each of the technical areas of the Society which would memorialize the significant achievements and milestones of each of its twelve technical committees and one interdisciplinary technical group during the first three quarters of the Society's first century.

With the cooperation of the technical committees, distinguished individuals are selected to review the history of their particular technical speciality and present a lecture which shows how that activity has developed and has contributed to the Society at large and to the broad field of acoustics as well. At the meetings in Columbus, Ohio and Atlanta, Georgia, the first four History Lectures were presented: Gabriel Weinreich on Musical Acoustics, Robert Beyer and David Blackstock on Physical Acoustics, Stanley Ehrlich on Engineering Acoustics and Ewart Wetherill on Architectural Acoustics. At each subsequent meeting two additional lectures will be scheduled including those in Noise and Speech Communication at this meeting in Newport Beach.

The invited lecturers have been asked to prepare a written manuscript of their lectures which will be published in a commemorative book for the 75th Anniversary of the Society to be celebrated in 2004. The Archives and History Committee and the individual technical committees/group welcome comments and suggestions on both the History Lecture Series and on the proposed ASA Diamond Anniversary Book. Volunteers to assist the committees would be most welcome too. Contact Henry Bass, Chair, Committee on Archives and History, pbass@sunset.backbone.olemiss.edu

Session 3pPA

Physical Acoustics: Thermoacoustics II

Larry A. Wilen, Chair

Department of Physics and Astronomy, Ohio University, Clippinger Laboratories, Athens, Ohio 45701-0882

Chair's Introduction—1:40

Contributed Papers

1:45

3pPA1. Time-averaged pressure drop produced by an abrupt change in a resonator's cross section. A. Doller, Anthony A. Atchley, and Roger Waxler (Grad. Prog. in Acoust., The Penn State Univ., State College, PA 16804, aatchley@psu.edu)

The behavior of steady, incompressible flows through constrictions, expansions, bends, and branches in pipes is very well documented and commonly used to design fluid handling systems over a wide range of Reynolds numbers, surface roughness, etc. However, the state of knowledge of high-amplitude acoustic flows in similar structures is much less mature, as demonstrated by recent developments in thermoacoustic devices [S. Backhaus and G. W. Swift, *J. Acoust. Soc. Am.* **107**, 3148–3166 (2000)]. The research presented here, motivated by this lack of understanding, focuses on the nature of predominately reactive, high-amplitude acoustic fields in the vicinity of changes in resonator cross section. Oberst-style resonators [H. Oberst, *Akust. Z.* **5**, 27–38 (1940)], consisting of two sections of straight brass pipe of different diameters and joined through either step-like or conical couplers, were constructed. The resonators are driven by a graphite piston at one end, and rigidly terminated at the other. The acoustic pressure is measured at four locations along the resonator. Of particular interest is the time-averaged pressure drop across the coupler section. Measurements of this pressure drop are reported as functions of drive amplitude for the different coupling configurations. [Work supported by the Office of Naval Research.]

2:00

3pPA2. Practical considerations for the design and construction of an acoustic dynamometer. Robert W. M. Smith, John F. Heake, and Steven L. Garrett (Penn State Appl. Res. Lab, P.O. Box 30, State College, PA 16804)

An instrumented, variable acoustic load of the type described by Fusco [Fusco *et al.*, *J. Acoust. Soc. Am.* **91**(4), 2229–2235 (1992)] and Gardner [D. L. Gardner and G. W. Swift, *Cryogenics* **37**(2), 117–121 (1997)] has been implemented for the testing of high power (greater than 1.5 kW) electrodynamic loudspeakers for thermoacoustic refrigerator applications. The load serves to facilitate performance measurements for these loudspeakers under various operating conditions, a role analogous to a dynamometer in conventional rotating machinery. Attached to a primary acoustic resonator, the acoustic dynamometer consists of a 2-in angle globe valve and other flow constrictions connected to an acoustic compliance. The design of these flow restrictions employed the acoustic “minor loss” model developed by Swift *et al.* [Swift *et al.*, *J. Acoust. Soc. Am.* **105**, 711–723 (1999)]. Measurements of the dynamometer performance agreed with the calculated acoustic impedance to within 16%. At powers exceeding 1 kW, agreement between the acoustic load dissipation and other measurements of loudspeaker power agree to within 3%. To illustrate the utility of an acoustic dynamometer, a brief summary of measurements on two moving-magnet loudspeakers will be presented. [Work supported by ONR.]

2:15

3pPA3. Wet-walled thermoacoustics. William V. Slaton and Richard Raspet (Dept. of Phys. and Astron. and The Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677, wmslaton@meta3.net)

An analytic solution of sound propagation in wet-walled tubes with a temperature gradient will be presented. The tube contains an inert gas-vapor mixture with a thin layer of condensed vapor coating the tube wall. The vapor phase condenses and evaporates from this layer during an acoustic cycle. This phased evaporation and condensation modifies traditional energy density and wave number equations. It is found that mass and heat transport act in parallel thereby increasing the acoustic energy density for proper choice of gas mixtures. This increase in energy density due to inert gas-vapor working fluids can be quite large and even dominate the traditional thermal transport contribution. The use of phased evaporation-condensation in gas mixtures provides a path to higher energy density in thermoacoustic systems. The effects on sound propagation, energy density, and efficiency due to the presence of an inert gas-vapor working fluid in tubes with wet walls will be discussed. [Work supported by the Office of Naval Research.]

2:30

3pPA4. Torsionally resonant toroidal standing-wave resonator. Scott R. Diehl, Steven L. Garrett (Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804), Milo Friesen, Roy L. Kessinger, Jr., Paul Receveur, and Leonard Schrank (Lynx Motion Technol. Corp., New Albany, IN 47151-0408)

A toroidal resonator with a rigid barrier can be used to generate high-amplitude standing waves if the resonator is oscillated in a torsional mode about its central axis [S. Garrett, US Pat. No. 5,953,921 (Sept. 21, 1999)]. In the following, a 2-1/2-in diameter schedule-40 steel pipe is bent into a 13-in radius torus to form the resonator. The torus is then deformed along its cross section to detune harmonic overtones and suppress shock wave formation. The torsional drive force is provided by rigidly attaching the resonator to the housing of a modified Lynx Model T468 Direct-Drive Servo Motor [Kessinger *et al.*, US Pat. No. 5,744,896 (April 28, 1998)]. The resonator/motor housing and shaft/rotor assemblies are made into a mechanical 2-DOF torsional oscillator by coupling their moments-of-inertia through an elastic restoring force. The torsional restoring force is provided by a quartet of semi-cylindrical stainless steel (17-7 PH) springs mounted between the resonator/housing and shaft/rotor assemblies. Measurements of modal anharmonicity and peak acoustic pressures at resonance will be reported and compared to DeltaE model results. [Work supported by the Office of Naval Research.]

2:45

3pPA5. Nostack thermoacoustics. Ray Scott Wakeland and Robert M. Keolian (Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804, wakeland@psu.edu)

The “stack” used in conventional standing-wave thermoacoustics depends on intrinsic thermodynamic irreversibilities for its operation. One approach to increasing efficiency is to eliminate the stack, retaining a small gap between heat exchangers in a standing wave. While a “nos-

tack'' device has a temperature span that is limited by the achievable pressure amplitude, it is expected to have greatly increased thermodynamic efficiency compared to a stack-based device with the same power and operating temperatures. Losses associated with flow through the heat exchangers, however, are expected to be higher in a nostack device. The standard theory of low-amplitude thermoacoustics does not apply to nostack devices, where the displacement amplitude is larger than the heat exchangers and the gap. As an alternative, an idealized thermodynamic model of nostack is combined with separate calculations of conduction loss and flow losses, including so-called "minor losses" arising from sudden changes in cross-sectional flow area. The results suggest that by positioning the heat exchangers very close to a pressure antinode to reduce flow velocity, efficiencies similar to Stirling-like thermoacoustic devices might be achieved with the nostack idea. [Work supported by the Office of Naval Research and the Pennsylvania Space Grant Consortium.]

3:00

3pPA6. Alternative-geometry traveling-wave thermoacoustic devices.

Larry Wilen and Gabriela Petculescu (Dept. of Phys. and Astron., Clipping Lab, Ohio Univ., Athens, OH 45701, wilen@helios.phy.ohiou.edu)

Ceperley has discussed the idea that an acoustic traveling wave passing through a regenerator executes a Stirling cycle [P. H. Ceperley, J. Acoust. Soc. Am. **77**, 1239–1244 (1985)]. He also emphasized that for an acoustic Stirling device to be useful, the impedance must be high or else viscous losses will dominate. Backhaus and Swift recently found a way to produce a useful device with high impedance and traveling wave phasing using a toroidal geometry [S. Backhaus and G. W. Swift, J. Acoust. Soc. Am. **107**, 3148–3166 (2000)]. We will discuss possible ways to realize these conditions with alternative geometries, and we will present some preliminary results for a refrigerator employing one such configuration. [Work funded by the Office of Naval Research.]

WEDNESDAY AFTERNOON, 6 DECEMBER 2000

CATAMARAN/TRIMARAN ROOMS, 1:00 TO 3:00 P.M.

Session 3pPP

Psychological and Physiological Acoustics: Potpourri (Poster Session)

Thomas Z. Strybel, Chair

Psychology Department, California State University at Long Beach, 1250 Bellflower Boulevard, Long Beach, California 90840

Contributed Papers

All posters will be on display from 1:00 p.m. to 3:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:00 p.m. and contributors of even-numbered papers will be at their posters from 2:00 p.m. to 3:00 p.m.

3pPP1. Effects of basilar membrane roughness and nonlinearity on stimulus frequency otoacoustic emissions. Carrick L. Talmadge (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677) and Arnold Tubis (Purdue Univ., West Lafayette, IN 47907)

Stimulus frequency otoacoustic emissions (SFOAEs) may be evoked via basilar membrane (BM) nonlinearity, BM roughness (random inhomogeneities), or a combination of these properties. Emissions generated by BM nonlinearity alone have phase gradients that are too small to account for the observed emission fine structure [e.g., Shera and Guinan (1999)]. Linear reflection via BM roughness can account for the main phase characteristics of the emissions [e.g., Shera and Zweig (1993); Talmadge *et al.* (1995)]. It is shown theoretically that, contrary to what is expected from naive considerations concerning phase derivatives and group delays, the physical time delays associated with both emission mechanisms are approximately equal to the round-trip cochlear wave travel time from the cochlear base to the tonotopic place for the stimulus frequency. A theoretical framework is presented for describing the combined effects of roughness and nonlinearity on the emission fine structure. For the case of cochlear roughness and a weak nonlinearity, it is shown that in some instances, the nonlinearity can significantly modify the level and phase behaviors of the fine structure and actually enhance the fine structure patterns. The implications of these results for the stimulus level dependence of SFOAE fine structure are also explored.

3pPP2. Hydrodynamics of otoacoustic emissions. Taha Jaffer and Hans Kunov (Inst. for Biomaterials and Biomed. Eng., Univ. of Toronto, 4 Taddlecreek Rd., Toronto, ON L4J 2X7, Canada, taha@ibme.utoronto.ca)

In this paper, a novel boundary condition for the classical cochlear model will be outlined that allows the realistic simulation of tone-burst transiently evoked otoacoustic emissions (TEOAEs). OAEs are sounds that are emitted from the inner ear and have been related to hearing function. Classical cochlear models suffer from an inability to realistically simulate OAEs because of the simplistic treatment of the hydrodynamics at the interface between the oval window, stapes, and cochlear fluid. In the classical cochlear model, there are two boundary conditions; the first describes the helicotrema, and the second details how fluid is displaced by the oval window in response to sound entering the ear. In this model, OAE production cannot be simulated because once the stimulating sound is stopped, the oval window is immobile and incapable of transmitting acoustical energy back out of the inner ear. In this paper, this boundary condition will be reconsidered in light of the additional fluid displacement caused by the movement of the cochlear partition. The addition of this boundary condition yields realistic simulations of tone-burst OAEs, both in time evolution and frequency content.

3pPP3. Modulated otoacoustic emissions reveal features of the human cochlear impedance at low frequencies. Torsten Marquardt (Dept. of Physiol., Univ. College London, Gower St., London WC1E6BT, UK, torsten.marquardt@ucl.ac.uk), Johannes Hensel, Guenther Scholz, and Dieter Mrowinski (Virchow Hospital, Augustenburger Platz 1, 13353 Berlin, Germany)

Basilar membrane (BM) displacement by loud low-frequency (LF) tones periodically alters the level of distortion-product otoacoustic emissions (DPOAEs) as a function of the phase of the modulating LF tone. For three human subjects the level of a modulating LF tone (15–320 Hz) was adjusted to maintain a constant DPOAE-modulation depth, indicating a constant periodic LF BM displacement. The resulting modulation LF-tone levels match the iso-loudness contours (ISO/R226) except for an antiresonance centered at about 55 Hz. Here, an increase in level of approximately 5 dB above the iso-loudness contour is required to maintain constant periodic LF BM displacement. The antiresonance separates two distinct regions of the cochlear impedance: a slope of 12 dB/octave at frequencies below 55 Hz suggests a mass-controlled impedance resulting from perilymph flow through the helicotrema; a slope of 6 dB/octave above 55 Hz suggests the existence of a traveling wave with resistive impedance. Psychophysical threshold experiments did not show the antiresonance. Antiresonances have been observed previously in animal experiments using cochlear microphonics or perilymph pressure measurements to estimate BM displacement. Modulated DPOAEs are a noninvasive method to investigate the BM movement at low frequencies.

3pPP4. Effect of tonal duration on auditory evoked potentials as a function of age. Jodi Ostroff, Kelly McDonald, Claude Alain (Rotman Res. Inst., Baycrest Ctr. for Geriatric Care, Univ. of Toronto, 3560 Bathurst St., Toronto, ON M6A 2E1, Canada), and Bruce Schneider (Univ. of Toronto, Mississauga, ON L5L 1C6, Canada)

Behavioral research has shown that temporal resolution decreases with age as evidenced by higher-gap-detection thresholds in older versus young adults with normal hearing. In the present study, we examined the neural correlates of the temporal resolution function in young, middle aged, and older subjects, using a brief Gaussian-shaped 2-kHz tone that increased by 2 ms in separate trial blocks. All stimuli elicited a negative potential that peaked at about 100-ms poststimulus (N1). The N1 amplitude increased linearly with increases in tonal duration. The growth function of the N1 amplitude was similar across the three age groups, although the slope of the function in older adults was shallower. Results suggest that the N1 response can resolve temporal information on the order of 2 ms. This electrophysiologic threshold parallels behavioral findings. The slower growth function of the N1 response in older adults may reflect a deficit in temporal processing. [Work supported by the Medical Research Council of Canada.]

3pPP5. The form of the expansive nonlinearity in cochlear implant stimulation. Monita Chatterjee (Dept. of Auditory Implants and Percept., House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057)

In electrical stimulation with cochlear implants, loudness grows as an expansive function of stimulus amplitude. The mathematical form of the nonlinearity is difficult to discern, primarily because of the limited current range over which measurements can be made without reaching uncomfortably loud sensations. Both power and exponential functions appear to fit measured loudness estimates reasonably well over the dynamic range. For a given pulse train, increasing the pulse phase duration (D) and/or the separation between active and return electrodes in the cochlear (M) results in increased loudness. Power function fits to the loudness functions show that while the exponent of the power function is insensitive to these stimulus parameters, the multiplicative constant is not. In contrast, the best fitting exponential functions show systematic changes in the exponent with changing M and D . The two kinds of functions also yield different predictions for the level-dependence of modulation sensitivity. When the nonlinearity is a power function (as in normal hearing), modulation sen-

sitivity is predicted to show negligible dependence on level; when the nonlinearity is exponential, a strong level-dependence is predicted. In cochlear implant listeners, modulation sensitivity is strongly level-dependent, tipping the balance in favor of an exponential nonlinearity. [Work supported by NIDCD.]

3pPP6. Speech recognition under conditions of frequency-place expansion and compression. Deniz Baskent (Dept. of Biomed. Eng., Univ. of Southern California, Los Angeles, CA 90089) and Robert Shannon (House Ear Inst., Los Angeles, CA 90057)

The mapping of frequency information onto the correct cochlear place is critical for speech recognition. In cochlear implants, the cochlear tonotopic range represented by the electrode array is smaller than the acoustic frequency range used in the speech processor. While this condition utilizes a wide range of speech frequency information, it results in a compression of the tonotopic pattern of speech information delivered to the brain. An alternative approach is to take the core frequency range necessary for speech and expand its representation in the cochlea, like an “acoustic fovea.” This study examined the effect of linear frequency-place compression and expansion on speech intelligibility for various cochlear locations and number of spectral channels. These conditions were presented to normal-hearing listeners using a noise band vocoder. The cochlear tonotopic range was held constant by employing the same carrier bands for each condition, while the frequency range of the analysis bands was changed. For each condition, the result was compared to that of the perfect tonotopic match, where the carrier and the analysis bands were perfectly matched. Speech recognition in frequency-place expansion and compression was always equal to or poorer than the matched condition. [Work supported by NIDCD.]

3pPP7. Horizontal and vertical minimum audible angle: Effects of stimulus onset asynchrony and azimuth. Thomas Z. Strybel, Eishi Adachi, and Simon Jeon (Dept. of Psych., California State Univ., Long Beach, Long Beach, CA 90840)

The effects of stimulus onset asynchrony (SOA, the onset-onset time difference) on the minimum audible angle (MAA) were measured for horizontal and vertical orientations and two azimuth locations. Each participant was tested in 16 conditions created by all combinations of four SOAs (25, 50, 100 and 400 ms), two orientations (horizontal and vertical), and two azimuth locations (0 and 90 deg). MAAs were measured with an adaptive procedure. The stimulus was high pass noise with a lower frequency cutoff of 4 kHz and amplitude of 55 dB A-weighted. The burst duration was 50 ms with a rise/decay time of less than 1 ms. As expected, horizontal MAAs were lowest at 0 deg and increased substantially at 90 deg; vertical MAAs were only minimally affected by azimuth. The effect of SOA was similar for both orientations and azimuths. In all conditions, the MAA decreased exponentially with SOA. The slope of the horizontal function at 0 deg was slightly steeper than the slopes of the remaining conditions.

3pPP8. Objective discrimination of extensity: Concurrent versus successive sound sources. Prisilia Tirtabudi and David Perrott (Dept. of Psych., California State Univ., Los Angeles, 5151 State University Dr., Los Angeles, CA 90032, tirtabudi@hotmail.com)

This study compared human ability in processing and detecting concurrent versus successive sound stimuli and the degrees of angular separation between sources. Bursts of broadband white noise at 85 dB (SPL) were presented with duration of either 1000, 300, 100, 30, or 10 ms. Angles between speakers for each duration time were at 4.5°, 9.0°, 18°, 27°, and 36°. Two pairs of bursts were presented at a time, either simultaneously or sequentially. Participants were to discriminate whether the first pair of bursts had a larger angular separation between sources than the second pair. Listeners' performance were almost similar between concur-

rent versus successive stimuli when the durations were 1000, 300, and 100 ms. However, as the duration of the stimuli dropped to 30 and 10 ms, so did the performance in the concurrent condition. In addition, as the angles between the two speakers grew farther apart in both conditions, the degrees of jnd (MAA) were also increasing. Results will be discussed in terms of the auditory system's ability to resolve spatially distributed sources that are set off either concurrently or sequentially.

3pPP9. Psychophysical tuning curves and filter shapes sharpen over time. Jennifer L. Repovsch and Sid P. Bacon (Psychoacoustics Lab., Dept. of Speech and Hearing Sci., Arizona State Univ., P.O. Box 87-1908, Tempe, AZ 85287-1908, spb@asu.edu)

Psychophysical tuning curves (PTCs) measured in simultaneous masking usually sharpen as a short signal is moved from the onset to the temporal center of a longer duration masker. Filter shapes derived from notched-noise maskers have not consistently shown this effect. One possible explanation for this difference is that the PTC paradigm uses a fixed signal level, whereas filter shapes typically have been derived from results where the noise masker was fixed in level. In the present study, PTCs and filter shapes were measured with a signal fixed in level at 10 dB SL. The masker was either a pure tone (PTCs) or a noise with a spectral notch placed symmetrically or asymmetrically about the 2-kHz signal frequency (filter shapes). The 20-ms signal was presented at the onset or temporal center of the 400-ms masker. For a given signal delay, the PTCs were substantially sharper than the filter shapes. Importantly, however, for all four subjects, the PTCs and the filter shapes sharpened on both the low- and high-frequency sides as the signal moved from the beginning to the temporal center of the masker. This sharpening was considerably greater for the filter shapes. [Work supported by NIDCD.]

3pPP10. Up/down asymmetry in FM identification. Michael Gordon and David Poeppel (Dept. of Biol., CEBH, Dept. of Linguist., Cognit. Neurosci. of Lang. Lab., Univ. of Maryland, College Park, MD 20742)

Temporal limits of the human auditory system with regard to identifying the direction of frequency modulation (FM) were investigated: How brief can frequency change be without compromising FM direction identification? Such abilities may relate to processing formant transitions. In a 2AFC paradigm, subjects binaurally heard linear FM sweeps rising or falling in frequency between 1000 and 1500 Hz. Ten FM rates were used for both FM directions. The FM rate was modulated by varying stimulus duration from 5 to 640 ms. Each FM stimulus was presented 20 times, in pseudorandom order, for a total of 400 trials. Subjects ($n=24$) identified the direction of frequency change. There was a marked difference between up and downward FMs. For upward FMs, 90% accuracy in direction identification was achieved at a rate as fast as 17 Hz/ms. A considerably slower rate, 3 Hz/ms, was needed for identifying downward FMs with an accuracy of 90%. Humans are better at identifying rapid upward FMs than downward ones. This result is congruous with studies revealing lower thresholds for detecting rapid upward FMs (Collins and Cullen, 1978) and higher magnitude evoked potentials elicited by upward FMs (Maiste and Picton, 1989). [Work supported by Comparative and Evolutionary Biology of Hearing Training Program.]

3pPP11. Greek bilingual listeners perceive the tritone stimuli differently from speakers of English. Magdalene H. Chalikia, Angelina M. Norberg (Psych. Dept., Minnesota State Univ., Moorhead, MN 56563), and Lena Paterakis (The American College of Greece, Aghia Paraskevi, Greece)

When listeners are presented with pairs of octave-complex tones related by a tritone interval (a half-octave), they hear the pattern as ascending or descending, according to an individual pitch class template. Deutsch (1991) has claimed that this template may be influenced by language. In order to test this hypothesis, data from Greek bilingual listeners were

collected and compared with data from Texas, California, and the south of England. The results show significant differences in how Greek listeners hear the tritone stimuli, as compared to listeners in the other groups. There is also evidence that the Greek listeners may have developed two different pitch class templates, possibly representing the influence of English and the influence of Greek.

3pPP12. Listeners in Sweden perceive tritone stimuli in a manner different from that of Americans and similar to that of British listeners. Magdalene H. Chalikia and Fredrik Leinfelt (Dept. of Psych., Minnesota State Univ., Moorhead, MN 56563)

The tritone stimuli consist of two tones, each of which has octave-related harmonics. The two tones are separated from each other by a half-octave (a tritone interval). The tones are typically presented successively, and listeners decide whether or not the second tone in the pair is perceived as ascending or descending, relative to the first one. Responses tend to vary across listeners for the same stimuli, and for different stimuli. In this study, participants from Stockholm, Sweden, were presented with such tritone pairs and were asked to determine if they heard each pair as ascending or descending. Deutsch's (1991) hypothesis, that language background may influence the perception of the tritone stimuli, was tested. All participants were speakers of both English and Swedish. Their data were compared to data of English monolinguals tested in Texas, and California, as well as listeners from the south of England. Preliminary results suggest that the overall perception pattern in Sweden differs from that found in Texas and California, but tends to be similar to the perception pattern found with English listeners.

3pPP13. The perception of tritone stimuli by listeners in the Midwest. Meredith Haugen and Magdalene H. Chalikia (Psych. Dept., Minnesota State Univ., Moorhead, Moorhead, MN 56563)

The tritone stimuli [Deutsch (1986)], consist of two tones, each with octave-related harmonics. The tones are separated from each other by a tritone (half-octave) interval. The tones are presented successively, and listeners decide whether or not the second tone in the pair is perceived as ascending or descending, relative to the first one. Listeners tend to hear the tritone pattern as ascending or descending, according to an individual pitch class template, which Deutsch (1991) claims may be influenced by language or English dialect. Some similarities have been found among people from different areas of the U.S., suggesting a canonical manner of perceiving the tritone stimuli, probably propagated by the media [Ragozzine and Deutsch (1994)]. The present study tested the media exposure hypothesis, by presenting tritone stimuli to listeners who grew up in the Midwest. Results show that Midwesterners show a different perceptual pattern relative to that of listeners from Texas and the south of England, but the same pattern with that heard by listeners from California. Spectral envelope effects were also examined, and suggest that the perceptual patterns found under high and low envelopes are similar to those found for Californians, but different from those found with listeners from Texas.

3pPP14. Investigation of the influence of visual stimuli on the breakdown of the echo threshold. Thomas Djelani (Institut fuer Kommunikationsakustik, Ruhr-Universitaet Bochum, 44780 Bochum, Germany)

Audiovisual interaction is already known from the ventriloquism effect [W. R. Thurlow and Ch. E. Jack, *Percept. Mot. Skills* **36**, 1171-1174 (1973)]. Synchronously presented auditory and visual stimuli are perceptually grouped and localized as one object. The performed experiments aimed to explore a possible influence of visual stimuli on echo suppression. They focused on investigating whether visual stimuli can also cause a breakdown of the echo threshold. One hypothesis is that the breakdown of echo suppression is caused by violations of the listeners' expectations

concerning the presented scenario. The question is whether this implausibility can be attributed to acoustical reasons. Two different kinds of visual implausibilities were tested by measuring echo threshold with an adaptive paradigm. In the build-up phase of both experiments, visual and auditory parts of the stimuli have been presented from the same direction. In the test phase, the visual stimulus' direction was altered. The main result of the investigations was that both experiments did not provide any evidence of a visually induced breakdown of the echo threshold. This holds true, even if the position of the auditory event corresponding to the direct sound was changed due to the ventriloquism effect by changing the corresponding visual stimulus' direction of incidence.

3pPP15. Monitoring and localizing simultaneous real-world sounds: Implications for the design of spatial audio displays. Diane L. Guettler (Dept. of Psych., California State Univ., Long Beach, 1250 Bellflower Blvd., Long Beach, CA 90840), Robert S. Bolia (Air Force Res. Lab., Wright-Patterson Air Force Base), and W. Todd Nelson (marchFIRST)

Numerous researchers have investigated the extent to which human listeners are able to localize free-field and virtual acoustic stimuli. Despite the abundance of this research, few studies have focused on the number of sounds a listener is able to localize simultaneously, and even fewer have employed naturally occurring or real-world sounds. Results from these types of experiments may have important implications at the basic research level within the fields of perception and attention, and may also prove useful for guiding the design of spatial audio displays in applied disciplines such as engineering and human factors. In the present study, listeners localized between one and three virtual acoustic stimuli, the positions of which were restricted to the horizontal plane under factorial combinations of seven spatial locations, three set sizes (one, two, or three sources), and three distinct acoustical sounds of a real-world nature (e.g., a crying baby). The paradigm employed required participants to monitor a set of simultaneously presented sounds and localize one specific sound when prompted. Performance was measured in terms of average localization error and percentages of front-back confusions. These results will be discussed in terms of their implication for the design of spatial audio interfaces for complex, multitask environments.

WEDNESDAY AFTERNOON, 6 DECEMBER 2000

CALIFORNIA SALON 5, 1:30 TO 2:45 P.M.

Session 3pSA

Structural Acoustics and Vibration: Acoustic Scattering from Elastic Structures

J. Gregory McDaniel, Chair

Department of Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02537-0339

Contributed Papers

1:30

3pSA1. Scattering from a partially filled elastic-shelled sphere. John A. Fawcett (Defence Res. Establishment Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada)

Scattering from spherical scatterers can be analytically formulated in terms of a harmonic series solution. The case of a shelled sphere with an evacuated or filled interior has been particularly well studied. In this paper, the case of a partially filled sphere is considered. A solution for the scattered field is sought in terms of a harmonic Legendre function series. A standard series expansion is used for the shell and the exterior of the sphere. A modified expansion, which satisfies the interface condition within the interior, is used for the inner region. Unlike the standard spherical scattering problem, the system of equations for the various orders (for a fixed azimuthal order) of the Legendre functions are now coupled. Numerical computations are presented for a steel-shelled sphere with various levels of water-fill.

1:45

3pSA2. Causal and unique reconstruction of nonminimum reflection phase for a class of submerged structures. Cory L. Clarke and J. Gregory McDaniel (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

An approach is presented for uniquely determining the phase of an acoustic reflection from an object whose impedance magnitude approximates the acoustic impedance of the surrounding medium. This approximation defines a semiannular region in the complex impedance plane that lies between the rigid and pressure-release limits. When this region is mapped onto the complex reflection coefficient plane, the approximation allows for a wide range of reflection coefficients. A Taylor series expansion for the square of the reflection coefficient magnitude is used to estimate

the phase of the reflecting impedance. The causality condition is invoked to recover the magnitude of the impedance from its phase. Once the magnitude and phase of the impedance is known, the complex-valued reflection coefficient algebraically follows regardless of whether or not the reflection coefficient is minimum phase. The approach is useful in experimental situations that only allow the accurate measurement of reflection magnitude and not phase. As an example, the approach is demonstrated on one-dimensional numerical simulations that simulate uncertainty in target location and ambient noise. [Work supported by ONR.]

2:00

3pSA3. Backscattering by tilted cylindrical shells above the coincidence frequency caused by helical flexural waves. Florian J. Blonigen and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

Generalized leaky Lamb waves excited by ultrasound incident on cylindrical shells in water produce important backscattering effects. At a certain critical tilt angle, leaky waves launched along the meridian of the cylinder have been shown to lead to large backscattering enhancements [S. F. Morse *et al.*, *J. Acoust. Soc. Am.* **103**, 785–794 (1998)]. At angles less than the critical tilt, these leaky waves follow helical paths and also make significant contributions. In the present work, experiments using tone bursts were performed on a stainless steel shell to investigate the contributions of a_0 leaky Lamb modes. The tone bursts were of sufficient duration to superpose helical wave bursts of successive circumnavigations, along with the meridional contribution near the critical tilt, to arrive at a steady state backscattering amplitude for the cylinder. This was compared against an approximate numerical partial wave series solution and a ray theory solution as a function of the tilt angle. The data follow the basic shape of the theoretical curves. However, the experimental amplitudes are

more suppressed at higher frequencies resulting from losses due to reflection of leaky waves at the cylinder's end. [Work supported by the Office of Naval Research.]

2:15

3pSA4. Acoustic scattering of metallic springs in elastic medium. Jinmin Zhang (DSO Natl. Labs., 20 Science Park Dr., Singapore 118230), Vijay K. Varadan, and Vasundara V. Varadan (Penn State Univ., State College, PA 16801)

In applications such as ultrasonic immersion testing, damping of flow noise in pipes and ducts and coating of acoustic control surfaces, it is very useful to reduce the reflection of acoustic waves from surfaces. Significant echo reduction can be achieved by addition of an anechoic coating. According to some general design guidelines for a single layer of homogeneous coating, a single layer of 15 dB coating must be at least one wavelength thick. For absorbing acoustic waves of low frequencies, inhomogeneities such as metallic fillers, cavities and irregular metallic helices can be introduced into the lossy matrix to substantially enhance the acoustic absorption by acoustic scattering. In this presentation, the acoustic scattering of a single metallic spring embedded in an infinite elastic medium has been investigated using a boundary element formulation. The cross sections of the scattered longitudinal and shear waves have been calculated for various spring sizes, orientations and wave numbers. It will be shown that helical inclusions are more efficient in scattering longitudinal incident waves than spherical inclusions of equal volume. Steel helices are found to be more efficient in converting the incident longitudinal waves into shear waves than helical cavities of equal size.

2:30

3pSA5. Lamb waves scattering by inhomogeneities in elastic waveguides. Natalya S. Gorodetskaya and Victor T. Grinchenko (Inst. of Hydromechanics NAS of Ukraine, 8/4 Zhelabov St., Kiev 252057, Ukraine, vin-igm@gu.kiev.ua)

Different types of conjunctions of elements with various mechanical properties are widely used in engineering practice. Such conjunctions strongly affected the process of acoustic energy transmission in constructions. To study the influence of the physical and geometrical discontinuities in conjunctions on the wave propagation process the composite waveguides have been considered. The method of the boundary problem solution taking into account local singularities of stress is presented. The quantitative data for characteristics of transmitted and reflected waves at the conjunction are given in a wide band of frequencies. At relatively low frequencies the estimations of energy flows by beam model are correct. When the frequency increases, the wave field near conjunction becomes more complicated. As a result the ratio between transmitted and reflected energy becomes a sharply changing function of frequency even in the domain when only one propagating wave exists. This phenomenon is caused by strong excitation of evanescent waves. Excitation of these waves was due to strong mismatch between the form of the incident wave and first transmitted and reflected waves. At the frequency when transmitted energy was minimum the mismatch was maximum. At the same frequency the maximum of normal stresses on the conjunction has been found.

WEDNESDAY AFTERNOON, 6 DECEMBER 2000

NEWPORT ROOM NORTH, 1:00 TO 3:00 P.M.

Session 3pSC

Speech Communication and Signal Processing in Acoustics: Signal Processing for Speech (Poster Session)

Shrikanth S. Narayanan, Chair

AT&T Laboratories-Research, 180 Park Avenue, Room D106, Florham Park, New Jersey 07932

Contributed Papers

All posters will be on display from 1:00 p.m. to 3:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:00 p.m. and contributors of even-numbered papers will be at their posters from 2:00 p.m. to 3:00 p.m.

3pSC1. Intelligibility of aircraft loudspeakers. Glenn E. Warnaka, Mark E. Warnaka (Future Technologies, LLC 1612 S. Allen St., State College, PA 16801), Peter Mapp (Peter Mapp Assoc., Colchester C03 4JZ, UK), and Bruce Shimizu (C&D Aerosp., 5412 Argosy Dr., Huntington Beach, CA 92649)

Commercial aircraft audio systems provide audible messages including greetings, details of the flight, comments on entertainment, and important safety information. Hence, the speech intelligibility of aircraft audio systems is of great importance. Distributed-mode loudspeakers (DMLs) are known to have wider sound dispersion, especially at higher frequencies, than conventional cone loudspeakers. As aircraft loudspeakers are located relatively close to the passengers, the narrower dispersion and beaming of the high frequencies from conventional cone loudspeakers can cause loss of intelligibility for passengers not on the axis of the loudspeaker and produce uneven sound distribution within the cabin. DMLs have demonstrated intelligibility superior to conventional loudspeakers in other applications such as public address systems. DMLs have other advantages in aircraft installations. They are very lightweight and can even be built into the internal trim, resulting in a nearly zero-weight loudspeaker. Their essentially flat configuration also requires less volume. This

paper presents a comparative evaluation of the speech intelligibility of current aircraft loudspeakers and DMLs as determined by the STI (Speech Transmission Index) using the RASTI (Rapid Speech Transmission Index) method in accordance with RTCA document DO-214. Tests were performed under representative conditions and for different seating arrangements.

3pSC2. Improved speech intelligibility in noise with a single-microphone noise reduction technique. Michael J. Nilsson, Xiaoling Fang, Robert M. Ghent, Jr., Patrick Murphy, and Victor H. Bray, Jr. (SONIC innovations, 2795 E. Cottonwood Pkwy., Ste. 660, Salt Lake City, UT 84121)

Reception threshold for sentences (RTS) as measured by a modified version of the HINT test [Nilsson *et al.*, J. Acoust. Soc. Am. **95**, 1085–1099 (1984)] were collected on 26 hearing-impaired listeners fit binaurally with digital hearing aids incorporating a nine-channel spectral subtraction technique of single-microphone noise reduction. Thresholds were measured in quiet and noise presented at a zero degree azimuth with the

subjects listening unaided, aided without noise reduction, and aided with noise reduction and no modification to the gain function when noise reduction was activated. Additional data included soundfield thresholds as a frequency-dependent measure of the impact of the noise reduction algorithm on sensitivity to soft sounds. RTS in quiet reveal a significant effect of noise reduction [$F(3,69)=46.9, p<0.01$] with noise reduction increasing RTS 3.4 dB relative to no noise reduction, from 40.4 dB (A) to 43.8 dB (A). RTS in noise reveal a significant effect of noise reduction [$F(3,69)=18.7, p<0.01$] with noise reduction decreasing RTS 1.8 dB relative to no noise reduction, from 0.0 dB S/N to 1.8 dB S/N. Soundfield thresholds reveal a loss of sensitivity mainly in frequencies below 1000 Hz. The noise reduction algorithm will be reviewed and discussed.

3pSC3. Automatic speech recognition for mobile hand-held devices.

Richard Rose, Shrikanth Narayanan, S. Parthasarathy, Aaron Rosenberg (AT&T Labs-Res., 180 Park Ave., Florham Park, NJ 07932), and Bojana Gajic (Norwegian Univ. of Sci. and Technol., Trondheim, Norway)

The implementation and evaluation of an automatic speech recognition (ASR) based task for a mobile, hand-held device is presented. The use of ASR on mobile devices will make environmental noise, and the design of transducers and compensation algorithms for dealing with noise, critical issues for the success of ASR based services on these devices. A description of the task is provided along with a set of compensation techniques that are used to compensate speaker independent hidden Markov models (HMMs) with respect to environment and transducer variability. Data were collected in a prototype application environment for a form-filling directory access task with speech and pen input and text output (48 subjects in an office environment; 21 in a cafeteria). A technique for combined environment/transducer compensation is presented for this task and is shown to significantly reduce the effects of environmental mismatch. The overall performance degradation was reduced from 41.7% to 10.4% for speech spoken through a far-field microphone in an office environment, and from 79.2% to 39.8% for the same transducer in a noisy cafeteria environment.

3pSC4. Soft-decision speech signal estimation. Leonid Krasny (Ericsson, Inc., Development Dr. 7001, Research Triangle Park, NC 27709, euslekr@rtp.ericsson.se)

Speech is often corrupted by background noise which degrades the performance of coding and recognition algorithms. This scenario is especially true in hands-free mobile communications where noise can be generated by the car engine, the road, and wind hitting the windows. It is essential to reduce the noise level without distorting the original speech signal. A well-known technique for solving this problem is spectral subtraction (SS), where the power spectrum of the noise is estimated when speech is absent and used to suppress noise when speech is present. Normally, a voice activity detector (VAD) is used to determine the presence of speech. In the conventional approach, VAD and SS algorithm act independently. Specifically, the speech signal is first detected by the VAD, and once the presence of the speech is detected, the SS algorithm is used to estimate the signal. Although each task can be optimized separately, the resulting algorithm is not optimal. In this paper, we present a novel approach for estimating the speech signal in which the detection and SS tasks are combined optimally. Our approach employs a soft-decision algorithm for detecting and estimating the speech signal, which leads to improved speech quality.

3pSC5. Time-dependent signal representations that are independent of sensor calibration. David N. Levin (Dept. of Radiol., Univ. of Chicago, Chicago, IL 60637, d-levin@uchicago.edu)

Suppose that the sensor arrays of two speech recognition systems, S_x and S_y , are calibrated differently so that they have different responses (x and y) to the same sound. It is difficult for pattern recognition software to

handle such variable sensory data. This paper shows how to avoid this problem: each device transforms its sensory data elicited by an evolving sound into the *same* "canonical" representation so that it is suitable for analysis by pattern recognition algorithms. Assume that the two systems have previously been exposed to samples of the same collection of sounds. These "training" data induce an affine-connected differential geometry on the sound manifold, which the two sensor arrays represent in different coordinate systems (x and y). Without communicating, the two systems can utilize this geometry to construct the same coordinate-independent (scalar) representation of any time-dependent sound of interest. This methodology is demonstrated by applying it to simulated data from devices with two sensory channels (e.g., for measuring the frequency and amplitude of tones). This approach mimics human perception, which also tends to be constant across systems (individuals) with different sensory distortions, as long as they have been exposed to similar collections of stimuli in the past.

3pSC6. Crime-detection speaker recognition system. Igor I. Gorban (Inst. of Math. Machines and Systems, 42 Ave. Acad. Glushkov, Kiev, 03187 Ukraine, gorban@immsp.kiev.ua), Vitalij N. Magera (Police State Res. Ctr. of Crime-Detection Expertise, Kiev, 01024 Ukraine), and Sergey V. Levij (Natl. Tech. Univ. of Ukraine, Kiev, 04055 Ukraine)

Crime-detection automatic speaker verification and identification (CASVI) system oriented to nonprofessional experts was described at the 137th ASA Meeting [I. I. Gorban, N. I. Gorban, and A. V. Klimentko, J. Acoust. Soc. Am. **105**, 1353 (1999)]. At the current meeting another speaker recognition system for professional crime-detection experts is presented. It consists of special units, Pentium-type computer, and programs realized speech-processing algorithms in automated and automatic modes. The system gives the possibility to research records made on different tape recorders, to fit tape speed, to decode speech records, to suppress noises, to correct frequency and nonlinear record distortions, to sort sounds, to measure and calculate different discriminate features, characteristics, and parameters usually used by crime-detection experts in their practical work, to compare the records, to form libraries and archives, and to make reports. The system has a simple and comfortable interface tailor-made for crime-detection expert aims. Police exploiting of the system has shown that utilization of the system raises productivity of expert work sometimes and essentially improves authenticity of phonoscope expertise.

3pSC7. Thai monophthongs classification using CDHMM. Ekkarit Maneenoi, Somchai Jitapunkul, Visarut Ahkuputra (Digital Signal Processing Res. Lab., Dept. of Elec. Eng., Faculty of Eng., Chulalongkorn Univ.), and Sudaporn Luksaneeyanawin (Chulalongkorn Univ.)

This paper presents Thai monophthongal vowels recognition. The Thai monophthongs were qualitatively recognized by the 3-state left-to-right continuous density hidden Markov model. The 18 monophthongs are qualitatively 9 different vowels, each of which has two members, short and long. The LPC cepstral coefficients were used and the temporal cepstral derivative was additionally utilized to compare efficiency of the additional feature with the single feature. Qualitative recognition means that short and long vowel pairs were categorized in the same model. Thai polysyllabic words were used in this research. The database consists of 2100 training phonemes from 30 speakers and 1378 testing phonemes from a different group of 20 speakers, respectively. The highest recognition rate of the single feature obtained from 18-order LPC cepstral coefficients is 86.983 percent, while the recognition rate of the 16-order LPC cepstral coefficients plus temporal derivative is 94.580 percent. The results indicate that all the LPC cepstral coefficients associated with temporal derivative have better recognition accuracy than those of LPC cepstral coefficients. It is concluded that the additional temporal derivative can improve recognition rate. The misclassification is analyzed and it is concluded that this resulted from excessively overlapped distributions of vowels in the low- and back vowel groups, respectively.

3pSC8. A machine learning technique for evaluating cues to stop place. Madelaine C. Plauche' (Dept. of Linguist., 1203 Dwinelle Hall, Univ. of California, Berkeley, CA 94720)

This paper is situated in a long line of phonetic studies that seek to determine and qualify the acoustic cues humans use to identify stop place. Acoustic information relevant to the identification of stop place resides in the relative amplitude (Cooper *et al.*, 1952; Ohde *et al.*, 1983) and spectral characteristics of the burst (Blumstein *et al.*, 1979; Stevens, 1999), as well as in formant transitions (Dorman *et al.*, 1996) and voice onset time (Klatt, 1975). Consonant confusion studies suggest that vocalic context is an important factor in the perception of neighboring stops. In the environment of a high front vowel (*i/*), for example, cues such as formant transitions and VOT may be neutralized, resulting in a dependence on transient features that are secondary cues in other contexts (Plauche' *et al.*, 1997). The present study estimates the effect of vocalic context on the relative ranking of acoustic cues for stops in CV sequences using a machine learning algorithm (decision trees) that evaluates a database of 1500 (English) CV tokens and their values for the acoustic features: (1) VOT, (2) energy and power of the burst and release, (3) spectrum at the burst, and (4) formant transitions into the following vowel.

3pSC9. Pitch determination by fundamental-harmonic extraction using MUSIC algorithm. Takahiro Murakami and Yoshihisa Ishida (Meiji Univ., 1-1-1, Higashi-mita, Tama-ku, Kawasaki, 214-8571 Japan)

Short-term analysis pitch determination was executed by using multiple signal classification algorithm, which is an eigen-based subspace decomposition method proposed by Schmidt (Ph.D. thesis, Stanford University, 1981). The MUSIC spectrum, based on subspace principles, is sharply peaked at the frequencies of the sinusoidal components of speech signals without almost receiving the influence of the noise. Since the harmonic structure of the power spectrum of speech signals becomes unclear in a high-frequency domain, fundamental-harmonic extraction is performed using the band-limited MUSIC spectrum. The influence of the noise decreases further by limiting the frequency domain to be analyzed, and calculation time has been shortened greatly. The IDFT of the logarithmic MUSIC power spectrum exhibits a strong peak at the position equal to the pitch period like the cepstral method. Results of pitch determination for male and female Japanese vowels illustrate that the proposed method is more excellent than the cepstral method and can estimate the pitch frequency not being influenced by the noise.

3pSC10. Measurement and visualization of speech quality with VIPER. Harald Mundt and Peter Daniel (Neutrik Cortex Instruments, Erzb.-Buchberger-Allee 14, D-93051 Regensburg, Germany, daniel@neutrik-cortex.de)

In this presentation the new VIPER speech quality measurement (VSQM) will be presented. Audible differences between two versions of a sound can be visually inspected in VIPER through the help of auditory distance spectrograms. The objective instrumental measures provided by VIPER will be validated with the results of listening experiments on speech quality (MOS) and compared with the performance of other speech quality measures (PAMS, TOSQA, MNB, PSQM+, PSQM). The comparison is based on the ITU-T Series P. supplement 23 database. The database has been developed in order to characterize the performance of the ITU-T speech codec G.729. Therefore, a wide range of different speech codes, in noisy and clean conditions, in various tandem conditions, with bit and frame errors of various intensity, was tested for different languages (English, French, Japanese). In order to quantify the assessment performances, the linear correlation coefficient and the standard deviation based on the mapped objective data and the subjective data are used.

3pSC11. Comparison between physical and auditory parametrization of speech corpus for the unit selection in text-to-speech synthesis. Minoru Tsuzaki (ATR Spoken Lang. Translation Res. Labs., 2-2 Hikoridai, Seika-cho, Soraku-gun, Kyoto, 619-0288 Japan, minoru.tsuzaki@slt.atr.co.jp)

It is important to design an appropriate cost function to improve the quality of speech produced by a corpus-based text-to-speech (TTS) synthesis. Although the final product of the TTS system is evaluated perceptually, the definition of cost functions has to be based on the physical parameters of speech signal. And a cost function will become inappropriate if the discrepancy between the physical and the perceptual measures becomes critically large. The discrepancy could be caused by two main factors, i.e., (1) inappropriate selection of parameters, and (2) inappropriate scaling. One may judge that the selection of physical parameters is inappropriate if changes in a single physical parameter result in changes in multiple perceptual attributes. One may also judge that the scaling of physical dimension is inappropriate if the mapping function of the physical space to the perceptual space appears to be critically nonlinear. Aiming at a better definition of cost functions, a signal analysis based on the standard linear prediction coding (LPC) method and one based on the models of the peripheral auditory system (DSAM-AIM) are compared. The results of principal component analysis for Japanese speech databases of a corpus-based TTS system, CHATR, are presented.

WEDNESDAY AFTERNOON, 6 DECEMBER 2000

SCHOONER/SLOOP ROOMS, 1:10 TO 3:00 P.M.

Session 3pUW

Underwater Acoustics: Propagation

Michael G. Brown, Chair

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

Chair's Introduction—1:10

1:15

3pUW1. Sound propagation using impulsive sources on the New Jersey Continental Shelf. Mohsen Badiey (Univ. of Delaware, Newark, DE 19716) and William M. Carey (Boston Univ., Boston, MA 02215)

An acoustic experiment was performed on the New Jersey Continental Shelf in a well surveyed area with known bottom and sub-bottom properties. The measurement system was a calibrated vertical array 14–18 km from a known geological feature. Calibrated airgun shots (AG) and special small omni-directional explosives shots (SO) were used to generate impulsive sounds monitored by a calibrated source hydrophone. The repeatability of the SO and AG shots was excellent. The sea state during this experiment was between 4 and 6 on the Beaufort scale and the ship was held in position on either side of the geological feature. The sound velocity profiles were stable over the measurement period due to the fully developed seas. A mixed layer depth of approximately 20–30 m was followed by a steep thermocline to a depth of 48 m with a gradient of approximately 0.5 sec and a relatively isothermal layer to the bottom which ranges between 70 m at the source location and 80 m at the receiving array. Measurements received on the vertical array were variable but over short periods of time remarkably consistent. Time of flight enabled the separation of the first three modes whose shape and arrival times were to agree with calculations. Comparisons of the arrival structure changes due to the feature and estimates of the backscatter from this feature are presented in relation to the mode arrival statistics for different groups of shots.

1:30

3pUW2. Measurement-based geoacoustic modeling for the Acoustic Characterization Test III. Ilya Rozenfeld, William Siegmann (Dept. of Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY 12180, rozeni@rpi.edu), William Carey (Boston Univ., Boston, MA 02215), and Peter Cable (GTE/BBN Technologies, New London, CT 06320)

The Acoustic Characterization Test III was performed in the oceanographically complex Strait of Korea to provide accurate measurements of sound transmission and coherence (array signal gain) under known environmental conditions. Bottom sampling and sub-bottom surveys, coupled with archival geophysical information, provided the basis for geoacoustic profiles of sound speed, density, and attenuation versus depth. These profiles and the measured bathymetry and water sound speed were used as input parameters for parabolic equations computations. Very good agreement was obtained between measured and calculated narrow-band transmission loss by employing slightly modified geoacoustic profiles and an attenuation profile with a near-water-sediment interface power-law frequency dependence of 1.8. This power law was determined through the use of an effective attenuation coefficient and a least-square analysis of transmissions from five narrow-band tones between 47 and 604 Hz. These results are consistent with measurements in other sandy–silty areas. Using these parameters, comparisons were made between independent measurements and calculations of broadband transmissions and of signal spread, and excellent agreement was found. [Work supported by ONR.]

1:45

3pUW3. Focused arrivals in shallow water propagation. Harry A. DeFerrari, Neil J. Williams, and Hien B. Nguyen (Univ. of Miami, Rosenstiel School of Marine and Atmospheric Sci., 4600 Rickenbacker Cswy., Miami, FL 33149)

Observations of acoustic pulse transmissions for several sites in the Florida Straits consistently show that a single intense pulse dominates the arrival pattern. For these sound channels, the propagation is by refracted modes that interact with the bottom below the critical angle resulting in favorable low-loss transmission. The intense arrivals come about from focusing of the waveguide and are always observed for downward refracting sound speed profiles whenever the source and receiver are deep in the channel. The foci correspond to caustics of ray theory or to the phase-

stationary summation of many modes. Such arrivals are often 20 dB higher than others and are of interest in that they will be the only detectable features at long ranges and have been shown to be stable and persistent enough for tomographic inversion. The coherence properties and fluctuation statistics of focused arrivals are presented and analyzed. Data from a 28-day propagation experiment are presented. The receptions of broadband pulse transmissions are compared for a 10-km propagation range for pulses with center frequencies from 100 to 3200 Hz. Simultaneous measurements of the sound speed field are used as model inputs to predict and compare spatial and temporal features of the focused arrivals.

2:00

3pUW4. Anomalous signal loss in the Yellow Sea, revisited: Coupling the acoustics with model-generated oceanographic realizations. Stanley A. Chin-Bing, David B. King, Alex C. Warn-Varnas, Robert A. Zingarelli (Naval Res. Lab., Stennis Space Center, MS 39529-5004, chinbing@nrlssc.navy.mil), and Kevin G. Lamb (Univ. of Waterloo, Waterloo, ON N2L 3G1, Canada)

In a seminal paper, Zhou *et al.* [J. Acoust. Soc. Am. **90**, 2042–2054 (1991)] introduced the concept of large anomalous signal loss due to a resonance effect caused by solitons (internal waves traveling along the thermocline). They assumed that solitons were present and produced a remarkable comparison between acoustic model predictions and their acoustic measurements taken over a four-year period. They did not take sufficient oceanographic measurements to confirm the existence of solitons. Numerous investigators have tried to duplicate their pioneering work in similar shallow-water environments with solitons present. Signal reduction, mode conversions, and resonance-like effects have been observed, but signal loss of the magnitude measured by Zhou *et al.* have not been observed. We have used a primitive equation soliton model and the tidal flow near the Shandong peninsula to generate soliton simulations that flow into the Yellow Sea near the region where Zhou *et al.* made their acoustic measurements. We are performing analysis similar to that of Zhou *et al.* on these soliton realizations to determine if large signal losses can occur. Results from this investigation will be presented and discussed. [Work supported by ONR/NRL and by a High Performance Computing DoD grant.]

2:15

3pUW5. Broadband acoustic variability due to internal solitary waves. Yongke Mu, Mohsen Badiey (College of Marine Studies, Univ. of Delaware, Newark, DE 19716), James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Stephen N. Wolf (Naval Res. Lab., Washington, DC 20375), and William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180)

A shallow water (70–90-meter-deep) broadband acoustic experiment with a source-receiver range of 15–18 km in an azimuthally dependent environment was designed as a part of the SWARM'95 field study. Temporal behavior of the water column was sampled every minute from two sources placed above and below the thermocline transmitting signals over a period of several hours while the water column was measured for the passage of internal waves. Spatial behavior was sampled in the vertical plane by hydrophones with spacing on the order of meters. Coherence of broadband acoustic waves for frequencies (20–300 Hz) is examined for the waveguide over different environmental conditions. The effect of environmental variability on coherence, in particular the sound-speed fluctuations in the water column due to the internal solitary waves, is noted as a function of acoustic frequency and azimuth. Analysis of the acoustic fluctuations over short time scales (10–15 minutes) may resolve the temporal decorrelation of the received signal due to internal waves. The vertical sampling of the received signal permits an analysis of arrival-angle fluctuations. Numerical simulations are performed, which provide support to the experimental observations. [Work supported by ONR.]

3pUW6. Ray dynamics in the AET experiment. Michael G. Brown (Rosenstiel School of Marine and Atmospheric Sci., Univ. of Miami, Miami, FL 33149, mbrown@rsmas.miami.edu), John A. Colosi (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Anatoly L. Virovlyansky, George M. Zaslavsky (Courant Inst. of Mathematical Sci., New York Univ., New York, NY 10012), Steven Tomsovic, and Michael A. Wolfson (Washington State Univ., Pullman, WA 99164)

A ray-based wave field description is employed in the analysis of measurements made during the November 1994 Acoustic Engineering Test (the AET experiment). In this experiment phase-coded pulse-like signals with 75-Hz center frequency and 37.5-Hz bandwidth were transmitted near the sound channel axis in the eastern North Pacific Ocean. The resulting acoustic signals were recorded on a moored vertical receiving array at a range of 3252 km. In our analysis both mesoscale and internal-wave-induced sound speed perturbations are taken into account. Much of this analysis exploits results that relate to the subject of ray chaos; these results follow from the Hamiltonian structure of the ray equations. It is argued that all of the important features of the measured AET wave fields are consistent with a ray-based wave field description in which ray trajectories are predominantly chaotic. [Work supported by ONR.]

3pUW7. Time variability of acoustic signals in a benign shallow-water area. Peter L. Nielsen, Martin Siderius, and Finn B. Jensen (SACLANT Undersea Res. Ctr., Viale San Bartolomeo 400, 19138 La Spezia, Italy)

SACLANTCEN conducted the ADVENT'99 experiment in the Strait of Sicily, Mediterranean, to assess the time variability of broadband acoustic signals in a benign shallow-water region. Broadband acoustic signals

(200–3800 Hz) were transmitted for up to 18 h over fixed paths of 2, 5, and 10 km. Dense sampling of the environment was performed including a sound-speed section every hour along the acoustic tracks. Band-averaged transmission loss is stable with time, which agrees well with modeling results. The Bartlett processor was applied to correlate the acoustic data. The correlation time varies from several hours at low frequency and short range to a few minutes at high frequency and longer ranges. Range- and time-dependent propagation modeling of the acoustic signals (200–800 Hz) shows behavior similar to the data. Particularly, the correlation time decreases abruptly from several hours to less than 1 hour at a frequency around 700 Hz for the 10-km data. This effect of the time-varying waveguide on the acoustic signals is correctly predicted by the propagation model. Severe problems using the acoustic signals coherently persist above a few hundred Hz and at ranges beyond a few kilometers, although the experiment was conducted under favorable conditions. The experimental data and the simulations illustrate the complexity of sound propagation in shallow water.

Plenary Session and Awards Ceremony

Katherine S. Harris, President
Acoustical Society of America

Presentation of Certificates to New Fellows

R. Dean Ayers	Dennis F. Jones
Soren Bech	Richard H. Love
Peter D'Antonio	Charles Thompson
Laurent J. D. Demany	Ji-Qing Wang
Mark A. Holden	Oleg A. Godin
Donald E. Hall	Hari S. Paul
Gordon R. Hamilton	Hans C. Strifors
W. Jack Hughes	

Presentation of Awards

Science Writing Award in Acoustics for Journalists to Kathryn Brown

Science Writing Award in Acoustics for Journalists to Roland Pease and Radek Boschetty

Science Writing Award for Professionals in Acoustics to William M. Hartmann

Distinguished Service Citation to John V. Bouyoucos

Distinguished Service Citation to F. Avril Brenig

Pioneers of Underwater Acoustics Medal to Darrell R. Jackson

Silver Medal in Physical Acoustics to Gregory W. Swift

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