Session 4aAA

Architectural Acoustics and Musical Acoustics: Recording Studio Acoustics

Jeff D. Szymanski, Cochair
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Alexander U. Case, Cochair
Fermata Audio and Acoustics, P.O. Box 1161, Portsmouth, New Hampshire 03802

Invited Papers

8:30

4aAA1. A review of the pertinent measurements and equations for small room acoustics. Douglas Jones (Dept. of Audio Arts & Acoust., Columbia College Chicago, 600 S. Michigan, Chicago, IL 60605)

Confusion about the appropriate measurements and calculations for use in small room acoustics persists. This paper presents an overview of the most important calculations to predict performance during the design phase and the most useful measurements to verify or troubleshoot performance after the facility is constructed. Special attention will be paid to the importance and relevance of reverberation in small rooms.

8:50

4aAA2. A consideration on physical tuning for acoustical coloration in recording studio. Yasushi Shimizu (Prog. in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY 12180-3590, shimiy@rpi.edu)

Coloration due to particular architectural shapes and dimension or less surface absorption has been mentioned as an acoustical defect in recording studio. Generally interference among early reflected sounds arriving within 10 ms in delay after the direct sound produces coloration by comb filter effect over mid- and high-frequency sounds. In addition, less absorbed room resonance modes also have been well known as a major component for coloration in low-frequency sounds. Small size in dimension with recording studio, however, creates difficulty in characterization associated with wave acoustics behavior, that make acoustical optimization more difficult than that of concert hall acoustics. There still remains difficulty in evaluating amount of coloration as well as predicting its acoustical characteristics in acoustical modeling and in other words acoustical tuning technique during construction is regarded as important to optimize acoustics appropriately to the function of recording studio. This paper presents a example of coloration by comb filtering effect and less damped room modes in typical post-processing recording studio. And acoustical design and measurement technique will be presented for adjusting timbre due to coloration based on psycho-acoustical performance with binaural hearing and room resonance control with line array resonator adjusted to the particular room modes considered.

9:10

4aAA3. If you can't take the room out of your mix, you can't take your mix out of the room! Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

The key issue in any recording studio is transferability—the ability of a mix to transfer to other listening environments outside the studio. For a mix to faithfully transfer to a wide range of acoustical environments, it must be created in a room with minimal acoustic distortion. The music industry is very aware of electronic distortion; however, the audible effects of acoustic distortion are only now being fully appreciated. The four forms of acoustic distortion are modal coupling, speaker boundary interference response, comb filtering and poor diffusion or a sparse spatial and temporal reflection density. These phenomena will be explained and methods to minimize them will be suggested.

9:30

4aAA4. NFL Films audio, video, and film production facilities. Russ Berger, Richard C. Schrag, and Jason J. Ridings (Russ Berger Design Group, 4006 Belt Line Rd., Ste. 160, Addison, TX 75001, russ@rbdg.com)

The new NFL Films 200,000 sq. ft. headquarters is home for the critically acclaimed film production that preserves the NFL's visual legacy week-to-week during the football season, and is also the technical plant that processes and archives football footage from the earliest recorded media to the current network broadcasts. No other company in the country shoots more film than NFL Films, and the inclusion of cutting-edge video and audio formats demands that their technical spaces continually integrate the latest in the ever-changing world of technology. This facility houses a staggering array of acoustically sensitive spaces where music and sound are equal partners with the visual medium. Over 90,000 sq. ft. of sound critical technical space is comprised of an array of sound stages, music scoring stages, audio control rooms, music writing rooms, recording studios, mixing theaters, video production control rooms, editing suites, and a screening theater. Every production control space in the building is designed to monitor and produce multi channel surround sound audio. An overview of the architectural and acoustical design challenges encountered for each sophisticated listening, recording, viewing, editing, and sound critical environment will be discussed.

9:50–10:10 Break

10:10

4aAA5. NFL Films music scoring stage and control room space. Russ Berger, Richard C. Schrag, and Jason J. Ridings (Russ Berger Design Group, 4006 Belt Line Rd., Ste. 160, Addison, TX 75001, russ@rbdg.com)

NFL Films' new 200,000 sq. ft. corporate headquarters is home to an orchestral scoring stage used to record custom music scores to support and enhance their video productions. Part of the 90,000 sq. ft. of sound critical technical space, the music scoring stage and its associated control room are at the heart of the audio facilities. Driving the design were the owner's mandate for natural light, wood textures, and an acoustical environment that would support small rhythm sections, soloists, and a full orchestra. Being an industry leader in cutting-edge video and audio formats, the NFLF required that the technical spaces allow the latest in technology to be continually integrated into the infrastructure. Never was it more important for a project to hold true to the adage of "designing from the inside out." Each audio and video space within the facility had to stand on its own with regard to user functionality, acoustical accuracy, sound isolation, noise control, and monitor presentation. A detailed look at the architectural and acoustical design challenges encountered and the solutions developed for the performance studio and the associated control room space will be discussed.

10:30

4aAA6. Home studio acoustic treatments on a budget. Gavin A. Haverstick (Auralex Acoust., Inc., 8851 Hague Rd., Indianapolis, IN 46256, gavinh@auralex.com)

Digital technology in the recording industry has evolved and expanded, allowing it to be widely available to the public at a significantly lower cost than in previous years. Due to this fact, numerous home studios are either being built inside or converted from bedrooms, dens, and basements. Hobbyists and part-time musicians that typically do not have the advantage of a large recording budget operate the majority of these home studios. Along with digital equipment, acoustic treatment has become more affordable over the years giving many musicians the ability to write, record, and produce an entire album in the comfort of their own home without having to sacrifice acoustical quality along the way. Three separate case studies were conducted on rooms with various sizes, applications, and budgets. Acoustical treatment such as absorption, diffusion, and bass trapping were implemented to reduce the effects of issues such as flutter echo, excessive reverberation, and bass build-up among others. Reactions and subjective comments from each individual studio owner were gathered and assessed to determine how effective home studios can be on a personal and professional level if accurately treated acoustically.

10:50

4aAA7. Studio with a view. Anthony K. Hoover (Cavanaugh Tocci Assoc., 327F Boston Post Rd., Sudbury, MA 01776)

Berklee College of Music (in Boston) needed a new studio in which to teach stereo mixing and critical listening. A small synthesis lab (adjacent to the main lobby, directly over the cafeteria kitchen, penetrated by exhaust ducts, and next to a bathroom) was chosen for renovation. The primary requirements were for maximum visibility to assure hopeful future engineers a full view of all the cool gear, and comfortable seating for fifteen students. The challenges, to be discussed, included isolation with a view, quiet HVAC, and great sound, in a space that was acoustically too small and in the wrong place. The best verification of success is its popularity, which has prevented the author from booking time for listening or testing.

11:10

4aAA8. Design of a critical listening classroom and studio. Bob Alach, Lou Clark (Alactronics, 192 Worcester St., Wellesley, MA 02481, bob@alactronics.com), William Carman (Univ. of Massachusetts—Lowell, Lowell, MA 01854), and Alex Case (Fermata Audio + Acoust., Portsmouth, NH 03802)

The Sound Recording Technology program at the University of Massachusetts—Lowell required design of a space that could serve three functions: a critical listening classroom for 30 students, a live room for the department's premier 24 track recording studio, and a 5.1 surround sound mix room. Accomplishing all of the above required a re-evaluation of room shape and treatments and aggressive use of variable acoustics. In addition, all the usual culprits of HVAC noise and vibration, limited space, a constrained budget, and a difficult to renovate reinforced concrete building added to the challenge. This paper reviews the key design and renovation challenges associated with completing this room, opened for use by students and faculty in the Spring 2003 semester.

Contributed Paper

11:30

4aAA9. Acoustics of a broadcast center. Sergio Beristain (E.S.I.M.E., IPN, IMA, P.O. Box 12-1022, Narvarte, 03001, Mexico D. F., Mexico, sberista@hotmail.com)

A broadcast system in Mexico City had to change facilities in order to concentrate in a single site all related broadcast stations and production studios in order to facilitate its normal operation. This led to a design which included the acoustic noise isolation and the interior acoustics of every studio and control room, together with the audio interconection, the electricity layout, the air conditioning system, the office building, etc. This paper presents the acoustics profile of the center, including final results of the construction as they were measured on completion of the installation. The complex has seven AM and FM broadcast stations, plus seven production studios for news, commercials and radio-novels plus an audio master control room, and everything was completed within four months.

11:45–12:00

Panel Discussion

Session 4aAB

Animal Bioacoustics: Nature's Orchestra—Acoustics of Singing and Calling Animals I

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

Chair's Introduction—8:00

Invited Papers

8:05

4aAB1. The chorus environment and the shape of communication systems in frogs. Vince Marshall (Div. of Biogical Sci., Univ. of Missouri–Columbia, 213 Tucker Hall, Columbia, MO 65211, vtm1da@mizzou.edu)

Many species of frogs breed in dense and structurally complex aggregations of calling males termed choruses. Females entering a chorus are faced with the tasks of recognizing and locating mates on the basis of their advertisement calls. The chorus environment poses particular challenges for communication as signalers and receivers will face high levels of background noise and interference between signals. For females, such conditions may decrease the efficiency of communication, with the consequences of increasing the time required to find a mate or errors in mate choice. For males, it will give rise to intense competition for the attention of females. Additionally, the chorus environment for a species is not static, and will vary over both spatial and temporal scales. This complex and dynamic environment has shaped the signals and signaling behaviors of frogs in sometimes surprising ways. In this talk, some of the implications of the chorus environment for both receivers and signalers is discussed. In particular, examples from North American hylid frogs are drawn upon and research on the role of signal timing in influencing the responses of females and plasticity in aggressive behavior between neighbors in choruses are discussed.

8:25

4aAB2. Patterns of fish sound production. David A. Mann (Univ. of South Florida, College of Marine Sci., 140 7th Ave. S., St. Petersburg, FL 33701, dmann@marine.usf.edu)

While vocalization and chorusing behavior has been intensively studied in frogs and birds, little has been done with fishes. This paper presents patterns of sound production in damselfish, toadfish, and spotted seatrout on seasonal, daily, and subdaily time scales. Chorus behavior ranges from highly coordinated behavior between neighboring toadfish to uncoordinated behavior in spotted seatrout. Differences in coordination of sound production (i.e., the degree of overlap in calls) can be related to differences in territoriality and modes of reproduction. Toadfish and damselfish are territorial fishes in which males guard benthic eggs laid in nests. Sciaenids (croakers and drums) spawn planktonic eggs, and form temporary aggregations of calling males.

8:45

4aAB3. Acoustic communication in *Panthera tigris*: A study of tiger vocalization and auditory receptivity. Edward J. Walsh (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131), Lily M. Wang (Univ. of Nebraska, Omaha, NE 68182), Douglas L. Armstrong, Thomas Curro, Lee G. Simmons (Henry Doorly Zoo, Omaha, NE 68108), and JoAnn McGee (Boys Town Natl. Res. Hospital, Omaha, NE 68131)

Acoustic communication represents a primary mode of interaction within the sub-species of *Panthera tigris* and it is commonly known that their vocal repertoire consists of a relatively wide range of utterances that include roars, growls, grunts, hisses and chuffling, vocalizations that are in some cases produced with extraordinary power. *P. tigris* vocalizations are known to contain significant amounts of acoustic energy over a wide spectral range, with peak output occurring in a low frequency bandwidth in the case of roars. von Muggenthaler (2000) has also shown that roars and other vocal productions uttered by *P. tigris* contain energy in the infrasonic range. While it is reasonable to assume that low and infrasonic acoustic cues are used as communication signals among conspecifics in the wild, it is clearly necessary to demonstrate that members of the *P. tigris* sub-species are responsive to low and infrasonic acoustic signals. The auditory brainstem response has proven to be an effective tool in the characterization of auditory performance among tigers and the results of an ongoing study of both the acoustical properties of *P. tigris* vocalizations and their auditory receptivity support the supposition that tigers are not only responsive to low frequency stimulation, but exquisitely so.

9:05

4aAB4. Humpback whale song: A new review. Adam S. Frankel (Marine Acoust., Inc., 706 Giddings Ave., Ste. 2A, Annapolis, MD 20764)

The humpback whale song has been described and investigated since the early 1970s. Much has been learned about the humpback whale social structure, but the understanding of the song and its function remains elusive. The hierarchical nature of the song structure was described early on: Songs can be sung for a long period, apparently by males, and primarily during the mating season. However, singers also become physically competitive, suggesting alternative mating strategies. There are a number of unique structural features of song. Its structure evolves over time and combination. The nature of song evolution strongly implies cultural transmission. Song

structure appears to be shared within an entire population, even though there appears to be little interchange of individuals between sub populations. Despite over thirty years of inquiry there are still numerous unanswered questions: Why is the song structure so complex? Is song a sexual advertisement, an acoustic space mediation mechanism, or both? How do females choose mates, or do they? What drives song evolution, and why is there so much variation in the rate of change? Are there nonreproductive functions of song? What prompts a male to begin or end singing? Our current understanding and the outstanding questions yet to be answered will be reviewed.

9:25

4aAB5. Signals and speciation in plant-feeding insects. Reginald Cocroft, Rafael Rodriguez (Biological Sci., Univ. of Missouri, Columbia, MO 65211), and Randy Hunt (Indiana Univ.–SE, New Albany, IN 47150)

Communication can play a role in speciation when differences in signaling systems reduce gene flow between diverging lineages. However, the observation that closely related species have different signals leaves open the question of cause and effect, because divergence in signals can also occur after speciation is completed. The relationship of signals and divergence is being investigated in the Enchenopa binotata species complex of treehoppers, a group of plant-feeding insects that communicate using substrate-borne vibrational signals. Speciation in the *E. binotata* complex is hypothesized to have resulted from repeated shifts to novel host plants. Sources of selection on the signals of these treehoppers can be studied not only in fully differentiated species, but also in experimental populations in the early stages of divergence. Some unique features of vibrational communication through plant stems will be discussed, as well as how shifts to novel host plants may influence signaling systems, and what role(s) communication may have played in the process of evolutionary divergence in this group.

9:45

4aAB6. Listening to Nature's orchestra with peculiar ears. David D. Yager (Dept. of Psych., Univ. of Maryland, College Park, MD 20742)

Insects use hearing for the crucial tasks of communicating with conspecifics and avoiding predators. Although all are based on the same acoustic principles, the diversity of insect ears is staggering and instructive. For instance, a South African grasshopper demonstrates that hearing conspecific calls is possible over distances 1 km with ears that do not have tympana. Actually, these creatures have six pairs of ears that play different roles in behavior. In numerical contrast, praying mantises have just a single ear in the ventral midline. The ear is very effective at detecting ultrasonic bat cries. However, the bioacoustics of sound transduction by two tympana facing each other in a deep, narrow slit is a puzzle. Tachinid flies demonstrate that directional hearing at 5 kHz is possible with a pair of ears fused together to give a total size of 1 mm. The ears are under the fly's chin. Hawk moths have their ears built into their mouthparts and the tympanum is more like a hollow ball than the usual membrane. As an apt last example, cicada ears are actually part of the orchestra: their tympana function both in sound reception and sound production.

10:05–10:20 Break

10:20

4aAB7. Perception of species-specific vocalizations by birds. Robert Dooling (Dept. of Psych., Univ. of Maryland, College Park, MD 20742, dooling@psyc.umd.edu)

The complexity of bird vocalizations has fascinated us throughout the ages and there has long been the suspicion that birds are capable of producing, perceiving, and learning features of their songs that are beyond the capabilities of human hearing. Other investigators point to similarities between song production and perception in birds and special perceptual and attentional processes that are involved in the perception of speech by humans. Evidence will be reviewed that birds are born with early perceptual predispositions for learning species-specific vocalizations and demonstrate unusual efficiency in perceiving species-specific vocalizations throughout adulthood. In spite of these similarities with humans, birds have other auditory perceptual biases that are clearly different from humans as in the perception of tonal patterns. Birds, like humans, show a number of phenomena that aid perception under adverse conditions such as the binaural release from masking and the precedence effect and the detection of vocalizations in noise appears to follow from general principles of masking which has implications for studies of the effect of noise on birds in nature. [Work supported by NIH.]

Contributed Papers

10:40

4aAB8. The vocal monotony of monogamy. Jeanette Thomas (Lab. of Sensory Biol., Western Illinois Univ. Quad Cities, 3561 60th St., Moline, IL 61265)

There are four phocids in waters around Antarctica: Weddell, leopard, crabeater, and Ross seals. These four species provide a unique opportunity to examine underwater vocal behavior in species sharing the same ecosystem. Some species live in pack ice, others in factice, but all are restricted to the Antarctic or sub-Antarctic islands. All breed and produce vocalizations under water. Social systems range from polygyny in large breeding colonies, to serial monogamy, to solitary species. The type of mating system influences the number of underwater vocalizations in the repertoire, with monogamous seals producing only a single call, polygynous species producing up to 35 calls, and solitary species an intermediate number of about 10 calls. Breeding occurs during the austral spring and each species carves-out an acoustic niche for communicating, with species using different frequency ranges, temporal patterns, and amplitude changes to convey their species-specific calls and presumably reduce acoustic competition. Some species exhibit geographic variations in their vocalizations around the continent, which may reflect discrete breeding populations. Some seals become silent during a vulnerable time of predation by killer whales, perhaps to avoid detection. Overall, vocalizations of these seals exhibit adaptive characteristics that reflect the co-evolution among species in the same ecosystem.

10:55

4aAB9. Songlike vocalizations from a Sumatran rhinoceros calf (*Dicerorhinus sumatrensis*). Elizabeth von Muggenthaler (Fauna Commun.s Res. Inst., P.O. Box 1126, Hillsborough, NC 27278) and Paul Reinhart (Cincinnati Zoo, Cincinnati, OH)

Within the last ten years the Sumatran rhino population has dropped 50%, and only 200–300 individuals exist, with five in captivity. Their native habitat is dense tropical forest and they are solitary, therefore much of their behavior remains unknown. Sumatrans are the smallest living rhino, standing 0.9–1.5 m tall, and are covered in coarse, reddish-brown hair. The first Sumatran rhinoceros born in captivity in 112 years, and the first calf ever recorded, is 17 months old and weighs 448 kg. At the Cincinnati Zoo this male calf was recorded from 1–3 m, using two Statham radio microphones, and one TCD-D8 Sony DAT recorder (9 Hz–22 kHz). Analysis, including power spectrums, spectrographic functions, and cross correlations were performed using National Instrument's Polynesia. Preliminary analysis indicates that the calf's vocalizations are similar in structure to adult Sumatran vocalizations, although there are some distinctions. ''Eeps'' and ''whales'' that are found in adult repertoires are produced by the calf. However, signals from the calf are higher in frequency, and the calf does not vocalize as consistently as the adults. The calf has yet to produce a ''whistle blow,'' which is an adult vocalization that has a strong infrasonic component.

11:10

4aAB10. Comparison of St. Lawrence blue whale vocalizations with field observations. Catherine Berchok (Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804, cberchok@yahoo.com), David Bradley, Thomas Gabrielson (Penn State Univ., State College, PA 16804), and Richard Sears (MICS, Inc., St., Lambert, QC J4P 1T3, Canada)

During four field seasons from 1998–2001, vocalizations were recorded in the presence of St. Lawrence blue whales using a single omnidirectional hydrophone. Both long duration infrasonic calls (~18 Hz, 5–20 s) as well as short duration higher frequency calls (85–25 Hz, ~2 s) were detected and compared with field observations. Two trends were noted. First, the long infrasonic call series were concentrated primarily in the deep (300 m) channel. These call series appear to compare well with blue whale vocalizations recorded by others in the deep open ocean. Second, the shorter audible calls were more evenly distributed over bathymetry and seem to be a form of short distance communication with at least one case occurring during an agonistic interaction. A comparison of these calls with biological parameters such as density of whales in the area, percentages of paired versus single whales, and numbers of males versus females will also be discussed. [Project supported by ARL/PSU, NSF, and the American Museum of Natural History.]

11:25

4aAB11. The acoustic field of singing humpback whales in the vertical plane. Whitlow W. L. Au (Marine Mammal Res. Prog., Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734), Adam A. Pack (Kewalo Basin Marine Mammal Lab., Honolulu, HI 96814), Marc O. Lammers (Hawaii Inst. of Marine Biol.), Louis Herman (Kewalo Basin Marine Mammal Lab.), Kimberly Andrews (Hawaii Inst. of Marine Biol.), and Mark Deakos (Kewalo Basin Marine Mammal Lab.)

A vertical array of five hydrophones was used to measure the acoustic field of singing humpback whales. Once a singer was located, two swimmers with snorkel gear were deployed to determine the orientation of the whale and to position the boat so that the array could be deployed in front of the whale at a minimum standoff distance of 10 m. The spacing of the hydrophones was 7 m with the deepest hydrophone deployed at depth of 35 m. An 8-channel TASCAM recorder having a bandwidth of 24 kHz was used to record the hydrophone signals. The location of the singer was determined by computing the time of arrival differences between the hydrophone signals. The maximum source level varied between individual units in a song, with values between 180 and 190 dB. The acoustic field determined by considering the relative intensity of higher frequency harmonics in the signals indicate that the sounds are projected in the horizontal direction with the singer's head canted downward 45 to 60°. High-frequency harmonics extended beyond 24 kHz, suggesting that humpback whales may have an upper frequency limit of hearing as high as 24 kHz.

THURSDAY MORNING, 1 MAY 2003 ROOMS 110/111, 9:00 A.M. TO 12:00 NOON

Session 4aAO

Acoustical Oceanography: General Topics in Acoustical Oceanography

Mohsen Badiey, Cochair
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Kathleen E. Wage, Cochair
Department of ECE, George Mason University, 4400 University Drive, Fairfax, Virginia 22030

Contributed Papers

9:00

4aAO1. Measuring ambient ocean bubble fields using a multibeam sonar. Steve Adelman, David L. Bradley, R. Lee Culver, and Thomas C. Weber (Appl. Res. Lab., Penn State Univ., P.O. Box 30, N. Atherton St., State College, PA 16804-0030, sga121@psu.edu)

For two weeks in July 2002 off the coast of San Diego, a 240 kHz SeaBat 8101 Multibeam Echosounder System was used to measure the backscatter from ambient bubble fields in the near surface layer. The sonar system was mounted approximately 12.5 meters below the surface on the hydraulic orientation unit of the research platform FLIP and was oriented so that its fan of 101 1.5 degree beams were looking at the surface at an angle of 45 degrees from the vertical. Data collection was part of a joint experiment involving the Applied Research Laboratory (ARL)/The Pennsylvania State University (PSU) and the Marine Physical Laboratory (MPL)/Scripps Institute of Oceanography (SIO). Although sea conditions were benign for most of the test period, wind wave activity near the end provided a number of opportunities to observe near surface bubble entrainment. The sonar system proved to be an effective tool for the observation of the lifetime, spatial structure and dimensions of the ambient bubble fields. [Work supported by ONR under Award No. N00014-02-1-0156.]

9:15

4aAO2. Some analytic solutions for the dispersion and attenuation of sound in bubbly seawater. Ralph R. Goodman and Jerald W. Caruthers (The Univ. of Southern Mississippi, Stennis Space Center, MS 39529)

The effect that air bubbles have on sound propagation is well known, and in the past three decades, due to advances in acoustic measurement technologies, has been the subject of several studies that investigate the bubble sizes and distributions due to breaking waves and ship wakes. The distribution of bubbles in size and total void fraction, and in time and space, plays a critical role in the determination of dispersion, refraction and attenuation. Bubble size distributions have been measured under different environments and by several techniques and have been represented usually with a power law or an exponential dependence. The authors have investigated a few distributions that are amenable to exact analytical solutions. These are power law distributions that have been chosen to give insight to realistic cases without significantly modifying the physics. The solutions to the integrals for dispersion and attenuation will be presented. They will be used to show the changes in the magnitudes of sound speed and attenuation under varying conditions.

9:30

4aAO3. On the underwater sound generated by the counterpropagating sea-surface wave. Konstantin A. Naugolnykh (Univ. of Colorado/Zeltech, 325 Broadway, Boulder, CO 80305) and Samuil A. Rybak (N. Andreev Acoust. Inst., Moscow 117036, Russia)

Two counterpropagating gravity-capillary waves of the same frequency, forming a standing wave on the sea surface, give rise to a sound wave traveling away from the surface [M. S. Longuet-Higgins (1950); L. M. Brekhovskikh (1966)]. The effect can be described in terms of three-wave nonlinear interaction [V. V. Goncharov, K. A. Naugolnykh, and S. A. Rybak (1977)]. The frequency and the wave number of the radiated wave are determined by the synchronism conditions. The surface waves propagating in the opposite directions can be generated by the action of the wind on the remaining swell of a cyclone or due to nonlinear interaction of the gravity-capillary waves. The progressive gravity-capillary waves superimposed on the flow produced by the finite-amplitude dominant gravity wave can be blocked at the points where its group velocity balance the convection by the larger-scale flow generating the reflected wave. As a result in the field of progressive waves produced by wind the counterpropagating wave appears generating the sound. [Work partially supported by Collaborative Linkage Grant No. 977 890.]

9:45

4aAO4. Rain parameter estimation using impact generated low-frequency acoustic signals. T. K. Mani, P. R. Saseendran Pillai, James Kurian, and Supriya M. Hariharan (Cochin Univ. of Sci. and Technol., Cochin 682022, India)

The role of rain generated acoustic signals in the frequency range 500 Hz–100 kHz to estimate the rain parameters is well known [Medwin *et al.*, J. Acoust. Soc. Am. **92**, 1613–1623 (1992)]. A study of utilizing the low-frequency component below 500 Hz, present in the rain generated acoustic noise, for estimation of rain parameters is performed in this paper. Raindrops falling from various heights are simulated using a drop generator. Gravity waves are produced in the water surface due to the drop impact and this causes a sinusoidal low-frequency damped pressure wave in water. This low-frequency signal is captured using a sensor assembly and analyzed. It is observed that this component is fairly constant in its frequency while amplitude is found to vary in accordance with the drop size as well as the velocity. Drop-size and its kinetic energy are determined and compared with those obtained from theoretical computations and by direct measurements. The results show that this new technique of analysis of rain generated acoustic signals yields rain parameters with good degree of accuracy. This method has the advantage of savings in computation time and simplicity of design.

10:00–10:15 Break

10:15

4aAO5. Sound fluctuations due to horizontal refraction in the SWARM-95 experiment. Mohsen Badiey (College of Marine Studies, Univ. of Delaware, Newark, DE 19716), Boris Katsnelson, Serguey Pereselkov (Voronezh State Univ., Voronezh 394006, Russia), James Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), and William Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180)

Fluctuations of the sound field measured in the SWARM-95 experiment are interpreted as manifestations of three-dimensional effects in shallow-water sound propagation. It is shown that features of the received sound signal, along an acoustic track placed at 5° to the crests of propagating internal soliton trains, can be explained by significant horizontal refraction of the sound waves. This refraction leads to a remarkable redistribution of the energy in the horizontal plane as has been recently remarked by other investigators. These features are specifically: (1) a high level of fluctuations of the sound energy (up to 6 dB) which are synchronous in time and have same amplitude over depth of waveguide and (2) anomalies in the arrival times and amplitudes of separate waveguide modes. A theoretical analysis based on the theory of vertical modes and horizontal rays is presented, along with computer modeling of the experiment. Analytical estimations of the observed effects as well as results from numerical modeling are in a good agreement with experimental observations and are consistent with simulations. [Work supported by ONR and US CRDF Award No. VZ-010-00.]

10:30

4aAO6. Frequency-dependent anamolies of sound propagation in the SWARM-95 experiment. Boris Katsnelson, Serguey Pereselkov (Voronezh State Univ., Universitetskaya Sq. 1, Voronezh 394006, Russia), Mohsen Badiey (College of Marine Studies, Univ. of Delaware, Newark, DE 19716), and James Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Frequency-dependent acoustic propagation is a subject of great interest in shallow-water regions. The phenomenon known as resonance absorption occurs when broadband sound signals propagate across the crests of internal solitary waves (ISWs) if the sound field satisfies the condition of space synchronization. In this paper we show that different anomalies in frequency dependence can take place when sound signals propagate along the wave fronts of ISWs. These anomalies result from the anisotropic properties of the ISWs that can produce frequency-dependent horizontal refraction of acoustic waves in the ocean. Sound propagation of broadband pulses in the SWARM-95 experiment is analyzed from this point of view. The nature of such behavior is connected with the selective (over modes and frequencies) character of thermocline perturbations caused by ISWs. [Work supported by ONR and US CRDF Award No. VZ-010-00.]

10:45

4aAO7. Internal wave observations and transmission losses in the Santa Barbara Channel. Jeff Dunne (Widener Univ., One Univ. Pl., Chester, PA 19013, dunne@pop1.science.widener.edu), Orest Diachok, and Stephen Wales (Naval Res. Lab., Washington, DC 20375)

Measurements from three vertical line arrays of temperature probes, deployed in a triangular configuration in the Santa Barbara Channel in August 2002, permitted determination of speed and direction of tidally generated internal solitary wave events (ISWEs). Two arrays were deployed 2 km apart along a 60-m isobath, with the third normal to the isobath. Sound propagation measurements were also conducted along the isobath. Highly correlated ISWEs were evident on all three arrays every 12 h. Measurements of time delays suggest that these waves emanated from a single site between two of the Channel Islands. Because the ISWE propagation direction was nearly normal to the acoustic propagation direction, acoustic mode coupling due to ISWEs is expected to be minimal. Time-series temperature measurements reveal that average near-surface sound speeds were systematically different when ISWEs were present or absent, and changed abruptly during transitions. Numerical simulations with a normal mode propagation code (C-SNAP) indicate that frequency-dependent differences in transmission loss on near-bottom hydrophones may be as large as 7 dB in narrow frequency bands (5%–10%) between 0.5 and 2 kHz when an ISWE is present versus when it is absent. [Work supported by ONR.]

11:00

4aAO8. Modeling the acoustic receptions at the NPAL array from the Kauai source. Michael D. Vera, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0225), and the NPAL Group[a)] (SIO-UCSD, APL-UW, WHOI)

Acoustic transmissions from a 75-Hz source near Kauai to a vertical line array near California were recorded as part of the North Pacific Acoustic Laboratory (NPAL) experiment. Extensive environmental measurements were also performed as part of the experiment and were intended to ensure correspondence between numerical simulations and the data. Despite the availability of this information, the process of identifying the recorded arrivals with predictions has not been a simple one. Since the source is near the seafloor at about 800 m depth, and the depth at the receiver is approximately 1800 m, acoustic interaction with the bathymetry has been explored as a possible complication. Ray simulations that allow for specular reflection from the bottom indicate that fully-refracted and bottom-interacting paths can reach the receiver range (about 3900 km) at similar travel times. The simultaneous presence of both kinds of acoustic energy could contribute to the identification difficulties. A series of parabolic-equation simulations have been performed for different geoacoustic parameters in an attempt to correspond more closely to the data. The sensitivity of the predictions to the method used to extract and interpolate the sound speeds has also been investigated. [Work supported by ONR.] [a)] J. A. Colosi, B. D. Cornuelle, B. D. Dushaw, M. A. Dzieciuch, B. M. Howe, J. A. Mercer, R. C. Spindel, and P. F. Worcester.

11:15

4aAO9. Whispering-gallery-mode trapping of sound in shallow water between an up-slope region and internal wave solitons. Allan D. Pierce and James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Double-trapping of sound occurs in shallow water when the ocean has a sloping bottom and when a packet of high amplitude internal wave solitons is propagating either up-slope or down-slope. The phase velocity of an adiabatic mode increases in the up-slope direction and also as one approaches the front of the soliton packet, so the mode's horizontal rays can be trapped for propagation that is nominally cross-slope. Because the variation of the horizontal phase velocity is considerably slower in the up-slope direction than it is in the vicinity of the soliton front, an analogy exists with the whispering-gallery effect (found near concave surfaces in architectural spaces). The equivalent radius of curvature of the ''reflecting surface'' is found in accord with the earth-flattening approximation to equal the phase velocity divided by its derivative with respect to the up-slope distance. Quantitative substantiation is given for models of sound and solitons in a realistic ocean (two constant sound speed layers separated by a thermocline). Double-trapped propagation has some attenuation because of energy leakage out through the width of the soliton packet.

11:30

4aAO10. Effects of internal solitary waves on the invariance of acoustic intensity striation patterns. Altan Turgut, Marshall Orr (Naval Res. Lab., Acoust. Div., Washington, DC 20375), Daniel Rouseff (Univ. of Washington, Seattle, WA 98105), James Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), and Ching Sang Chiu (Naval Postgrad. School, Monterey, CA 93943)

Effects of Internal Solitary Waves (ISWs) on the acoustic intensity striation patterns are studied using a unique data set collected in the South China Sea during the ONR Asian Seas International Acoustics Experiment (ASIAEx). The data set contains measured broadband (270–330 Hz) acoustic fields on a 32-element, ~400 m aperture horizontal array describing two regimes where the ISW packet being present and not present in a 19 km long propagation track. The geometry of the ISW packet was captured by a Synthetic Aperture Radar (SAR) image when it entered the propagation track at a 50 deg angle. It has been observed that ISWs introduce larger intensity fluctuations as well as frequency-shifts in the striation patterns when they are in the propagation track. However, the slope of the striations seems to be less affected by the presence of ISWs. Results from several broadband simulations are also presented to describe both acoustic intensity fluctuations and variations in the striation patterns in terms of acoustic mode coupling and mode refraction induced by ISWs. [Work supported by ONR.]

11:45

4aAO11. Comparison of broadband mode arrivals at ranges of 3515 km and 5171 km in the North Pacific. Kathleen E. Wage (George Mason Univ., MS1G5, 4400 University Dr., Fairfax, VA 22030) and The ATOC Group[a)]

The Acoustic Thermometry of Ocean Climate (ATOC) provided an opportunity to observe signals propagating in the low-order modes of the ocean waveguide. Understanding the fluctuations of these mode signals is an important prerequisite to using them for tomography or other applications. In previous work, we characterized the cross-mode coherence and temporal variability of the low-order mode arrivals at 3515 km range [Wage *et al.*, J. Acoust. Soc. Am. (in press)]. This study compares the mode arrivals for two different ranges : 3515 km and 5171 km, using data from the ATOC vertical line arrays at Hawaii and Kiritimati. We discuss the mode intensity and coherence statistics for each of the arrays and examine mean arrival time trends over the year-long deployment. Experimental results are compared to PE simulations of propagation through a realistic background environment perturbed by internal waves of varying strengths. The dependence of mode statistics on the path-dependent changes in the background sound speed and the parameters of the internal wave field is explored. [Work supported by an ONR Ocean Acoustics Young Faculty Award.] [a)] A. B. Baggeroer, T. G. Birdsall, C. Clark, J. A. Colosi, B. D. Cornuelle, D. Costa, B. D. Dushaw, M. A. Dzieciuch, A. M. G. Forbes, B. M. Howe, D. Menemenlis, J. A. Mercer, K. Metzger, W. H. Munk, R. C. Spindel, P. F. Worcester, and C. Wunsch.

Session 4aBB

Biomedical Ultrasound/Bioresponse to Vibration, Signal Processing in Acoustics and Physical Acoustics: High-Intensity Focused Ultrasound (HIFU) and Imaged-Guided Therapy I

Ronald A. Roy, Chair
Department of Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215

Invited Papers

8:20

4aBB1. Acoustic hemostasis. L. Crum, M. Andrew, M. Bailey, K. Beach (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Seattle, WA 98105), A. Brayman, F. Curra, P. Kaczkowski, S. Kargl, R. Martin, and S. Vaezy (Univ. of Washington, Seattle, WA 98105)

Over the past several years, the Center for Industrial and Medical Ultrasound (CIMU) at the Applied Physics Laboratory in the University of Washington has undertaken a broad research program in the general area of High Intensity Focused Ultrasound (HIFU). Our principal emphasis has been on the use of HIFU to induce hemostasis; in particular, CIMU has sought to develop a small, lightweight, portable device that would use ultrasound for both imaging and therapy. Such a technology is needed because nearly 50% of combat casualty mortality results from exsanguinations, or uncontrolled bleeding. A similar percentage occurs for civilian death due to trauma. In this general review, a presentation of the general problem will be given, as well as our recent approaches to the development of an image-guided, transcutaneous, acoustic hemostasis device. [Work supported in part by the USAMRMC, ONR and the NIH.]

8:45

4aBB2. HIFU treatment of liver cancer—Successes and failures. Gail ter Haar, Ian Rivens (Phys. Dept., Inst. of Cancer Res., Royal Marsden Hospital, Sutton, Surrey SM2 5PT, UK), James Kennedy (HAIFU Unit, Churchill Hospital, Oxford OX3, UK), and Feng Wu (Chongqing Univ. of Medical Sci., Chongqing 400016, PROC)

Clinical trials of the HIFU treatment of liver cancer have been underway in the UK at the Royal Marsden Hospital since December 1997, and at the Churchill Hospital since November 2002. Royal Marsden treatments are undertaken using a prototype device known as the Teleson, while those at the Churchill are performed using the machine produced by the Chongqing HAIFU company in China. Both sites have demonstrated the ability to ablate significant volumes of a tumor within the liver. Despite differences in ultrasound exposure delivery, these treatments have highlighted some of the problems associated with the clinical use of extracorporeal HIFU. These problems lie primarily in the areas of targeting, treatment optimization and monitoring of ablation. These problems will be discussed and potential solutions suggested. [Work funded by the UK Department of Health, the Institute of Cancer Research and UTL.]

9:10

4aBB3. Large ultrasound phased arrays for noninvasive focal treatments. Kullervo Hynynen, Gregory Clement, Nathan McDannold, Chris Connor, and Natalia Vykhodtseva (Dept. of Radiol./MRI, Brigham and Women's Hospital, 75 Francis St., Boston, MA 02115)

Ultrasound applicators have become more advanced with the use of large scale phased arrays. These arrays can control the focal spot location and shape and compensate for the ultrasound beam distortions introduced by tissue layers. It has been demonstrated that sharp focus can be formed even through the human skull. However, the exposure control becomes complex with these new arrays, and online monitoring is required to provide clinical safety. For the past few years, magnetic resonance imaging has been tested for imaging the temperature distributions induced by focused ultrasound treatments. Both phantom and animal experiments have shown that the proton resonant frequency is temperature dependent and that maps of the temperature change can be obtained within seconds with phase subtraction methods. Recent clinical treatments have verified the usefulness of large scale phased arrays and temperature mapping during ultrasound surgery.

9:35

4aBB4. Controlled ultrasonic tissue erosion. Charles Cain (Dept. of Biomed. Eng., Univ. of Michigan, Ann Arbor, MI 48109)

Controlled ultrasonic tissue erosion has many clinical applications, including the placement of very precise sharply defined perforations in biological interfaces and membranes with focused ultrasound. With carefully chosen acoustic parameters, tissue can be rapidly eroded away at a constant etching rate. The maximum erosion rate for minimal propagated energy is obtained by using very short high intensity pulses. The etching rate is higher with shorter pulse durations. For short pulses less than 10 cycles of the drive frequency, an optimum pulse repetition rate exists which maximizes the etching rate. Higher gas saturation in the surrounding medium reduces the etching rate and reduces the spatial sharpness of the holes produced. Most of the erosion appears to be produced in the first several cycles of the therapy pulse. For example, a series of short (about 3 cycles) focused pulses of 2100 W/cm^2 (Isppa) at 788 kHz can erode a very well defined 2 mm diameter hole in a 1 mm thick sample of fresh pork atrial posterior wall in about 1 min at the optimum pulse repetition rate (about 18 kHz). Controlled ultrasonic tissue erosion may provide an effective image guided noninvasive tool in treatment of neonatal patients with hypoplastic left heart syndrome. Without the mixing of oxygenated blood across perforations placed in the atrial septum, these infants do not survive.

10:00–10:15 Break

Contributed Papers

10:15

4aBB5. High intensity focused ultrasound (HIFU) for treatment of T1/T2 prostate cancer. N. Sanghvi (Focus Surgery, Inc., Indianapolis, IN 46226), T. Gardner, and M. Koch (Indiana Univ. School of Medicine, Indianapolis, IN 46226)

This FDA approved phase I/II clinical trial is to evaluate the safety and efficacy of the Sonablate device (Focus Surgery, Inc.) for the treatment of organ confined prostate cancer. 20 patients with biopsy proven prostate cancer, Gleason $\leqslant 7$ and PSA $\leqslant 10$ were treated under general anesthesia. Outcome data included serum PSA collected at day 3, 14, 30, 90, 180, PSA nadir (mean/median), and biopsy results at 6 months. Quality of life was assessed using the International Prostate Symptom Score, International Impotence and Erectile Function score, and the SF-36 health survey. The mean patient age is 62.0, Gleason score of 6.18, PSA of 5.2, and prostate size 26.0 g m. Mean PSA results were 5.62, 44, 20, 1.68, 0.87, and 0.44 ng/ml at screening, 48–72 hours, 14 days, 30 days, 90 days and 180 days, respectively. There was one patient (9%) with a positive TRUS biopsy at 6 months, which resulted in a retreatment. There were no rectal injuries. Average pre-treatment IPSS, IIEF, and SF-36 scores were 9.55, 16.1, and 103.5. At the 30 day follow-up, they were 18.3, 3, and 97.4, respectively. HIFU is a minimally invasive modality that achieves complete prostatic ablation and is efficacious in the treatment of low-stage prostate cancer.

10:30

4aBB6. High intensity focused ultrasound (HIFU) treatment of BPH: results of a multi-center phase III study. N. Sanghvi (Focus Surgery, Inc., Indianapolis, IN 46226), T. Gardner, M. Koch, R. Bihrle, R. Foster (Indiana Univ. School of Medicine, Indianapolis, IN 46226), M. Resnick, A. Seftel (Case-Western Reserve Univ., Cleveland, OH), I. Grunberger (Long Island College Hospital, Brooklyn, NY), C. Stiedle (North East Indiana Res. Hospital, Fort Wayne, IN), and J. Corchan (Presbyterian Hospital, Dallas, TX)

The five centers phase III trial was to show that HIFU can treat prostate tissue thermally for symptomatic relief of BPH and improve flow rates. At five sites, 68 BPH patients were treated with the Sonablate device (Focus Surgery, Inc. Indianapolis, IN). A urethral Foley catheter was inserted into the urethra to aid in positioning and was kept *in-situ* during the treatment. A cooling device was used to cool the rectal wall. The patients returned home within a few hours after the procedure. The Foley catheter was kept electively to avoid any incidence of acute urinary retention following the therapy. The catheter was removed after 4–5 days. The average treatment time was 38 minutes. The patients were treated without pain, blood loss or complications. At 90 days post treatment, average Qmax and AUA Symptom Scores improved from 8.7 ml/s to 12.66 ml/s (48%) and 23.06 to 11.62 (52%), respectively. Significant prostate tissue changes took place before and after the treatment. 80% of the patients had cavity formation at the site of treatment at the bladder neck and prostate. Nonsurgical HIFU therapy is safe and effective for providing symptomatic relief of BPH symptoms and the treatment can be performed as an outpatient procedure.

THURSDAY MORNING, 1 MAY 2003 ROOMS 108/109, 10:00 A.M. TO 1:00 P.M.

Session 4aED

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

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

Corinne M. Darvennes, Cochair
Tennessee Technological University, Box 5014, Cookeville, Tennessee 38505

Approximately 75 female students from rural high schools are invited to participate in this session. They will have the opportunity to sense the excitement of ''Doing Acoustics'' by participating in about 20 acoustics demonstrations. ASA members are free to play with the toys as well, as long as they do not interfere with student participation.

Session 4aPAa

Physical Acoustics: Nonlinear Acoustics and Resonators

Ray S. Wakeland, Chair
Graduate Program in Acoustics, Pennsylvania State University, Applied Science Building, Room 217, University Park, Pennsylvania 16804

Contributed Papers

8:00

4aPAa1. Parametric excitation of an acoustic mode. Bruce Denardo, Derek B. Smith, Larry P. Varnadore (Dept. of Phys., Naval Postgraduate School, Monterey, CA 93943), and Wayne E. Prather (Miltec Res. and Technol., Inc., Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 39677)

One means of driving an oscillatory mode of a system is to modulate a parameter upon which the natural frequency of the mode depends. This ''parametric excitation'' occurs most readily if the modulation frequency is twice the frequency of the mode. In addition, the modulation amplitude must exceed a threshold that depends upon the dissipation of the mode. The response amplitude then grows exponentially until it is saturated by a nonlinearity of the system, so large amplitudes are possible. Previous efforts to parametrically excite an acoustic mode will be briefly described, and the theory for a length-modulated tube will be discussed. An analysis of this case indicates that it is feasible to parametrically excite the fundamental mode. Results of an experiment that is currently in progress will be presented. The tube has ambient length 2.0 m and diameter 30 cm, and the length is modulated by low-frequency loudspeakers at the ends. Sulfur hexafluoride gas is employed in order to decrease the required drive frequency and thus increase the maximum drive amplitude. A smooth constriction in the middle of the tube serves to detune the second mode, so that this mode is not directly excited.

8:15

4aPAa2. Effect of forcing function on nonlinear acoustic standing waves. Joshua R. Finkbeiner (Illinois Inst. of Technol., Grad. College, 3300 S. Federal St., Chicago, IL 60616, finkjos@iit.edu), Xiaofan Li, Ganesh Raman (Illinois Inst. of Technol., Chicago, IL 60616), Christopher C. Daniels (Ohio Aerosp. Inst., Cleveland, OH 44142), and Bruce M. Steinetz (NASA Glenn Res. Ctr., Cleveland, OH 44135)

Nonlinear acoustic standing waves of high amplitude have been demonstrated by utilizing the effects of resonator shape to prevent the pressure waves from entering saturation. Experimentally, nonlinear acoustic standing waves have been generated by shaking an entire resonating cavity. While this promotes more efficient energy transfer than a piston-driven resonator, it also introduces complicated structural dynamics into the system. Experiments have shown that these dynamics result in resonator forcing functions comprised of a sum of several Fourier modes. However, previous numerical studies of the acoustics generated within the resonator assumed simple sinusoidal waves as the driving force. Using a previously developed numerical code, this paper demonstrates the effects of using a forcing function constructed with a series of harmonic sinusoidal waves on resonating cavities. From these results, a method will be demonstrated which allows the direct numerical analysis of experimentally generated nonlinear acoustic waves in resonators driven by harmonic forcing functions.

8:30

4aPAa3. Determination of dimensionless attenuation coefficient in shaped resonators. Christopher C. Daniels (OAI, 22800 Cedar Point Rd., Cleveland, OH 44142), Joshua R. Finkbeiner (Illinois Inst. of Technol., Grad. College, 10 W. 32nd St., Chicago, IL 60616), Bruce M. Steinetz (NASA Glenn Res. Ctr., Cleveland, OH 44135), Xiaofan Li, and Ganesh Raman (Illinois Inst. of Technol., Chicago, IL 60616)

The value of the dimensionless attenuation coefficient is an important factor when numerically predicting high-amplitude acoustic waves in shaped resonators. Both the magnitude of the pressure waveform and the quality factor rely heavily on this dimensionless parameter. Previous authors have stated the values used, but have not completely explained their methods of determination. This work fully describes the methodology used to establish this important parameter. Over a range of frequencies encompassing the fundamental resonance, the pressure waves were experimentally measured at each end of the shaped resonators. At the corresponding dimensionless acceleration, the numerical code modeled the acoustic waveforms generated in the resonator using various dimensionless attenuation coefficients. The dimensionless attenuation coefficient that most closely matched the pressure amplitudes and quality factors of the experimental and numerical results was determined and will be used in subsequent studies.

8:45

4aPAa4. Numerical simulation of streaming in high amplitude standing wave resonators. Said Boluriaan and Philip Morris (Dept. of Aerosp. Eng., Penn State Univ., 233 Hammond Bldg., University Park, PA 16802)

Classical theories of acoustic streaming in a long standing wave resonator predict a streaming velocity with a parabolic profile across the resonator. Recently, an analysis by Menguy and Gilbert [Acustica **86**, 249–259 (2000)] has shown that the parabolic streaming velocity profile is obtained only for low acoustic pressure amplitudes. At a higher amplitude, the streaming velocity deviates from a parabolic distribution. In a more recent experiment, Thompson and Atchley [J. Acoust. Soc. Am. **111**, 2418 (2002)] have also observed streaming velocities that are different from those predicted by the classical theory when a high amplitude standing wave is used. Furthermore, there are indications that the type of the boundary condition (adiabatic or isothermal, for instance) may affect the mean velocity. In the present research, numerical simulations are performed for a high amplitude standing wave in a resonator. A time-accurate, high-order, finite difference approach on a clustered rectangular grid structure is used and the full Navier–Stokes equations are solved. Results are presented for a wide range of operating parameters and are compared with both the Menguy and Gilbert analysis and the experiments. Moreover, the time evolution of the streaming velocity is investigated.

9:00

4aPAa5. Acoustic streaming at high amplitudes in plane and cylindrical channels. Mark F. Hamilton, Yurii A. Ilinskii, and Evgenia A. Zabolotskaya (Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712-1063)

At low amplitudes, acoustic streaming generated by standing waves in a plane channel is similar to streaming in a cylindrical tube. However, numerical simulations based on fully nonlinear equations for the streaming flow reveal significant differences at high amplitudes, at which Reynolds numbers for the streaming exceed unity. In these calculations, the acoustic field driving the streaming is prescribed independently. With increasing acoustic amplitude, the centers of the streaming vortices shift, the streamlines lose their symmetry, and the transverse velocity distribution departs from a parabolic profile. The vortex centers shift in opposite directions in the cylinder and the plane channel. In the cylinder the vortex centers move toward antinodes of the acoustic velocity field as the amplitude is increased, whereas in the plane channel the centers move toward the nodes. Simultaneously, the transverse streaming velocity profiles are observed to become narrower in some regions and broader in others. In general, these nonlinear streaming effects are more pronounced for the plane channel and therefore must be taken into account for Reynolds numbers smaller than for the cylinder. The influence of harmonic generation in the acoustic field will also be discussed. [Work supported by ONR.]

9:15

4aPAa6. Systematic acoustic loading of an aeroacoustic whistle. W. V. Slaton and J. C. H. Zeegers (Dept. of Appl. Phys., Eindhoven Univ. of Technol., P.O. Box 513, 5600 MB Eindhoven, The Netherlands)

Strong self-sustained acoustic oscillations may occur in a gas pipe network under certain gas flow velocities within the network. The pipe network under consideration consists of a main pipe, with a variable mean airflow, with two closed coaxial side branches of variable but equal length joined to the main pipe at right angles. Coupling between resonant acoustic standing waves and instabilities of the shear layers separating the flow in the main pipe from the stagnant gas in the closed side branches leads to strong acoustic oscillations (easily 10% of the mean pressure) at a frequency corresponding to the half-wavelength acoustic mode defined by the total side-branch length. The acoustic power available in the resonant mode has been measured to compare with theoretical predictions and previous two-microphone measurement techniques as well as to inspire confidence when designing an aeroacoustically driven thermoacoustic heat pump. The acoustic load consists of a variable acoustic resistance and coupled acoustic compliant volume, which is similar to devices used to measure power in thermoacoustic engines. Data will be presented which characterize the performance of the aeroacoustic sound source under variable acoustic loading. [Work supported by Shell International Exploration and Production B.V.]

9:30

4aPAa7. Bifurcations of diffusive soliton solutions to Kuznetsov's equation. Pedro M. Jordan (Code 7181, Naval Res. Lab., Stennis Space Ctr., MS 39529, pjordan@nrlssc.navy.mil)

Exact traveling wave solutions are determined for Kuznetsov's equation, a nonlinear PDE of 3rd order which describes finite amplitude acoustic disturbances in thermoviscous Newtonian fluids. Specifically, it is shown that traveling wave solutions exist, and assume the form of diffusive solitons, if and only if the Mach number is less than or equal to a bifurcation value. It is also shown that the wave speed v is always supersonic, that $\mathrm{Max}[v]$ occurs at the bifurcation value of the Mach number, and that a shock develops as the Reynolds number tends to infinity. Finally, special cases and asymptotic results are listed, a relationship to Burgers' equation is noted, and 3-D bifurcation diagrams are given.

THURSDAY MORNING, 1 MAY 2003 ROOMS 209/210, 10:00 TO 11:45 A.M.

Session 4aPAb

Physical Acoustics: Ultrasonics and Inhomogeneous Media

David L. Gardner, Chair

Material Division, Los Alamos National Laboratory, M.S. K764, Los Alamos, New Mexico 87545

Contributed Papers

10:00

4aPAb1. On the ability of bounded inhomogeneous waves to experimentally verify the behavior of infinite inhomogeneous plane waves. Nico F. Declercq, Joris Degrieck (Dept. of Mech. Constr. & Prod., Soete Lab., Ghent Univ., Sint Pietersnieuwstraat 41, B-9000 Gent, Belgium), and Oswald Leroy (IRC-KULAK Univ., E. Sabbelaan 53, 8500 Kortrijk, Belgium)

Infinite inhomogeneous plane waves have been controversial because of the appearance of a reflection coefficient exceeding unity in some cases. Even though it had already been shown that this is consistent with the principle of energy conservation, and results in a shift in position, full acceptance of the notion of infinite inhomogeneous plane waves appeared only after some experiments have been performed that produce results that are in agreement with theory. However, experiments can only be performed by means of bounded inhomogeneous waves. The current paper answers the ultimate question why bounded inhomogeneous waves are able to behave as if they were infinite inhomogeneous waves and what exactly is the physical and mathematical connection between bounded inhomogeneous waves and infinite inhomogeneous waves. [Work supported by The Flemish Institute for the Encouragement of the Scientific and Technological Research in Industry (I.W.T.).]

10:15

4aPAb2. Backward displacement of ultrasonic waves reflected from a corrugated interface. A. A. Teklu and M. A. Breazeale (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, 1 Coliseum Dr., Univ., MS 38677)

A backward displacement can occur when an ultrasonic beam is reflected from an interface having a superimposed periodicity. This was first observed by Breazeale and Torbett [Appl. Phys. Lett. **29**, 456 (1976)]. They observed that at a certain critical angle the reflected beam was shifted at 6 MHz and not at 2 MHz, in agreement with the theory of Tamir and Bertoni [J. Acoust. Soc. Am. **61**, 1397–1413 (1971)]; however, the angle predicted by assuming that the shift was caused by a Rayleigh surface wave did not agree with experiment. Recent experiments using the

4a THU. AM

schlieren technique showed backward displacement at an incident angle of 25 deg, in good agreement with the value 23.69 deg calculated using the inhomogeneous wave theory [Briers *et al.*, J. Acoust. Soc. Am. **106**, 682–7 (1999) (private communication with N. F. Declercq)]. The inhomogeneous wave theory also predicts a new kind of leaky surface wave at an incident angle of 22.55 deg [N. F. Declercq (private communication)], again in agreement with our measurements. Schlieren photographs of the backward-displaced beams at a water-brass interface are presented and discussed, as is the effect of the presence of an evanescent wave at the interface.

10:30

4aPAb3. Analysis of particular phononic structures using single integral equations. Fadoulourahmane Seydou, Ramani Duraiswami, and Nail A. Gumerov (Perceptual Interfaces and Reality Lab., Inst. for Adv. Computational Studies, Univ. of Maryland, College Park, MD 20742)

A fast method for determining phononic (and photonic) bandgaps in composite materials is developed. It is known that in the propagation of waves in a 3D medium containing N scatterers arranged periodically, there exist refractive indices for which such structures have bandgaps, i.e., frequencies for which no waves can propagate inside. Our task is to find the frequencies that generate these prohibited waves. This requires the solution of an eigenvalue problem for the Helmholtz operator. To solve this problem we choose an alternate route which uses boundary integral equations. We derive a single integral equation on each of the interfaces between the outer region and the scatterers, considering a general transmission boundary condition, by using a hybrid method using layer potentials and Green's formula. This approach reduces the number of unknowns considerably in comparison to other methods, but requires the treatment of large dense matrices and many matrix vector multiplications. To remedy this, we use the Fast Multipole Method. For solving the eigenvalue problem we discuss two methods: the Newton method and a method based on the Cauchy formula. Details of the numerical implementation, and results will be presented for different cases: sound hard, sound soft, impedance and transmission boundary conditions. [Work partially supported by NSF Award 0219681 is gratefully acknowledged.]

10:45

4aPAb4. Band-gap engineering in periodic acoustic stub tuners: Spectral gaps and transmission. Manvir S. Kushwaha (Inst. of Industrial Sci., Univ. of Tokyo, 4-6-1 Komaba, Meguro-Ku, Tokyo 153-8505, Japan), X. F. Wang, and P. Vasilopoulos (Concordia Univ., Montreal, QC H3G 1M8, Canada)

A theoretical investigation on acoustic wave propagation in a periodically stubbed waveguide is reported. In general the waveguide segments and stubs are made of different materials. The acoustic wave in such a system has two independent polarizations: out-of-plane and in-plane modes. The band structure and transmission spectrum are studied for diverse geometries using a simple and efficient version of the transfer-matrix method. For the *same material* between the waveguide and *symmetric* stubs the width of some gaps can change, upon varying the stub length or width, by more than one order of magnitude. A further modulation can be achieved for *different material* between the stubs and the main waveguide or if the stubs are *asymmetric*. The gaps in the band structure of an infinitely long system correspond to those in the transmission spectrum of the same system but with *finite* number n of units. For n finite (i) there exist pseudogaps that gradually turn into complete gaps with increasing n and (ii) the introduction of defects gives rise to states in the gaps and leads to transmission resonances.

11:00

4aPAb5. Group velocity manipulation in simple acoustic band gap filters. W. M. Robertson, C. Baker, and C. Brad Bennett (Dept. of Phys. and Astron., Middle Tennessee State Univ., Murfreesboro, TN 37132)

A simple experimental configuration is presented in which the group velocity of acoustic pulses is adjusted from much slower to much faster than the speed of sound. The experiment is an acoustic analog of two much studied optical phenomena: the superluminal group velocity achieved when optical pulses are tunneled through regions of high absorption or attenuation, and slow (or even stopped) light propagation in the vicinity of strong material dispersion, typically realized using electromagnetically induced transparency in atomic vapors. For the acoustic experiments described here, attenuation and dispersion were created using periodically structured waveguides. The periodicity results in one-dimensional acoustic band gaps, frequency intervals in which propagation of sound is strongly suppressed. The introduction of defects in the perfect periodicity leads to narrow transmission bands within the band gaps. There is strong normal dispersion in the vicinity of these defect modes leading to very slow group velocities. By tuning the carrier frequency of acoustic pulses through the forbidden and defect regions the group velocity can be adjusted from 0.22 to 1.8 times the speed of sound in the straight unstructured waveguide. These results are shown to be consistent with a straightforward theoretical model.

11:15

4aPAb6. Modified Born approximation for solving scattering problems. Igor L. Oboznenko (Kyiv Polytechnic Univ., Kyiv 03057, Ukraine), Vladimir Genis (Drexel Univ., Philadelphia, PA 19104), and Dat H. Tran (Nagoya Univ., Nagoya 464-8603, Japan)

Scattering of ultrasonic waves in inhomogeneous media is described by the inhomogeneous differential equations. Such equations could be solved using a method of consecutive approximations, such as the Born approximation. The Born approximation is applicable when $(n-1)<1$ and $2ka(n-1)<1$, where n is the refraction index and ka is the wave dimension of the scatterer. For the Born approximation, it is assumed that the acoustic field inside of a scatterer is substituted by the acoustic field of the incident wave, along with the wave number of the surrounding media. In this work, the modified Born approximation is used, where the acoustic field inside of a scatterer is substituted by the acoustic field of the incident wave, along with the wave number of the scatterer. A similar approach is used for solving the scattering problems of multilayered scatterers, which have weak scattering properties. The computed and experimental scattering characteristics for the elastic scatterers with various acoustical impedance are presented. It is demonstrated that the modified Born approximation more accurately describes the scattering problems for the scatterers with $ka>1$.

11:30

4aPAb7. Modeling scattering enhancements at isolated resonances using energy conservation, reciprocity, symmetry, and the optical theorem. Philip L. Marston and Curtis F. Osterhoudt (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Sound scattered by some objects in water exhibits isolated narrow resonances that are sufficiently large in amplitude to dominate the low-frequency scattering. Examples include the quadrupole mode of thin spherical shells and of solid plastic spheres [B. T. Hefner and P. L. Marston, J. Acoust. Soc. Am. **107**, 1930–1936 (2000)] and organ-pipe modes of water-filled pipes [C. F. Osterhoudt and P. L. Marston, J. Acoust. Soc. Am. **110**, 2773 (2001)]. This presentation concerns simple methods for approximating the scattering. In the case of spheres, ray theory for the backscattering reduces to a simple form for high-Q modes: Eq. (58) of Marston [J. Acoust. Soc. Am. **83**, 25–37 (1988)]. This result gives the

backscattering form function at resonance (in the usual normalization) to have the magnitude $2(2n+1)/ka$. Here n is the partial wave index associated with the mode of the sphere and ka is the product of the wave number and the sphere radius. This result may also be derived directly from energy conservation and the optical theorem. Scattering amplitudes associated with high-Q organ pipe resonances of open cylindrical pipes are also derived here by a related method using the energy conservation, reciprocity, symmetry, and the optical theorem.

THURSDAY MORNING, 1 MAY 2003 ROOM 206, 8:00 A.M. TO 12:00 NOON

Session 4aPPa

Psychological and Physiological Acoustics: Spatial Hearing, Binaural Processing and Masking (Poster Session)

Marc A. Fagelson, Chair

Communicative Disorders, East Tennessee State University, Box 70643, Johnson City, Tennessee 37614-0643

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon. To allow for extended viewing time, posters will remain on display until 4:00 p.m.

4aPPa1. Median plane localization: dependence on spectral content. Gerald Ng and H. Steven Colburn (Hearing Res. Ctr., Boston Univ., 44 Cummington St. #427, Boston, MA 02215, geraldng@bu.edu)

Subjects were asked to report the elevation of noise stimuli differing in their spectral content. Some stimuli contained a single frequency band differing in center frequency and bandwidth; others contained multiple equal-amplitude frequency bands. All of the stimulus frequencies were in the range between 1 kHz and 16 kHz. Speakers were located in the median-sagittal plane at six frontal locations (−30, −15, 0, 15, 30, 45), as well as above (90) and behind (180) the listener. The experiments were conducted in a sound-treated room with the speakers concealed from view. Stimuli were 200 ms in duration and presented from only one speaker at a time. On each trial subjects used a pen to mark the perceived stimulus location(s) on a pre-printed coordinate diagram; subjects could indicate the perception of multiple sources and accurately represent auditory image widths and/or ambiguities. Preliminary data confirm historical results showing that localization of single noise bands depends on center frequency and bandwidth. When localizing multiple-band stimuli, responses depended on band density: with a few sparse bands, most subjects heard multiple source locations; for high band densities, subjects heard a single source at or near the true speaker location. [Work supported by NIDCD (R01 00100).]

4aPPa2. Cocktails for Franssen: Asynchronous transient bias and other attentional factors in auditory localization. William M. Whitmer, Christopher A. Brown, Raymond H. Dye, Jr., and Noah F. Jurcin (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626, wwhitme@luc.edu)

The cocktail-party paradigm was applied to nonspeech signals. Pure-tone stimuli were used in an extension of the Franssen effect, an auditory illusion wherein the location of a slow-onset signal is perceived to be the same as a simultaneous, contralateral sudden-onset signal. Listeners heard simultaneous sudden-onset (transient) and contralateral slow-onset (steady-state) tones in a reverberant environment with a second delayed transient from a third azimuthal location. Results showed that the Franssen effect was either maintained or ''reset,'' but not reduced. The ongoing steady-state tone was perceived either at the initial-transient and then delayed-transient location, or at the initial transient location only. None of the listeners showed a location bias to the delayed transient tone when its frequency differed from the initial signal frequency or was replaced with noise. In additional conditions based on the ''false Haas effect,'' consonant–vowel pairs representing transient and steady-state signals were segregated contralaterally in a reverberant space. Results showed no resemblance to the Franssen effect. In general, results indicated that the role of attention is fundamental to the localization of an ongoing stimulus. [Work supported by NIH.]

4aPPa3. Effects of reverberation and experience on distance perception in simulated environments. Matthew Schoolmaster, Norbert Kopco, and Barbara G. Shinn-Cunningham (Boston Univ. Hearing Res. Ctr., Boston, MA 02215, shinn@cns.bu.edu)

Individually measured head-related-transfer functions were used to simulate different acoustic environments in order to see how listener experience influences auditory distance perception. Three environments were simulated: an anechoic space, a classroom with the listener in the room center, and the same classroom with the listener in the room corner. Each subject completed two series of trials consisting of 360 trials in each environment. Each series consisted of six experimental sessions, with only one session performed per day. In the fixed-room series, one environment was tested in each session (ordered randomly). In the mixed-room series, trials from the three environments were intermingled within each session. Listeners were divided into two groups: Group A performed the fixed-room series followed by the mixed-room series, while Group B did the reverse. Sources were simulated from ahead and to the right of the listener at distances ranging from 0.15 to 1.7 m. Preliminary analysis indicates that performance improves with experience in the fixed-room but not the mixed-room trials. These results suggest that listeners learn to calibrate auditory distance percepts based on recent experience with the reverberation and echoes in a particular environment. [Work supported by AFOSR and the Sloan Foundation.]

4aPPa4. Cues for "front-back confusion." Mitsuo Matsumoto, Kiyoaki Terada, Mikio Tohyama (Kogakuin Univ., 2665-1 Nakano-machi, Hochioji, Tokyo 192-0015, Japan, matsu@tohyama.cc.kogakuin.ac.jp), and Hirofumi Nakajima (Nittobo Acoust. Eng. Co., Ltd., 1-13-12 Midori, Sumida-ku, Tokyo 130-0021, Japan)

Front-back confusion in the median plane disturbs sound image localization, especially for a sound image simulated using a transaural system and a dummy head. In the median plane, the spectral cue is a major cue for front-back discrimination. The front-back difference in HRTF characteristics between real heads and a dummy head was measured. For real heads, the frequency amplitude at 1.5 kHz with a front-source HRTF was significantly smaller than with a rear-source HRTF. In contrast, for the dummy head, no significant difference was found. Two signals band-limited up to 8 kHz in a subjective test were used: one had a dip amplitude frequency of -15 dB at 1.5 kHz on the frequency axis (characteristic of gfronth) and the other had a peak of 15 dB (characteristic of grearh) at the same frequency. A sound image reproduced by an actual frontal sound source and the signal with the peak was perceived ambiguously. A sound image reproduced by an actual rear sound source and the signal with the dip was perceived to be ambiguous.

4aPPa5. Auditory performance in an open sound field. Kim F. Fluitt, Tomasz Letowski, and Timothy Mermagen (US Army Res. Lab., Auditory Res. Team, AMSRL-HR-SD, Bldg. 520, kfluitt@arl.army.mil)

Detection and recognition of acoustic sources in an open field are important elements of situational awareness on the battlefield. They are affected by many technical and environmental conditions such as type of sound, distance to a sound source, terrain configuration, meteorological conditions, hearing capabilities of the listener, level of background noise, and the listener's familiarity with the sound source. A limited body of knowledge about auditory perception of sources located over long distances makes it difficult to develop models predicting auditory behavior on the battlefield. The purpose of the present study was to determine the listener's abilities to detect, recognize, localize, and estimate distances to sound sources from 25 to 800 m from the listing position. Data were also collected for meteorological conditions (wind direction and strength, temperature, atmospheric pressure, humidity) and background noise level for each experimental trial. Forty subjects (men and women, ages 18 to 25) participated in the study. Nine types of sounds were presented from six loudspeakers in random order; each series was presented four times. Partial results indicate that both detection and recognition declined at distances greater than approximately 200 m and distance estimation was grossly underestimated by listeners. Specific results will be presented.

4aPPa6. Insensitivity to large differences in interaural correlation. John F. Culling (School of Psych., Cardiff Univ., Cardiff CF11 9BQ, UK)

Sensitivity to differences in interaural correlation was measured using an adaptive threshold technique. The 2-down/1-up adaptive track began at the opposite correlation value from the standard (e.g., starting at 1 and converging on -1). Each 3I-FC trial was followed by correct/incorrect feedback. Steps in the adaptive track were initially linear and switched to logarithmic as the track passed zero interaural correlation. First, four listeners were trained to asymptote using a band of noise (462–540 Hz). The listeners were then trained under identical circumstances, but with remote flanking bands (0–330 and 719–3000 Hz) of interaurally correlated, synchronously gated noise, equal in spectrum level to the target band. The listeners were also occasionally given an adaptive track beginning at zero interaural correlation. With no flanking bands, three listeners experienced difficulty learning to adapt from 1 to -1, frequently recording threshold differences in correlation close to 2. Little difficulty was experienced when adapting from -1 to 1 and none when adapting from zero in either direction. When remote flanking bands were added, all listeners experienced greater difficulty. Only one listener learned to converge on -1. The salience of cues used by the listeners seemed to decrease when comparing high positive and negative correlations.

4aPPa7. Binaural versus better-ear listening. Jacob W. Scarpaci, N. I. Durlach, and H. Steven Colburn (Hearing Res. Ctr. and Dept. of Biomed Eng., Boston Univ., Boston, MA 02215, scarpaci@bu.edu)

Advantages of binaural over monaural hearing in noisy environments are reduced when the monaural stimulation is derived from the monaural signal with the better signal-to-noise ratio (better-ear listening). In the reported experiments, conducted in a soundproof room with two speakers and a custom-designed, noise-cancellation headset, speech intelligibility in the presence of interference was measured for both binaural and better-ear configurations. The headset, which incorporated two microphones (located at the two ears) and two insert earphones, was used to present binaural stimulation or better-ear (better-microphone) monaural stimulation. Although the results varied significantly with the locations of the target and interference sources, the advantage of binaural listening over better-ear listening was no more than a few dB. In addition to reporting the data obtained in these experiments, relations to previous work on better-ear listening and CROS hearing aids, as well as to current work on cochlear implants, are discussed. [Work supported by NIDCD (00100).]

4aPPa8. Modulation masking of lateralization based on envelope interaural phase differences. Stanley Sheft and William A. Yost (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626, ssheft@luc.edu)

Modulation-filterbank models discard envelope phase above very low modulation rates. To evaluate the role of relative phase in binaural envelope processing, modulation masking of envelope-based lateralization was measured with a left/right task. A diotic 400-ms wideband noise was modulated by a two-tone function. An interaural phase difference was applied to the probe component of the modulator with the masker component always diotic. The probe modulation rate was 80, 160, or 320 Hz with masker rate varying from 40 to 1280 Hz. Masker modulation interfered with probe lateralization at all probe rates. With 80-Hz probe modulation, the masking function was asymmetric with maskers above the probe more effective than those below. At the higher probe modulation rates, interference was generally greatest when the probe and masker rates were close. However, function asymmetry persisted at these higher rates. In some conditions, a diotic phase shift of the masker had a significant effect. Unlike monaural results, binaural envelope processing showed high-rate phase sensitivity and asymmetric tuning. The asymmetric results and masker-phase effects are consistent with viewing the envelope manipulations as interaural gating asynchronies described by group delay in the modulation domain. [Work supported by NIH.]

4aPPa9. Speech perception in tight acoustic spaces. Nandini Iyer, Rama Ratnam, Sandeep Phatak (Beckman Inst., 405 N. Mathews Ave., Urbana, IL 61801), Charissa Lansing, and Albert Feng (Beckman Inst. & Univ. of Illinois at Urbana—Champaign, Urbana, IL 61801)

Acoustic cues for sound localization and speech perception in reverberant environments are more complex than those in anechoic conditions. The ability to utilize these cues deteriorates as a function of age. This study examined the speech perception ability of elderly listeners in well-defined, enclosed spaces and identified signal-processing strategies that might enhance this ability. For this, we performed tests in three environments: a car, a plywood cube, and a conference room. Pre-recorded monosyllabic words (Modified Rhyme Test) were played in quiet or in the presence of an 8-talker babble from a loudspeaker and recorded using two microphones placed at KEMARs ears. These signals were played to 10 young and 10 elderly listeners with normal hearing. We evaluated speech perception abilities of the listeners as a function of the following variables: (1) the location and distance of the loudspeaker from the listener, and (2) the number of reflecting surfaces. We tested the hypothesis that diotic presentations of the signal (better ear signal to both ears, or active steering followed by a presentation of the summed signal) improved speech perception abilities in these situations.

4aPPa10. Discrimination between harmonic tone complexes and bandpass noise signals presented within lowpass noise. Steven van de Par, Armin Kohlrausch, and Jeroen Breebaart (Philips Res., Prof. Holstlaan 4, 5656 AA, Eindhoven, The Netherlands)

Threshold levels were measured at which harmonic tone complex targets could be *discriminated* from bandpass noise targets both presented within a diotic 2-kHz lowpass filtered noise masker with 65 dB SPL. In the 3IFC paradigm, one interval contained a sine-phase tone complex (20-Hz spacing) while the remaining intervals contained a bandpass noise with the same bandwidth, center frequency (1 kHz) and overall level as the tone complex. In addition, *detection* thresholds were measured for both targets using the same masker. When interaurally in-phase targets with a 100-Hz bandwidth were used, discrimination thresholds were about 8 dB higher than the detection thresholds. When out-of-phase targets were used, the difference between discrimination and detection thresholds was 18 dB, which was mainly due to a decrease of about 15 dB of the detection thresholds (BMLD). For 1500-Hz wide targets, the discrimination and detection thresholds were more similar. For the in-phase condition, the discrimination and detection threshold overlapped while for the out-of-phase conditions, discrimination thresholds exceeded the detection thresholds by only 5 dB. Assuming that discrimination is based on the processing of the temporal structure of the targets, this processing seems to be more efficient for wide- than for narrow-band targets.

4aPPa11. Basilar-membrane nonlinearity effects on tones and speech. Judy R. Dubno, Amy R. Horwitz, and Jayne B. Ahlstrom (Dept. of Otolaryngol.–Head & Neck Surgery, Medical Univ. of South Carolina, 135 Rutledge Ave., P.O. Box 250550, Charleston, SC 29425, dubnojr@musc.edu)

The contribution of nonlinearities in the basilar-membrane response to the understanding of speech in noise was estimated by measuring growth of simultaneous masking for tones and speech. Speech signals were bandpass-filtered nonsense syllables; consonant recognition and pure-tone thresholds were measured as a function of masker level in two conditions: (1) masker within the passband of the speech and (2) masker below the passband of the speech. With this low-frequency masker, the effect of the masker at the ''signal'' place is assumed to grow linearly whereas the growth of response to the speech at the ''signal'' place is compressed. Growth-of-masking functions were also measured for short-duration tones in narrowband maskers centered at or below the signal frequency. Subjects were younger and older adults with normal hearing; these older adults were included given that age-related changes in nonlinearities may occur even with minimal threshold elevation. For both subject groups, large individual differences were observed in upward spread of masking and in the extent to which the subsequent variance in audibility accounted for the variance in speech-recognition scores. These results will be discussed along with estimates of basilar-membrane nonlinearities obtained from subject's growth-of-masking functions and otoacoustic emissions. [Work supported by NIH/NIDCD.]

4aPPa12. Upward spread of informational masking in normal-hearing and hearing-impaired listeners. Joshua M. Alexander and Robert A. Lutfi (Univ. of Wisconsin–Madison, Waisman Ctr., 1500 Highland Ave., Madison, WI 53705)

Thresholds for pure-tone signals of 0.8, 2.0, and 5.0 kHz were measured in the presence of a simultaneous multitone masker in 15 normal-hearing and 8 hearing-impaired listeners. The masker consisted of fixed-frequency tones ranging from 522–8346 Hz at 1/3-octave intervals, excluding the 2/3-octave interval on either side of the signal. Masker uncertainty was manipulated by independently and randomly playing individual masker tones with probability $p=0.5$ or $p=1.0$ on each trial. Informational masking (IM) was estimated by the threshold difference ($p=0.5$ minus $p=1.0$). Decision weights were estimated from correlations of the listener's response with the occurrence of the signal and individual masker components on each trial. IM was greater for normal-hearing listeners than for hearing-impaired listeners, and most listeners had at least 10 dB of IM for one of the signal frequencies. For both groups, IM increased as the number of masker components below the signal frequency increased. Decision weights were also similar for both groups—masker frequencies below the signal were weighted more than those above. Implications are that normal-hearing and hearing-impaired individuals do not weight information differently in these masking conditions and that factors associated with listening may be partially responsible for the greater effectiveness of low-frequency maskers. [Work supported by NIDCD.]

4aPPa13. Criteria placement in staircase procedures. Virginia M. Richards and Rong Huang (Dept. of Psych., Univ. of Pennsylvania, 3815 Walnut St., Philadelphia, PA 19104, rongh@psych.upenn.edu)

Data from several masking studies are evaluated to determine whether observers alter their decision criteria as signal levels increase and decrease according to an adaptive staircase algorithm. In a Yes/No procedure, a 1000-Hz signal tone is added to one of twelve maskers. Each masker is composed of six equal-amplitude tones with frequencies drawn at random from a 200–5000 Hz range. In two conditions, the maskers are either fixed across several sessions or randomly drawn prior to each trail. Data from at least 15 staircases are combined, and for each signal level the overall hit and false alarm rates are determined. Regardless of condition, fixed or random, observers tend to adopt a single criterion across all signal levels. If it is assumed that in the random condition a single criterion is used across all trials, the location of the mean of the no-signal distributions relative to the criterion can be estimated. For the current data, the pattern of no-signal distributions is not obviously related to masker properties. [Work supported by NIH.]

4aPPa14. Development of a fast method for determining psychophysical tuning curves. Aleksander Sek, Ewa Skrodzka (Inst. of Acoust., A. Mickiewicz Univ., 61-614 Poznan, Umultowska 85, Poland), and Brian C. J. Moore (Dept. of Exp. Psych., Univ. of Cambridge, Downing St., Cambridge CB2 3EB, UK)

A psychophysical tuning curve (PTC) is usually measured by determining the level of a narrowband noise required to mask a fixed low level tone, for several masker center frequencies. PTCs could be useful clinically for assessing frequency selectivity and for the diagnosis of dead regions in the cochlea. When the signal frequency falls in a dead region, the tip of the PTC is shifted away from the signal frequency. However, PTCs determined in the traditional way are too time consuming for use in clinical practice. A fast method for determining PTCs is being developed and evaluated. This uses a band of noise that sweeps in center frequency. A Bekesy method is used to track the masker level required for threshold. For normally hearing subjects, the new method gives stable results, and PTCs similar in shape to those determined in the traditional way, when the masker sweeps over a 2-octave range in about 4 minutes, and the level changes by 1–2 dB/s. Preliminary results using hearing-impaired subjects also show a good agreement with the traditional method. However, further work is required to determine optimum values for the noise bandwidth and rate of change of frequency and level.

4aPPa15. Phase rotation thresholds and its use in watermarking technique. Aleksander Sek, Yoiti Suzuki, Ryouichi Nishimura, and Kotaro Sonoda (Res. Inst. of Elec. Commun., Tohoku Univ., Katahira 2-1-1, Aoba-ku, Sendai 980-8577, Japan)

Strong demands for the protection of copyrights for high fidelity audio recordings as well as broadcasting signals via a network brought about a development of so-called watermarking techniques. One of them is based on a phase modulation referred to in here as a phase rotation (PR). This method assumes a poor sensitivity of the auditory system to the phase changes in a signal and embeds a watermark in the recording by means of a filter with linear phase characteristics. However, the sensitivity of the auditory system to the phase rotation has not been studied in detail, especially for complex signals. In this paper we present a preliminary result of threshold measurements of the phase rotation applied to different types of signals. It appears that the thresholds for detecting the phase rotation, when expressed by means of a time group delay, depend on the type of the signal and the frequency region that the phase rotation is applied for. The detection of the phase rotation is approximately independent on the rotation rate up to 120 Hz. When the phase rotation was applied to several components of a harmonics complex, the highest values of the thresholds were observed in the low frequency region.

4aPPa16. Psychometric functions for informational masking. Robert A. Lutfi, Doris J. Kistler, Michael R. Callahan, and Frederic L. Wightman (Waisman Ctr., Univ. of Wisconsin, Madison, WI 53706)

The method of constant stimuli was used to obtain complete psychometric functions (PFs) from 44 normal-hearing listeners in conditions known to produce varying amounts of informational masking. The task was to detect a pure-tone signal in the presence of a broadband noise and in the presence of multitone maskers with frequencies and amplitudes that varied at random from one presentation to the next. Relative to the broadband noise condition, significant reductions were observed in both the slope and the upper asymptote of the PF for multitone maskers producing large amounts of informational masking. Slope was affected more for some listeners while asymptote was affected more for others. Mean slopes and asymptotes varied nonmonotonically with the number of masker components in much the same manner as mean thresholds. The results are consistent with a model that assumes trial-by-trial judgments are based on a weighted sum of dB levels at the output of independent auditory filters. For many listeners, however, the weights appear to reflect how often a nonsignal auditory filter is mistaken for the signal filter. For these listeners adaptive procedures may produce a significant bias in the estimates of threshold for conditions of informational masking. [Work supported by NIDCD.]

4aPPa17. Effect of signal-masker similarity and signal uncertainty on informational masking. Eunmi L. Oh (Samsung AIT, P.O. Box 111, Suwon 440-600, Korea, oh@sait.samsung.co.kr) and Robert A. Lutfi (Univ. of Wisconsin, Madison, WI 53706)

An experiment was conducted to determine the relative importance of signal-masker similarity and signal uncertainty as factors affecting informational masking. The masker was a complex of fixed-frequency tones whose levels varied independently and at random on each presentation. The signal was a complex of fixed-frequency tones whose levels were sampled from one of two Gaussian distributions differing in mean (SD $=5$ dB). The levels of the signal tones were sampled independently (IS condition) or were the same as determined by a single sample (SS condition). The listener's task was to select the signal from the distribution with the higher mean. It was expected that if uncertainty were the dominant factor better performance would be obtained in the IS condition—this due to the reduction in uncertainty associated with multiple independent ''looks'' at the signal. If, however, similarity were the dominant factor better performance would be obtained in the SS condition—this due to the dissimilarity of signal and masker spectra (flat versus irregular). Four of six listeners showed the first result, the remaining two the second result. The remaining two listeners, however, showed results inconsistent with a similarity effect with maskers having flat spectra. [Work supported by NIDCD.]

4aPPa18. Acceptance of background noise as a function of speech presentation level. Clifford A. Franklin, Jr., Anna K. Nabelek, and Samuel B. Burchfield (Dept. of Audiol. and Speech Pathol., Univ. of Tennessee, South Stadium Hall, Knoxville, TN 37996-0740)

The acceptance of background noise while listening to speech (ANL) at different speech presentation levels was assessed. Twenty listeners (10 male) between 18–30 years with normal hearing listened to a narrative at speech presentation levels of 20, 34, 48, 62, and 76 dB HL, then adjusted the background noise to the highest intensity level that they would be willing to accept for an extended listening period. The ANL is the intensity of the speech presentation level minus the intensity level of the background noise. The group mean ANLs for presentation levels of 20, 34, 48, 62, and 76 dB HL were 10.60, 14.25, 17.10, 21.80, and 24.55 dB, respectively. The group mean ANLs differ by approximately three and one half decibels between each presentation level. This difference between adjacent speech presentation levels is representative of a linear function. The average MCL was 43 dB HL with a standard deviation of 6.7 dB. The group mean ANL for speech presented at MCL was 15.5 dB with a standard deviation of 7.27 dB. A MANOVA for repeated measures indicated a statistically significant main effect for speech presentation level ($F=18.624$, $p=0.001$). [Work partially supported by NIDCD (NIH) R01 DC 05018.]

4aPPa19. Comparison of acceptance of background noise and speech reception threshold in quantifying the hearing aid directivity benefit. Melinda C. Freyaldenhoven, Anna K. Nabelek, and Samuel B. Burchfield (Dept. of Audiol. and Speech Pathol., The Univ. of Tennessee, South Stadium Hall, Knoxville, TN 37996-0740)

Hearing aid directivity benefit was compared as improvement in acceptance of background noise and speech reception threshold (SRT). Forty adult subjects were tested wearing binaural hearing aids in omnidirectional and directional listening conditions. Acceptance of background noise was determined by having subjects select their most comfortable listening level (MCL) for a story delivered from a loudspeaker (0). Next, speech babble was added (180) and the subjects selected the maximum background noise level (BNL) which was acceptable while listening to and following the story. The MCL minus the BNL yielded the acceptable noise level (ANL), all in dB. The difference between the ANL for the omni-directional and directional conditions is the directivity benefit. The SRT was determined by delivering spondaic words (0) at the subjects MCL. Next, speech babble was delivered (180) and adjusted until the subject could repeat 50% of the spondees. The difference between the SRT for the omni-directional and directional conditions is the directivity benefit. Mean directional benefit, $\mathrm{ANL}=3.50$ dB and $\mathrm{SRT}=3.60$ dB, were not significantly different. The individual ANLs and SRTs were significantly correlated ($r=-0.36$, $p=0.002$). The ANL procedure appears to be a viable tool for quantifying hearing aid directivity benefit. [Work supported by NIDCD (NIH) 3 RO1 DC 05018-01S1.]

4aPPa20. Relationship between acceptance of background noise and hearing aid use. Anna K. Nabelek, Samuel B. Burchfield, and Joanna D. Webster (Dept. of Audiol. and Speech Pathol., The Univ. of Tennessee, South Stadium Hall, Knoxville, TN 37996-0740, samburch@utk.edu)

Background noise produces complaints among hearing-aid users, however speech-perception-in-noise does not predict hearing-aid use. It is possible that hearing-aid users are complaining about the presence of background noise and not about speech perception. To test this possibility, acceptance of background noise is being investigated as a predictor of hearing-aid use. Acceptance of background noise is determined by having subjects select their most comfortable listening level (MCL) for a story. Next, speech-babble is added and the subjects select the maximum background noise level (BNL) which is acceptable while listening to and following the story. The difference between the MCL and the BNL is the acceptable noise level (ANL), all in dB. ANLs are being compared with hearing-aid use, subjective impressions of benefit (APHAB), speech perception in background noise (SPIN) scores, and audiometric data. Individuals who accept higher levels of background noise are more successful users than individuals who accept less background noise. Mean ANLs are 7.3 dB for full-time users ($N=21$), 12.6 dB for part-time users ($N=44$), and 13.8 dB for rejecters ($N=17$). ANLs are not related to APHAB, SPIN, or audiometric data. Results for about 120 subjects will be reported. [Work supported by NIDCD (NIH) RO1 DC 05018.]

4aPPa21. Relation between measures of speech-in-noise performance and measures of efferent activity. Brad Smith, Ashley Harkrider, Samuel Burchfield, and Anna Nabelek (Dept. of Audiol. and Speech Pathol., Univ. of Tennessee, 457 South Stadium Hall, Knoxville, TN 37996)

Individual differences in auditory perceptual abilities in noise are well documented but the factors causing such variability are unclear. The purpose of this study was to determine if individual differences in responses measured from the auditory efferent system were correlated to individual variations in speech-in-noise performance. The relation between behavioral performance on three speech-in-noise tasks and two objective measures of the efferent auditory system were examined in thirty normal-hearing, young adults. Two of the speech-in-noise tasks measured an acceptable noise level, the maximum level of speech-babble noise that a subject is willing to accept while listening to a story. For these, the acceptable noise level was evaluated using both an ipsilateral (story and noise in same ear) and a contralateral (story and noise in opposite ears) paradigm. The third speech-in-noise task evaluated speech recognition using monosyllabic words presented in competing speech babble. Auditory efferent activity was assessed by examining the resulting suppression of click-evoked otoacoustic emissions following the introduction of a contralateral, broad-band stimulus and the activity of the ipsilateral and contralateral acoustic reflex arc was evaluated using tones and broad-band noise. Results will be discussed relative to current theories of speech in noise performance and auditory inhibitory processes.

4aPPa22. Effects of multiple background talkers on word recognition and response awareness. Edward L. Goshorn and Elizabeth K. Robertson (Speech Dept., P.O. Box 3165 Tech Station, Louisiana Tech Univ., Ruston, LA 71272, egoshorn@ltparts.latech.edu)

Effects of background talkers (0, 1, 2, 3, 4, 5, 7, 10, and 14) on word recognition and awareness of errant/accurate responses were examined. Diagnostic Rhyme Test (DRT) words and background talkers were presented at 70 dB SPL (sound field) to ten normal-hearing subjects. DRT words and background talkers were digitally processed to produce equal VU meter levels. Three replicates were obtained for each condition. Performance measures were: (1) percent correct, corrected for guessing, transformed to rational arcsine units (PCGRAU); (2) subject's awareness of accurate responses (AA); (3) subject's awareness of errant responses (AE); and (4) a geometric-based symmetric awareness (SA). Awareness measures were derived from subject's confidence ratings to DRT stimuli. Both informational and direct masking effects were present. PCGRAU varied nonlinearly as number of talkers increased. One talker provided significantly more masking than two talkers and was equally effective as three, four, five, seven, and ten talkers as well as speech-spectrum noise. Fourteen talkers provided the most masking and was equally effective as white noise. In general, AA tended to diminish while AE tended to improve as additional background talkers were added. SA was best for 14-talker background noise but poorest for one-talker and speech-spectrum noise.

4aPPa23. Speech perception in gated noise: The effects of spectral and temporal resolution and auditory streaming. Su-Hyun Jin and Peggy B. Nelson (Dept. of Commun. Disord., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455)

Hearing-impaired (HI) listeners often report difficulty understanding speech in the presence of background noise, even in the presence of mild degrees of hearing loss. In addition, HI listeners show significantly less benefit from fluctuations in noise than do normal hearing (NH) listeners. Furthermore, HI listeners often show less accuracy of processing frequency and temporal information in acoustic signals and grouping them into a whole meaningful speech. The purpose of this study is to examine differences in performances of NH and HI listeners for speech perception in various types of noise. Three hypotheses will be examined in this study. First, the amount of masking release in HI listeners with mild hearing loss will be smaller than in NH listeners even when speech and noise are presented at intensities sufficient to overcome the hearing loss. Masking release will be measured for consonant recognition and sentence recognition by subtracting the percent correct in steady noise from that in gated noise. Second, the performance of HI listeners on spectral resolution, temporal resolution, and auditory streaming tasks will be significantly poorer than that of NH listeners. Third, the performance differences in these tasks may account for differences in masking release.

4aPPa24. Noise suppression algorithm based on the auditory masked threshold in listeners with cochlear hearing loss. Kathryn Arehart, Jessica Rossi-Katz (Speech-Lang.-Hearing Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309), Ajay Natarajan, and John Hansen (The Ctr. for Spoken Lang. Res., 594 UCB, Boulder, CO 80309-0594)

This study describes the formulation and evaluation of a new noise suppression scheme, with the goal of improving speech-in-noise perception by hearing-impaired listeners. Arehart *et al.* [Speech Commun. (2003)] implemented and evaluated a noise suppression algorithm based on an approach that used the auditory masked threshold in conjunction with a version of spectral subtraction to adjust the parameters used in the subtraction process based on the masked threshold of the noise across the frequency spectrum. That original formulation was based on masking properties of the normal auditory system, with its theoretical underpinnings based on MPEG-4 audio coding [Johnston (1988)]. This paper describes details of a new enhancement formulation based on masking characteristic of cochlear hearing loss. The new algorithm improves on previous formulations in two ways. First, the algorithm is implemented with generalized minimum mean-square error estimators, which provide improvements over spectral subtraction estimators. Second, estimation of the auditory masked thresholds and masking spreading functions are adjusted to address elevated thresholds and broader auditory filters characteristic of cochlear hearing loss. In addition to algorithm details, results of the algorithm evaluation (using objective quality measures; intelligibility and quality measures in hearing-impaired listeners) will be presented. [Work supported by the Whitaker Foundation.]

Session 4aPPb

Psychological and Physiological Acoustics: Pitch, Speech Perception and Loudness

Craig A. Champlin, Chair

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Contributed Papers

8:15

4aPPb1. Pitch discrimination as a function of the inter-stimulus interval: Evidence against a simple model of perceptual memory. Laurent Demany, Gaspard Montandon, and Catherine Semal (CNRS and Univ. Victor Segalen, 146 rue Leo Saignat, F-33076 Bordeaux, France, Laurent.Demany@psyac.u-bordeaux2.fr)

A listener's ability to compare two sounds separated by a silent time interval T is limited by a sum of ''sensory noise'' and ''memory noise.'' The present work was intended to test a model according to which these two components of internal noise are independent and, for a given sensory continuum, the memory noise depends only on T. In three experiments using brief sounds (<80 ms), pitch discrimination performances were measured in terms of d' as a function of T (0.1–4 s) and a physical parameter affecting the amount of sensory noise (pitch salience). As T increased, d' first increased rapidly and then declined more slowly. According to the tested model, the relative decline of d' beyond the optimal value of T should have been slower when pitch salience was low (large amount of sensory noise) than when pitch salience was high (small amount of sensory noise). However, this prediction was disproved in each of the three experiments. It was also found, when a ''roving'' procedure was used, that the optimal value of T was markedly shorter for very brief tone bursts (6 sine cycles) than for longer tone bursts (30 sine cycles).

8:30

4aPPb2. *F*0 discrimination interference: Effects of resolved tone complexes and noise on fundamental frequency discrimination of unresolved complex tones. Hedwig Gockel, Robert P. Carlyon (MRC Cognition and Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 2EF, UK, hedwig.gockel@mrc-cbu.cam.ac.uk), and Christopher J. Plack (Univ. of Essex, Wivenhoe Park, Colchester CO4 3SQ, UK)

$F0$ discrimination of a 400-ms complex tone with only unresolved components (''target'') was investigated in the absence and presence of a synchronously gated resolved complex tone (''interferer''). The target and the interferer were bandpass filtered from 1375–15 000 Hz and 125–625 Hz, respectively. In a 2I-2AFC task, listeners indicated the interval containing the target with the higher pitch. The nominal $F0$ of the target was 88 Hz; that of the interferer was constant across the two intervals and was either 88 Hz or increased by various amounts. Although the target and interferer were in well-separated frequency regions, performance (percent correct) dropped by about 16% when the interferer's $F0$ was 88 Hz. The impairment was halved when the interferer's $F0$ was 10% higher than that of the target, and almost eliminated when it was 30% higher. Increasing the level of a 1375-Hz low-pass-filtered noise, gated synchronously with the target and the interferer ($F0$ equaled 88 Hz), improved performance, further demonstrating that the deterioration produced by the resolved complex was not due to peripheral masking. The results are consistent with a form of across-frequency interference at the level of pitch perception. [Work supported by EPSRC Grant GR/R65794/01.]

8:45

4aPPb3. Effects of relative frequency, absolute frequency, and phase on fundamental frequency discrimination: Data and an autocorrelation model. Joshua Bernstein and Andrew Oxenham (MIT Res. Lab. of Electronics and Harvard-MIT Speech & Hearing Bioscience & Technol. Prog., 77 Massachusetts Ave., Cambridge, MA 02139, jgbern@mit.edu)

Fundamental frequency ($F0$) difference limens (DLs) were measured versus $F0$ for sine- and random-phase harmonic complexes bandpass-filtered into low- or high-frequency regions, with 3-dB passbands of 2.5–3.5 and 5–7 kHz, respectively. In all cases, $F0$ DLs decreased dramatically with increasing $F0$ as approximately the tenth harmonic appeared in the passband. Generally, $F0$ DLs were similar in both frequency regions for complexes with similar harmonic numbers and phase relationships. However, $F0$ DLs were larger in the high-frequency than the low-frequency region for random-phase complexes containing only harmonics above the tenth, suggesting a possible role for additional fine-structure information in the low-frequency region. The dependence of $F0$ discrimination on relative frequency presents a significant challenge to autocorrelation (AC) models of pitch perception, in which predictions generally depend more on absolute frequency and phase locking. To represent this relative frequency effect, a ''lag window'' modification to the Meddis and O'Mard [J. Acoust. Soc. Am. **102**, 1811–1820 (1997)] AC model was introduced, restricting each channel's AC representation to a limited range of lags relative to the center frequency (CF). Thus in each channel, the AC model responds best to $F0$'s whose dominant harmonics fall near the CF. [Work supported by NIH Grant Nos. 5T32-DC-00038 and R01-DC-05216.]

9:00

4aPPb4. Evidence for distinct types of "perfect pitch." David A. Ross (Dept. of Diagnostic Radiol., Yale School of Medicine, Box 208043, New Haven, CT 06520), John C. Gore (Vanderbilt Univ. Medical Ctr., Nashville, TN XX372), and Lawrence E. Marks (John B. Pierce Lab., New Haven, CT 06519)

The ability to identify and reproduce sounds of specific frequencies, typically called ''perfect pitch,'' is remarkable and uncommon. Whether this skill is learned early in life or inherited has been a matter of great controversy. Further, a substantial literature suggests that ''perfect pitch'' may be heterogeneous. Previously, we proposed a model to account for heterogeneity. The model subdivides individuals capable of naming notes accurately into two groups: possessors of true absolute pitch (AP), who automatically encode the frequency of all tonal stimuli, precategorically and independent of their source; and possessors of heightened tonal memory (HTM), who identify tones by comparing them to a memorized tonal template. The ability of individuals with HTM to identify tonal stimuli should depend strongly on the tones' acoustical properties, such as timbre or chroma. Three experiments sought to test this hypothesis directly. Individuals claiming ''perfect pitch'' were recruited and initially classified as having AP or HTM. Consistent with the model, the two groups differed significantly in their sensitivity to the targets' timbre, chroma, and tonal context, suggesting that they may use different mecha-

nisms to identify tonal stimuli. The model may help reconcile the long-standing controversy between early learning and genetic theories of ''perfect pitch.''

9:15

4aPPb5. Effects of real and illusory glides on pure-tone frequency discrimination. Johannes Lyzenga, Robert P. Carlyon (MRC Cognition and Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 2EF, UK), and Brian C. J. Moore (Univ. of Cambridge, Cambridge CB2 3EB, UK)

Pure-tone frequency difference limens (DLs) were measured for 500-ms, 4-, and 6-kHz pure tones, where the interval between the initial and final constant-frequency portions of the tones in each 2IFC trial consisted of either a silent gap, a frequency glide, or a noise burst. The noise was inserted to create the illusion of the tone continuing through the gap. The interval between the two constant-frequency portions was 0, 10, 50, or 200 ms; a condition with a 500-ms frequency glide without constant-frequency portions was also used. To prevent subjects from using information from the end point of the glides, a frequency rove of approximately four ERBs was used in all conditions. It was found that DLs were lower for the glide than for the gap condition, supporting the conclusion that the auditory system contains a mechanism specific for the detection of dynamic frequency changes [A. Sek and B. C. J. Moore, J. Acoust. Soc. Am. **106**, 351–359 (1999)]. For a number of subjects, the DLs were smaller for the noise condition than for the gap condition, suggesting that this mechanism may be able to operate on a stage of processing at which an illusory glide has been introduced.

9:30

4aPPb6. Comparison of behavioral discrimination, MMN, and P300 to speech and nonspeech stimuli. Joanna Webster, Ashley Harkrider, and Mark Hedrick (Dept. of Audiol. and Speech Pathol., Univ. of Tennessee, 457 South Stadium Hall, Knoxville, TN 37996)

Our objective is to examine the relation between central auditory processes and discrimination of speech (consonant–vowel) and nonspeech (frequency glide) stimuli. Behavioral responses and auditory evoked potentials (MMN and P300) of ten adults were evaluated to synthetically generated consonant–vowel (CV) speech and nonspeech contrasts. The CVs were two within-category stimuli and the nonspeech stimuli were two frequency glides whose frequencies matched the formant transitions of the CV stimuli. It was found that listeners exhibited significantly better behavioral discrimination to the nonspeech versus speech stimuli in same/different and oddball behavioral paradigms. MMN responses were present in all subjects to both stimulus contrasts, and were not significantly different with regard to stimulus type. P300's were present in nine of ten subjects to both stimulus contrasts. However, the CV speech contrasts produced P300's with significantly smaller amplitudes and longer latencies than those to the nonspeech stimuli. These results suggest that the stimuli were processed differently when measured behaviorally and with the P300, but not when measuring the MMN. The enhanced discrimination of the frequency glide stimuli versus the CV stimuli of analogous acoustical content supports the idea that different levels of processing mediate the auditory perception of speech versus nonspeech stimuli.

9:45

4aPPb7. Phoneme contrasts with two, same-talker speakers. Mark Ericson (2610 Seventh St., Wright–Patterson AFB, OH) and Pamela Mishler (Dept. of Veteran Affairs, Dayton, OH)

People often have difficulty hearing speech in the presence of concurrent conversations. This well-known cocktail-party effect can be parsed into energetic and informational masking effects. The purpose of this study was to measure and model the effects of energetic and informational masking that occur when two words, spoken by the same talker are heard at the same time. The word identification test used in the experiments was the Modified Rhyme Test (MRT). The MRT was used as both the stimulus and the masker, which afforded a multitude of consonant contrasts. The phrases were presented monaurally at 75 dB SPL over Sennheiser HD-520 headphones to four normal hearing listeners. The independent variables included 30 pairs of MRT word lists spoken by three male and three female talkers. The dependent variable was the percent correct identifications of the two consonants. Listeners performed at 94% correct for the first word choice and 72% correct for the second word choice. The distribution of errors was analyzed by place of articulation, manner of articulation, and speech-to-speech-ratio of the phoneme pairs and compared to articulation index predictions for speech intelligibility.

10:00–10:15 Break

10:15

4aPPb8. Tactual displays of consonant voicing to supplement speechreading. Hanfeng Yuan, Charlotte M. Reed, and Nat Durlach (Res. Lab. of Electron., MIT, Rm. 36-757, 77 Massachusetts Ave., Cambridge, MA 02139)

This research is concerned with the development of tactual displays of voicing to supplement speechreading in persons with profound hearing impairment. The voicing cue was based on the onset-time difference between amplitude envelopes derived from two different filtered bands of speech (a low-pass band at 350 Hz and a high-pass band at 3000 Hz). This envelope-onset-asynchrony cue was presented through a multifinger tactual stimulator. The amplitude envelopes of the low- and high-frequency bands were used to modulate the amplitude of a 50-Hz sinusoid presented to the thumb and a 250-Hz sinusoid presented to the index finger (respectively). Acoustic measurements of envelope-onset asynchrony in CVC syllables indicate that this timing difference provides a reliable and robust cue for voicing. Perceptual measurements were made of the ability to discriminate (1) temporal-onset order for sinusoidal signals (fixed in frequency but roving in amplitude and duration); and (2) eight pairs of consonant contrasts in CVC syllables under conditions of speechreading alone, tactual cue alone, and the combined condition. Relations among the results for the two perceptual tasks, as well as for the associated acoustic measurements, will be discussed. [Work supported by NIH/NIDCD.]

10:30

4aPPb9. Effects of pitch, level, and tactile cues on speech segregation. Rob Drullman and Adelbert W. Bronkhorst (TNO Human Factors, P.O. Box 23, 3769 ZG Soesterberg, The Netherlands, drullman@tm.tno.nl)

Sentence intelligibility for interfering speech was investigated as a function of level difference, pitch difference, and presence of tactile support. A previous study by the present authors [J. Acoust. Soc. Am. **111**, 2432–2433 (2002)] had shown a small benefit of tactile support in the speech-reception threshold measured against a background of one to eight competing talkers. The present experiment focused on the effects of informational and energetic masking for one competing talker. Competing speech was obtained by manipulating the speech of the male target talker (different sentences). The PSOLA technique was used to increase the average pitch of competing speech by 2, 4, 8, or 12 semitones. Level differences between target and competing speech ranged from −16 to +4 dB. Tactile support (B&K 4810 shaker) was given to the index finger by presenting the temporal envelope of the low-pass-filtered speech (0–200 Hz). Sentences were presented diotically and the percentage of correctly perceived words was measured. Results show a significant overall increase in

intelligibility score from 71% to 77% due to tactile support. Performance improves monotonically with increasing pitch difference. Louder target speech generally helps perception, but results for level differences are considerably dependent on pitch differences.

10:45

4aPPb10. Phase effects in forward masking for normally hearing and hearing-impaired subjects. Brian C. J. Moore, Thomas Stainsby, and Esme Terasewicz (Dept. of Exp. Psych., Univ. of Cambridge, Downing St., Cambridge CB2 3EB, UK)

The forward masking produced by harmonic complex tones depends on the phases of the masker components; phases giving ''peaky'' waveforms on the basilar membrane result in less forward masking than phases giving less peaky waveforms. This difference has been attributed to the effects of peripheral compression and suppression, which depend on the active mechanism in the cochlea. Hence, no phase effect would be expected for subjects with moderate cochlear hearing loss. Growth-of-masking functions were measured for forward maskers containing the first 40 harmonics of a 100-Hz fundamental, with components added in cosine (C) or random (R) phase. The signal frequency was 1 or 2 kHz. For three normally hearing subjects, the R masker produced considerably more masking than the C masker, at high masker levels. Of four subjects with moderate cochlear hearing loss, one showed no effect of masker phase, but for the other three there was a significant effect for at least one signal frequency. It is suggested that peripheral compression and suppression play a role in the phase effect, but other factors, possibly depending on the operation of the efferent system, are involved. [Work supported by MRC (UK).]

11:00

4aPPb11. Role of short-term memory in loudness comparisons. Wolfgang Ellermeier (Dept. of Acoust., Fredrik Bajers Vej 7 B5, DK-9220 Aalborg Ost, Denmark) and Birgit Werner (Boston Univ., Boston, MA 02215)

In an earlier study of the auditory discrimination of time-varying noise bursts [Ellermeier and Schrödl, J. Acoust. Soc. Am. **108**, 2596 (2000)], listeners were found to place greater weight on the beginning and end of the sounds than on the middle portion. To investigate whether this outcome is due to primacy and recency effects in short-term memory which tend to be sensitive to manipulations of the inter-stimulus interval (ISI), the ISI separating the two noise bursts in a 2IFC task was varied systematically. Six participants performed loudness comparisons on 1-s samples of white noise randomly changing in level every 100 ms. In different blocks of trials, the two noise bursts to be compared were either separated by a 500-ms or a 2-s ISI. COSS analysis [Berg, J. Acoust. Soc. Am. **86**, 1743–1746 (1989)] of the overall loudness judgments revealed elevated weights for the beginning and end of the noises, as in the earlier study. These weighting patterns were largely unaffected by the manipulation of ISI, suggesting that the temporal weights found characterize loudness integration in general, and are not just due to idiosyncrasies of the timing used in the 2IFC procedure.

11:15

4aPPb12. Objective limits for four varieties of putative loudness adaptation. Ernest M. Weiler (Psychoacoustics Lab., CSD, CAHS, ML #394, Univ. of Cincinnati, Cincinnati, OH 45267-0394, ernest.weiler@uc.edu), Hongwei Dou, Joel S. Warm, and David E. Sandman (Univ. of Cincinnati, Cincinnati, OH 45267-0394)

The four varieties included three monaural techniques: (1) tone decay (TD); (2) simple adaptation (SA); (3) ipsilateral comparison paradigm (ICP); and the binaural (4) simultaneous dichotic loudness balances (SDLB). ''Loudness adaptation'' indicates that over time, under some conditions, there is a perceived decrease in loudness, when the initial baseline stimulus is progressively assessed. The authors have found the following limits: (1) the classic TD occurred within about 30 dB of threshold for all values tested (250 to 8000 Hz). (2) Except near threshold, SA for the loudness of a continuous unmodulated tone was observed at or above 40 dB when the stimulus reached 6000 Hz, or more. (3) ICP adaptation, which depends on at least 5-s intensity modulation, was found at all values tested from 40 to 80 dB, and from 250 to 8000 Hz. It correlates significantly with TD, with suppression of transient evoked otoacoustic emissions, and at 8000 Hz with SA adaptation. (4) Binaural SDLB adaptation has been repeatedly found at 20 to 100 dB, at all frequencies tested. Factor analysis indicated that binaural SDLB has at least four component factors, including binaural interaction. Unless factors are separated, SDLB adaptation does not correlate with monaural adaptation.

11:30

4aPPb13. Cancelled harmonics—How high does the effect go? Matthew J. Goupell, Peter Xinya Zhang, and William M. Hartmann (Phys. and Astron., 4230 BPS Bldg., Michigan State Univ., East Lansing, MI 48824)

Demonstration Number 1 in the IPO-NIU-ASA collection of auditory demonstrations (compact disk) periodically cancels and reinserts a harmonic of a complex tone having a 200-Hz fundamental and 20 equal-amplitude harmonics. This procedure causes a listener to hear out the manipulated harmonic as a separate tone. In this way, the demonstration exposes harmonics 1 through 10. The following question arises: What is the highest harmonic that can be made audible, and what is responsible for the limitation? Listening experiments, using random harmonic phases, fundamental frequencies (f_0) from 50 to 2000 Hz, and a maximum harmonic frequency of 20 kHz, show that for high fundamental frequencies ($f_0 > 200$ Hz) the highest audible harmonic *frequency* is insensitive to f_0 and is only about 10 percent less than the highest audible sine frequency in quiet. For lower fundamental frequencies, the highest audible harmonic *number* tends to be insensitive to f_0 and is 50–70 for normal hearing listeners. In this region the highest audible harmonic number can be predicted from noise-masked threshold data, but with a large uncertainty. [Work supported by NIDCD.]

11:45

4aPPb14. Underwater loudness for pure tones: Duration effects. Edward A Cudahy, Derek Schwaller, David Fothergill, and Keith Wolgemuth (Naval Submarine Medical Res. Lab., Box 900, Groton, CT 06349-5900)

The loudness of underwater pure tones was measured by loudness matching for pure tones from 100 to 16,000 Hz. The standard was a one second tone at 1000 Hz. The signal duration was varied from 20 milliseconds to 5 seconds. Subjects were instructed to match the loudness of the comparison tone at one of the test frequencies to the loudness of the standard tone. Loudness was measured at the threshold, the most comfortable loudness, and the maximum tolerable loudness. The intensity of the standard was varied randomly across the test series. The subjects were bareheaded U.S. Navy divers tested at a depth of 3 meters. All subjects had normal in-air hearing. Tones were presented to the right side of the subject from an array of underwater sound projectors. The sound pressure level was calibrated at the location of the subject's head with the subject absent. Loudness increased and threshold decreased as duration increased. The effect was greatest at the lowest and highest frequencies. The shape of the loudness contours across frequency and duration derived from these measurements are different from in-air measurements. [Research supported by ONR.]

Session 4aSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration

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Contributed Papers

9:00

4aSA1. Radiation efficiency of a fluid-loaded rib-stiffened finite cylindrical shell. Y. F. Hwang and W. K. Bonness (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804-0030)

Due to the difficulty of computing the free modes of heavily fluid-loaded structures, radiation efficiency of such structures has been frequently calculated from the forced vibratory responses, which are usually contaminated with the direct radiation of the force field. This paper discusses two methods of calculating the radiation efficiency of a fluid-loaded rib-stiffened finite cylindrical shell. Solutions in both methods are obtained from Fourier series expansion of fluid loading and rib dynamic stiffness in terms of the shell *in vacuo* modes. In the first method, the forced radiated power and the spatial mean-square vibratory velocity are calculated excluding the supersonic elements of the forcing function. In the second method, free modes of the fluid-loaded and rib-stiffened shell are calculated from the coupled equations of motion in terms of *in vacuo* modes as generalized coordinates. From this, the modal radiated power and the modal spatial mean-square velocity are calculated. Radiation efficiency is determined from the ratio between the radiated power and the product of rc, surface area, and the spatial mean-square vibratory velocity. Calculated results from the two methods are discussed. Membrane waves' contributions to the radiation efficiency are also discussed. [Work supported by ONR, Code 333.]

9:15

4aSA2. Vibration of fluid-loaded hemi-prolate spheroidal shells. Jeffrey E. Boisvert (Naval Undersea Warfare Ctr., Newport, RI 02841, boisvertje@npt.nuwc.navy.mil) and Sabih I. Hayek (Penn State Univ., University Park, PA 16802)

The equations of motion for nonaxisymmetric vibration of hemi-prolate spheroidal shells of constant thickness were derived using Hamilton's principle. The shell is clamped at the equator and is excited by mechanical surface force fields. The shell theory used in this derivation includes shear deformations and rotatory inertias. The displacements and rotations were expanded in an infinite series of comparison functions. The shell is fluid-filled and is submerged in an infinite fluid medium. The external and internal fluid loading impedances were computed using expansions of prolate spheroidal wavefunctions in each domain. The dynamic response of the fluid-loaded shell was determined using an axisymmetric normal surface force as the excitation input. Numerical results were obtained for the driving and transfer mobilities for several shell thickness-to-length ratios ranging from 0.005 to 0.1, and for various shape parameters, "a," ranging from an elongated hemi-spheroidal shell ($a=1.01$) to a hemispherical shell ($a=100$). Results are presented for various combinations of external and internal fluid loading, and comparisons are made to the *in-vacuo* shell vibration. [Work supported by ONR and the Navy/ASEE Summer Faculty Program.]

9:30

4aSA3. Chaotic properties and eigenfrequency distribution of Semi-Stadium 2-D Fields. Mikio Tohyama, Yoh-ichi Fujisaka, and Kazuaki Yoshida (2665-1, Nakano-machi Hachioji-shi, Tokyo 192-0015, Japan, ctdeneuv@sin.cc.kogakuin.ac.jp)

This paper describes the eigenfrequency distribution of semi-stadium 2-D fields whose geometrical figure is formed by rectangular and hyper-circular segments. The sound ray propagation in the fields exhibits chaotic properties when order n of the hyper-circular parts defined by $x^n+y^n=r^n$ decreases to 2. The chaotic behavior can be estimated according to the degree of freedom of the gamma distribution that represents the eigenfrequency spacing statistics and the correlation dimensions of the chaotic structure. The experimental and simulation results of the 2-D-membrane and plate vibration analysis show that the degree of freedom of the gamma distribution changes from 1 to 2 including nonintegers as the boundary of the 2-D space changes from rectangular (n: infinity, "regular field") to stadium (n: 2, "irregular field"), which corresponds to the change in the correlation dimensions from 1 to 2 for the sound ray propagation in the field. The family of gamma distributions includes the Wigner distribution, which Lyon [J. Acoust. Soc. Am. **45**, 545–565 (1969)] assumed for irregularly shaped boundaries as the case where the degree of freedom is 2.

9:45

4aSA4. Development of a new spectral energy formulation based on structural driving point mobilities or impedances. Seungbo Kim and Raj Singh (Acoust. and Dynam. Lab., The Ohio State Univ., 206 W. 18th Ave., Columbus, OH 43210-1107, kim.873@osu.edu)

A new approximate method is proposed here that characterizes harmonic kinetic and potential energies of a system based on the driving point impedance or mobility. The proposed complex-valued formulation is compared with an existing, real-valued dynamic energy approximation method [J. Sound Vib. **217**, 351–386; **247**, 683–702]. The scope of this study is limited to the frequency domain analysis and harmonic excitation is applied to a linear time-invariant (LTI) system. Longitudinal and flexural motions of a finite beam along with some discrete system examples are employed to illustrate the proposed scheme. Our method is based on an alternate interpretation of the associated driving point transfer functions and it approximates total time-averaged harmonic kinetic and potential energies. Numerical results show that our method yields a more accurate estimate than the existing method. Further, the proposed scheme is found to be insensitive to the driving point measures as consistent results using either impedance or mobility formulations, unlike the existing method, are predicted. Our characterization method is suitable for a moderately damped system at low and mid-frequencies, like the existing method.

10:00

4aSA5. Assessment of a spectral energy formulation based on driving point structural mobilities or impedances. Seungbo Kim and Raj Singh (Acoust. and Dynam. Lab., The Ohio State Univ., 206 W. 18th Ave., Columbus, OH 43210-1107, kim.873@osu.edu)

To determine the contributions of parallel paths or interactions between structural components at any frequency, dynamic kinetic and potential energies are needed. However, an exact method to characterize an arbitrarily damped system in kinetic and potential energy forms is currently not available. One approximate characterization method examines time-averaged dynamic energies of subsystem(s) via the driving point transfer functions [J. Sound Vib. **217**, 351–386 and **247**, 683–702]. This concept, based on a real-valued energy formulation, is critically examined here. Linear time-invariant (LTI) system is assumed and harmonic excitation is applied to the system. Finite continuous (bar and beam) structures along with discrete system examples are utilized to illustrate concepts. Predictions show that the energy estimates deviate from the exact values, especially at higher frequencies and with higher damping. Further, the results from the impedance formulation differ from the ones from the mobility representation. The cause of discrepancies is explained by interpreting the transfer functions involved.

10:15

4aSA6. How can one tell if noisy frequency response data contain a modal contribution? Christopher W. Moloney and Jerry H. Ginsberg (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr., Atlanta, GA 30332-0405, gtg824d@prism.gatech.ed)

A key question that arises in some techniques for experimental modal analysis is whether coherent modal information is buried within a frequency band of complex frequency response data. This question is especially problematic if the data are contaminated by a substantial level of noise. Furthermore, some identification algorithms introduce small anomalies to the data in the course of processing. To explore techniques for resolving this issue frequency response data for a one-degree-of-freedom system is generated analytically and contaminated with various levels of white noise. Five statistical measures are computed for a sequence of bands that are obtained by forming overlapping halves of the previous frequency bands. The five measures are the expected value, mean-square, and variance of the frequency response in each band, as well as eigenanalysis of the autocorrelation matrix and wavelet decomposition. Graphical results are presented and discussed, and an analytical assessment method is proposed.

10:30

4aSA7. A comparison of the rational fraction polynomial method and the algorithm of mode isolation for fitting noisy data. Matthew S. Allen and Jerry H. Ginsberg (Georgia Inst. of Technol., Atlanta, GA 30332-0405)

A multitude of methods exist for fitting linear, modal models to mechanical system response data. Many of these methods involve rearranging the modal equations of motion of the system so that a linear-least-squares solution is possible. Some methods require simplifying assumptions, such as light damping, while most popular methods are exact for clean data. Because the overall process is nonlinear, two different exact, linear-least-squares methods, can give different results for noisy data. This work compares two methods which are exact for linear, viscous, state-space (or nonproportionally damped) systems, the well known Rational Fraction Polynomial Method (RFP) and the Algorithm of Mode Isolation (AMI). It is shown that while RFP performs exceptionally for clean data, it is much less robust than AMI for noisy data. The performance of both algorithms is compared when applied to noise contaminated analytical data for a multi-degree of freedom frame structure. The frame can be tuned so that high damping ratios and heavy modal coupling are present. [Work supported under a National Science Foundation Graduate Research Fellowship.]

10:45

4aSA8. Experiments and interpretation of surface waves acoustically generated on spherically endcapped cylindrical shells. A. Claude Ahyi, Hui Cao, P. K. Raju (Dept. of Mech. Eng., Auburn Univ., Auburn, AL 36849), and H. Überall (Catholic Univ. of America, Washington, DC 20064)

The impact of an acoustic pulse on a submerged elastic shell (which we shall assume evacuated) generates three types of circumferentially propagating surface waves: those that are analogous to plate waves of type $A_0(A_1,A_2,\ldots)$ and $S_0(S_1,S_2,\ldots)$, and a Scholte–Stoneley wave of type A that propagates in the surrounding fluid. A computer program devised by Hui Cao *et al.* renders visualizations of the generation and propagation of circumferential pulses on spherically endcapped shells, visualized by the re-radiation into the surrounding fluid of A, A_0, and S_0 waves in sequential pictures. These surface waves are generated experimentally by us in the laboratory from an ultrashort-pulse source, at axial incidence on a hemispherically endcapped glass tube. Sequential visualizations of the re-radiated pulses are obtained by using the Schardin–Cranz Schlieren method. These observations lead to an experimental measure of the group velocity dispersion curves of the surface waves.

11:00

4aSA9. Phonon localization methods through time and space: Experimental SAW phonon aspects on a cylindrical shell. Loic Martinez (ECIME IUP GE, 5 mail Gay-Lussac, F 95 031 Neuville sur Oise Cedex, France), Bruno Morvan, and Jean Louis Izbicki (Universit du Havre, 76610 Le Havre, France)

A transient experimental study of Surface Acoustic Wave (SAW) propagation along a 1D medium as a function of time leads to a 2D space–time signal collection. Previous studies have shown that 3D space-wave number–frequency representation $S(x,k,f)$ allows the characterization of the space transient aspects of SAW generation [L. Martinez, J. Acoust. Soc. Am. **105**, 952 (1999)]. In order to analyze the time transient aspect of phonon propagation, the 3D time–wave Number-Frequency representation $Z(t,k,f)$ is proposed. The $Z(t,k,f)$ matrix is obtained by short time Fourier transforming each time signal of the 2D time–wave number representation. The $Z(t,k,f)$ representation is used to experimentally investigate SAW generation and propagation around a cylindrical shell (the relative thickness is equal to 0.03) surrounded by water and excited by a pulse (0.1 s duration). For the air filled shell, the $Z(t,k,f=\text{const})$ slices show the time sequence of the continuous flow of incident phonons striking the insonified side of the shell and their reflection or conversion in SAW phonons. The complex frequencies, direction of propagation and the time origin of all the phonons are identified. For the first time, the elastic Franz waves due to diffraction around the shell are also observed. For the water-filled shell the $Z(t,k,f=\text{const})$ slices point out two new major features. On the one hand, phonons trapped inside the fluid column of the shell are clearly identified. On the other hand, when the initial incidence angle enables the generation of a SAW, the SAW is generated at each internal striking point. A precise experimental ray model is deduced from these results, allowing us to link this transient experimental study with the theoretical results based on a steady state approach.

Session 4aSC

Speech Communication: Acoustical and Perceptual Characteristics of Special Speech Registers

Sarah H. Ferguson, Chair
Department of Speech, Language, Hearing Sciences and Disorders, University of Kansas, 3001 Dole Center, 1000 Sunnyside Avenue, Lawrence, Kansas 66045

Chair's Introduction—8:30

Invited Papers

8:35

4aSC1. Production and perception of clear speech. Ann R. Bradlow (Dept. of Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, abradlow@northwestern.edu)

When a talker believes that the listener is likely to have speech perception difficulties due to a hearing loss, background noise, or a different native language, she or he will typically adopt a clear speaking style. Previous research has established that, with a simple set of instructions to the talker, ''clear speech'' can be produced by most talkers under laboratory recording conditions. Furthermore, there is reliable evidence that adult listeners with either impaired or normal hearing typically find clear speech more intelligible than conversational speech. Since clear speech production involves listener-oriented articulatory adjustments, a careful examination of the acoustic-phonetic and perceptual consequences of the conversational-to-clear speech transformation can serve as an effective window into talker- and listener-related forces in speech communication. Furthermore, clear speech research has considerable potential for the development of speech enhancement techniques. After reviewing previous and current work on the acoustic properties of clear versus conversational speech, this talk will present recent data from a cross-linguistic study of vowel production in clear speech and a cross-population study of clear speech perception. Findings from these studies contribute to an evolving view of clear speech production and perception as reflecting both universal, auditory and language-specific, phonological contrast enhancement features.

9:00

4aSC2. Elderspeak. Susan Kemper (3090 Dole, Gerontology Ctr., 1000 Sunnyside, Univ. of Kansas, Lawrence, KS 66045)

Elderspeak has been assumed to be an accommodation to the perceived communication needs of older adults as it involves a slow rate of speaking, simplified syntax, vocabulary restrictions, and exaggerated prosody. It also is judged to be patronizing and disrespectful in that its use presumes that the older adult is cognitively impaired. Using a controlled referential communication task, my colleagues and I have investigated elderspeak addressed by young adults to older adults. Our studies suggest that the speech register is composed of two sets of parameters. One set of parameters is linked to the perception that the older listener is cognitively impaired; these parameters affect how much information is conveyed and include semantic elaborations such as expansions and repetitions of previous map directions. The other set of parameters includes fluency, prosody, and grammar. The modifications to fluency and prosody do not appear to benefit older listeners. In these studies, the young adults' use of elderspeak did improve the performance of the older listeners on the referential communication task. The use of elderspeak by the young partners appeared to trigger older adults' perceptions of themselves as cognitively impaired, consistent with the ''communicative predicament of aging'' model of Ryan *et al.* (1986).

9:25

4aSC3. An acoustic comparison of two women's infant- and adult-directed speech. Jean Andruski and Shiri Katz-Gershon (Audiol. & Speech-Lang. Pathol., Wayne State Univ., 581 Manoogian Hall, Detroit, MI 48201)

In addition to having prosodic characteristics that are attractive to infant listeners, infant-directed (ID) speech shares certain characteristics of adult-directed (AD) clear speech, such as increased acoustic distance between vowels, that might be expected to make ID speech easier for adults to perceive in noise than AD conversational speech. However, perceptual tests of two women's ID productions by Andruski and Bessega [J. Acoust. Soc. Am. **112**, 2355] showed that is not always the case. In a word identification task that compared ID speech with AD clear and conversational speech, one speaker's ID productions were less well-identified than AD clear speech, but better identified than AD conversational speech. For the second woman, ID speech was the least accurately identified of the three speech registers. For both speakers, hard words (infrequent words with many lexical neighbors) were also at an increased disadvantage relative to easy words (frequent words with few lexical neighbors) in speech registers that were less accurately perceived. This study will compare several acoustic properties of these women's productions, including pitch and formant-frequency characteristics. Results of the acoustic analyses will be examined with the original perceptual results to suggest reasons for differences in listener's accuracy in identifying these two women's ID speech in noise.

9:50

4aSC4. Predicting children's hyperarticulate speech during human-computer error resolution. Sharon Oviatt, Rachel Coulston (Dept. of Computer Sci. & Eng., Oregon Health & Sci. Univ., Beaverton, OR 97006, oviatt@cse.ogi.edu), and Courtney Darves (Psych. Dept., Univ. of Oregon)

When speaking to interactive systems, people sometimes *hyperarticulate*—or adopt a clarified form of speech that has been associated with increased recognition errors. The goal of the present study was to provide a comprehensive assessment of the type and magnitude of linguistic adaptations in children's speech during human-computer error resolution, and to compare these adaptations with those typical of adult hyperarticulation. A study was conducted in which twenty-four 7- to- 10-year-old children interacted with a simulated conversational system, which permitted a comparison of their verbatim repetitions immediately before and after system recognition errors. Matched original-repeat utterance pairs then were analyzed for acoustic, prosodic, and phonological adaptations. Like adult speech, the primary hyperarticulate changes in children's speech included durational phenomena such as lengthening of pauses and the speech segment, and a more deliberate, hyper-clear articulatory style. However, children's speech also displayed large increases in amplitude that are not typical of adult hyperarticulation, as well as substantially larger magnitude adaptations than those observed in adult speech. These results corroborate and generalize the Computer-elicited Hyperarticulate Adaptation Model, and have implications for improved error handling in next-generation spoken language and multimodal systems. [Work supported by NSF Grant No. IIS-0117868.]

10:15–10:30 Break

Contributed Papers

10:30

4aSC5. Does an infant-directed speaking style aid in the separation of different streams of speech? Rochelle S. Newman (Dept. of Hearing & Speech Sci. & Prog. in Neurosci. & Cognit. Sci., Univ. of Maryland, College Park, MD 20742, rnewman@hesp.umd.edu), Tammy Weppelman, and Isma Hussain (Dept. of Psych., Univ. of Iowa)

This work explores whether an infant-directed speaking style (IDS) may be easier to separate from background speech noise than is adult-directed speech (ADS). Many of the acoustic cues found in infant-directed speech are similar to ones shown to be important in adult stream segregation (such as differences in voice pitch and pitch variability). In addition, using an infant-directed speech style may serve to make the talker's voice more dissimilar from any background speech (which is likely to be adult-directed). We explored this issue in three different ways: by examining adults' ability to separate two different streams of speech varying in register, by examining whether adult speakers use IDS to a greater extent when in the context of noise, and by examining whether infants' preference for IDS over ADS might be greater in the context of background noise. Implications of these three lines of work will be discussed. [Work supported by NSF and NICHD.]

10:45

4aSC6. Acoustic characteristics of listener-constrained speech. Simone Ashby and Fred Cummins (Dept. of Computer Sci., Univ. College Dublin, Belfield, Dublin 4, Ireland, simone.ashby@ucd.ie)

Relatively little is known about the acoustical modifications speakers employ to meet the various constraints—auditory, linguistic and otherwise—of their listeners. Similarly, the manner by which perceived listener constraints interact with speakers' adoption of specialized speech registers is poorly understood. Lindblom's Hyper & Hypo (H&H) theory offers a framework for examining the relationship between speech production and output-oriented goals for communication, suggesting that under certain circumstances speakers may attempt to minimize phonetic ambiguity by employing a "hyperarticulated" speaking style (Lindblom, 1990). It remains unclear, however, what the acoustic correlates of hyperarticulated speech are, and how, if at all, we might expect phonetic properties to change respective to different listener-constrained conditions. This paper is part of a preliminary investigation concerned with comparing the prosodic characteristics of speech produced across a range of listener constraints. Analyses are drawn from a corpus of read hyperarticulated speech data comprising eight adult, female speakers of English. Specialized registers include speech to foreigners, infant-directed speech, speech produced under noisy conditions, and human-machine interaction. The authors gratefully acknowledge financial support of the Irish Higher Education Authority, allocated to Fred Cummins for collaborative work with Media Lab Europe.

11:00

4aSC7. Very loud speech over simulated environmental noise tends to have a spectral peak in the $F1$ region. Sten Ternstrom, Mikael Bohman (Speech Music & Hearing, Kungliga Tekniska Hoegskolan, Drottning Kristinas Vaeg 31, SE-100 44 Stockholm, Sweden, sten@speech.kth.se), and Maria Sodersten (Huddinge Univ. Hospital, SE-141 86 Stockholm, Sweden)

In some professions, workplace noise appears to be a hazard to the voice, if not to hearing. Several studies have shown that teachers and sports instructors, for example, are more prone to voice problems than average, prompting research on loud voice. Since on-location recordings are in many ways impractical, the running speech of 23 untrained speaker subjects (12 female, 11 male) was instead recorded in several types of loud noise that was presented over high-quality loudspeakers. Using adaptive cancellation techniques, the noise was then removed from the recordings, thus exposing the strained voices for analysis. The experiment produced a large body of data, only one aspect of which is reported here. In most subjects, the vowel spectrum as a function of voice SPL showed the expected behavior for low to moderate efforts, but developed a very pronounced peak in the $F1$ region at the highest efforts. This peak can be ascribed to the concerted action of several acoustic mechanisms, including source waveform asymmetry, $F1$ approximating one of the lower partials, and increased formant Q values due to a longer closed phase. [Work supported by the Swedish Council for Working Life and Social Research, Contract No. 2001-0341.]

11:15

4aSC8. Effect of speaking styles on the relevance of the Speech Transmission Index in the presence of reverberation. Sander J. van Wijngaarden and Tammo Houtgast (TNO Human Factors, P.O. Box 23, 3769 ZG Soesterberg, The Netherlands)

The Speech Transmission Index (STI) predicts speech intelligibility under various speech degrading conditions, including noise and reverberation. Measurements of sentence intelligibility in noise and reverberation, for three different talkers adopting different speaking styles, indicate that the STI is sometimes inaccurate in the presence of reverberation. For a trained talker speaking clearly, the STI predictions are accurate, but for conversational speech by untrained talkers, the effect of reverberation is underestimated. By using a wider range of modulation frequencies than prescribed for the standardized STI calculation (up to 31.5 Hz instead of 12.5 Hz), more accurate predictions are obtained for conversational speech. This can be understood by comparing the envelope spectrum between talkers and speaking styles: conversational speech tends to spread

more of the energy in its envelope spectrum to higher modulation frequencies. Since higher modulation frequencies are more susceptible to reverberation, conversational speech tends to be affected more by reverberation than clear speech. To investigate the range of between-talker variations, envelope spectra were calculated for a population of 134 talkers. Upper boundaries of the STI modulation frequency range of 12.5 and 31.5 Hz seem approximately appropriate for the 5th and 95th percentile of this population.

THURSDAY MORNING, 1 MAY 2003 ROOM 203, 10:30 A.M. TO 12:00 NOON

Session 4aSP

Signal Processing in Acoustics: Arrays and Beamforming

Alan W. Meyer, Chair
Lawrence Livermore National Laboratory, P.O. Box 808, L-154, Livermore, California 94551

Contributed Papers

10:30

4aSP1. High resolution beamforming for small aperture arrays. Chris Clark, Tom Null (Miltec Res. and Technol., Inc., NCPA, 1 Coliseum Dr., Univ., MS 38677), and Ronald A. Wagstaff (Univ. of Mississippi, MS 38677)

Achieving fine resolution bearing estimates for multiple sources using acoustic arrays with small apertures, in number of wavelengths, is a difficult challenge. It requires both large signal-to-noise ratio (SNR) gains and very narrow beam responses. High resolution beamforming for small aperture arrays is accomplished by exploiting acoustical fluctuations. Acoustical fluctuations in the atmosphere are caused by wind turbulence along the propagation path, air turbulence at the sensor, source/receiver motion, unsteady source level, and fine scale temperature variations. Similar environmental and source dependent phenomena cause fluctuations in other propagation media, e.g., undersea, optics, infrared. Amplitude fluctuations are exploited to deconvolve the beam response functions from the beamformed data of small arrays to achieve high spatial resolution, i.e., fine bearing resolution, and substantial SNR gain. Results are presented for a six microphone low-frequency array with an aperture of less than three wavelengths. [Work supported by U.S. Army Armament Research Development and Engineering Center.]

10:45

4aSP2. Characterization of the left/right bearing ambiguities for passive line arrays. Frank A. Boyle and David E. Grant (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

The symmetry of a straight line underwater acoustic array produces a left/right bearing ambiguity that makes it impossible to determine which side of the array a source is located on. A simple technique for resolving left/right ambiguities is to break the symmetry by curving the array. Under certain common conditions, however, significantly curved arrays still possess left/right ambiguities that depend on the source's range and the beamformer's focal length. These ambiguities can be modeled, given a source and array configuration, to produce a prediction of source localization capability. A map can be generated that indicates the spatial regions where a source is likely to be localized on the incorrect side of the array. The presentation will include a description of how the ambiguities are modeled, and simulated results. Considerations on how curved arrays can be configured to minimize ambiguities will be discussed.

11:00

4aSP3. Multiple speech signal enhancement using a microphone array. Heather E. Ewalt and Michael T. Johnson (Dept. of Elec. and Computer Eng., Marquette Univ., Milwaukee, WI 53223, heather.ewalt@mu.edu)

The ability to extract and enhance a primary speech signal from an environment with multiple speakers is an important issue. While methods exist for a variety of beamforming techniques [M. Brandstein and D. Ward, *Microphone Arrays: Signal Processing Techniques and Applications* (Springer, New York, 2001)] as well as for multi-source filtering in stationary noise [H. Saruwatari *et al.*, "Speech Enhancement Using Nonlinear Microphone Array With Noise Adaptive Complementary Beamforming," Proc. of IEEE ICASSP, pp. 1049–1052 (2000)], the theory has yet to be developed for integrating spatial filtering with additional enhancement methods to deal with non-stationary interference from interfering talkers. This paper presents a novel method for incorporating multiple parallel beamformers with traditional speech enhancement algorithms, particularly the Wiener filter and spectral subtraction. By iteratively improving the spectral magnitude estimates of each speech source, substantial improvement in overall signal separation can be obtained. The performance of the algorithm is illustrated using a simulated multiple speaker environment with resulting SNR and sSNR plots. [Work supported by the GAANN Fellowship.]

11:15

4aSP4. Beamforming for a microphone array embedded in asymmetrically shaped objects. Philippe Moquin, Stéphane Dedieu (Mitel Networks, 350 Legget Dr., Kanata, ON K2K 2W7, Canada), and Rafik Goubran (Carleton Univ., Ottawa, ON K15 5B6, Canada)

Broadband frequency invariant beamforming for circular arrays or linear arrays are quite common but not when they are embedded in a diffracting structure. Meyer [J. Acoust. Soc. Am. **109**, 185–193 (2001)] describes arrays embedded in a diffracting sphere, and provides an analytical solution for the wave equation in acoustics. For arrays of simple shape like circular rings embedded in a more complex shape one must make use of numerical methods (e.g., boundary element methods). Microphone arrays in shapes that are not symmetric or axisymmetric can also be solved this way but result in very asymmetrical beams. One example of such an obstacle is a telephone incorporating a microphone array. This presentation will show results from simulations and measurements of a six-microphone array. A design approach to obtain reasonably well behaved beams relies on constrained optimization, with a constraint build using a set of vectors containing the sensor signal for acoustic waves with specific directions of arrival. [Work supported in part by Carleton University.]

11:30

4aSP5. Inferring array geometry from multiple sources of opportunity. Frank A. Boyle and David E. Grant (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

A beamformer's performance is sensitive to the accuracy of the element location estimates. Array element localization is a reasonably simple task when a source is present with a known location. The problem is more difficult with a source whose position is unknown. Techniques commonly require assumptions to be made regarding the source's range. A method for determining the array's geometry from multiple sources of opportunity has been formulated. Array geometry is determined from multiple observations of parallax from each source, for a collection of hydrophone pairs in the array. The array's geometry is built up from the combined orientations of hydrophone pairs. In its current form, the technique requires accurate knowledge of hydrophone pair separation. The presentation will include an application of the method to simulated array data and a discussion of current limitations.

11:45

4aSP6. Beamforming using indefinite term for arbitrarily located microphones. Hirofumi Nakajima, Yamanaka Takaaki, and Hiroshi Nakagawa (Nittobo Acoust. Eng. Co., Ltd., 1-13-12, Midori, Sumida-ku, Tokyo 130-0021, Japan, nakajima@noe.co.jp)

The design of filter coefficients for a beamforming algorithm constructed of finite impulse response filters and adders was investigated. Designing filter coefficients can be represented as a linear inverse problem, and using the indefinite term of the linear equation, a filter having the smallest sidelobe beam pattern based on the least-square estimation criterion can be designed. This design method is represented by a closed form equation using singular value decomposition and a pseudo-inverse matrix, so it does not need parameter adjustment, as do adaptive methods. It can be used for arbitrarily located microphones; it was applied to a sound visualization system using a spherical microphone array (SMA). At the meeting, this design method and its directivity performance when applied to a SMA will be described.

THURSDAY MORNING, 1 MAY 2003 ROOM 102, 8:00 A.M. TO 12:00 NOON

Session 4aUW

Underwater Acoustics and Acoustical Oceanography: High Frequency Sediment Acoustics

Eric I. Thorsos, Chair

Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, Washington 98105-5598

Chair's Introduction—8:00

Invited Papers

8:05

4aUW1. Underwater sand acoustics: A perspective derived from the sediment acoustics experiment (SAX99). Kevin L. Williams, Eric I. Thorsos, Darrell R. Jackson, Dajun Tang, and Steve G. Kargl (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St, Seattle, WA 98105-6698, williams@apl.washington.edu)

The sediment acoustics experiment (SAX99) included investigations of the following three questions. What are the dominant mechanisms responsible for backscattering from sand sediment? What are the dominant mechanisms responsible for subcritical penetration into sand? What are the appropriate constitutive equations for sand? In this paper a summary is presented of APL-UW SAX99 experiments and data/model comparisons relevant to each question. Perspectives are also given on some of the issues that remain or arose during SAX99 and the associated analyses. In general, these issues are tied to the frequency dependencies seen in the data but not fully captured by present models. For backscattering the issue is that as the frequency increases the measured backscattering strength does not follow predictions based on surface roughness scattering models. In the case of penetration it is a frequency cutoff effect seen in SAX99 buried array data but seemingly violated in the detection of buried objects near the SAX99 site. Regarding the constitutive equations, it is the frequency dependence of the attenuation above 50 kHz. Recent experiments will be described that have been motivated by these issues. Finally, research proposed as part of a follow-on sediment acoustics experiment (SAX04) will be outlined. [Work supported by ONR.]

8:25

4aUW2. Time domain modeling of seafloor scatter at low grazing angle using the small slope approximation. Eric Pouliquen, Lucie Pautet, and Gaetano Canepa (Saclant Undersea Res. Ctr., Viale San Bartolomeo, 400, 19138 La Spezia, Italy, pouliq@saclantc.nato.int)

Sea surface and bottom scattering components are usually quantified only in terms of scattering strength (SS). The SS corresponds to an ensemble averaged plane wave intensity scattered from a unit surface at a unit distance. However for non-steady-state cases when the acoustic footprint approaches the size of the acoustic wavelength, higher moment statistics and probability distribution functions (PDF) provide additional and essential information for detection and classification purposes. As a step toward higher moment scattering prediction the time domain model BORIS [Pouliquen *et al.*, J. Acoust. Soc. Am. **105** (1999)] has been extended to low grazing angles. It uses the fourth-order small slope approximation (SSA-4) and the small perturbation theory for interface and volume scattering, respectively. This bistatic 3-D model accounts for the full sensing geometry and sonar properties. It uses stochastic realizations of the boundaries and volumes with controlled statistics. BORIS-SSA offers numerous possibilities for practical applications. In particular it provides an objective assessment of detection and classification performances for most current sonar systems (conventional and synthetic aperture sonars) operating in realistic and complicated environments (cluttered or patchy). The model principles and simulation results will be presented and illustrated with real sonar data.

8:45

4aUW3. Tempo and scale of biogenic effects on high-frequency acoustic propagation near the marine sediment–water interface in shallow water. Peter Jumars (Darling Marine Ctr., Univ. of Maine, 193 Clark's Cove Rd., Walpole, ME 04573)

Organisms have natural scales, such as lifetimes, body sizes, frequencies of movement to new locations, and residence times of material in digestive systems, and each scale has potential implications for acoustic effects. The effects of groups of organisms, like organisms themselves, aggregate in space and time. This review, including an assortment of unpublished information, examines examples of such aggregations, many of them documented acoustically. Light synchronizes many activities. Macroscopic animals forage primarily under cover of darkness. This phasing applies both to animals that extend appendages above the sediment–water interface and to animals that leave the seabed at night. Whereas their bottom-modifying activities are concentrated in nocturnal or crepuscular fashion, the bottom-modifying activities of the visual feeders follow a different phasing and often dominate the rate of change in acoustic backscatter from the interface. Light also acts through its effects on primary production, often concentrated in a very thin surficial layer atop the seabed. The supersaturation of oxygen does, and microbubble nucleation may, result. Where tidal velocities are large, light-set patterns are often tidally modulated. Activities of animals living below the seabed, however, remain a mystery, whose primary hope for solution is acoustic. [Work supported by ONR and DEPSCoR.]

9:05

4aUW4. The evolution of rippled seafloor topography with acoustic implications. Michael D. Richardson (Marine Geosciences Div., Naval Res. Lab., Stennis Space Center, MS 39529-5004) and Peter Traykovski (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Rippled seafloors are often responsible for anisotropic patterns of acoustic backscattering and allow penetration of high-frequency energy into the seafloor below the critical angle. Both natural and manipulative experiments conducted during the Sediment Acoustic Experiment (SAX99) demonstrate the importance of understanding the temporal evolution and characterizing the spatial statistics of naturally occurring ripple fields for prediction of sonar performance and the detection of buried targets. Current and wave-induced sand ripples evolve in a more or less predictable pattern. Numerous empirical and semi-empirical predictive models, based on well-established principles of sediment transport, allow prediction of ripple wavelength, height, and shape. Degradation of sand ripple fields, especially by biological processes such as feeding, burrowing, and emergence is less known and has not been modeled. The temporal evolution of rippled topography measured with sector scanning sonar in high-energy environments is presented. These high-fidelity and nearly continuous observations coupled with measurements of bottom currents and near-bottom wave-induced orbital motion provide improved insights and new models of the evolution of rippled seafloor topography. In low-energy environments (SAX99 and the proposed SAX04) the longer times between storms allow characterization of rates of biological processes which destroy ripple structure and create isotropic roughness. [Work supported by ONR.]

9:25

4aUW5. Time and space scales of bedform evolution in sandy nearshore and inner shelf environments. Alex E. Hay (Dept. of Oceanogr., Dalhousie Univ., Halifax, NS B3H 4J1, Canada)

Recent experiments in wave-forced nearshore and inner shelf environments have demonstrated through acoustic and optical imaging techniques a remarkable range of bedform patterns in mobile sandy sediments at sub-10 m horizontal scales. The observed bedstates differ both in the level of complexity of their spatial patterns, and in the time scales of their response to changes in hydrodynamic forcing conditions. Yet, the (re)occurrence of these different bed states at a given location can be quite repeatable and, once formed, the migration of the bedforms characterizing a particular state can in some instances be predicted reasonably well from the local hydrodynamics. These results provide a basis against which the present capability for predicting bedstate evolution in wave-forced mobile bed environments can be assessed. It is argued that, with the possible exception of a few simple cases, current models are able to predict neither the level of complexity in the observed characteristic patterns, nor the time evolution of the bed given the prior history of the forcing. Nevertheless, the repeatability of bedstate occurrence, and the reasonably good predictions of migration velocity and shear stress at the bed, together suggest that such a predictive capability may be achievable in the not-too-distant future.

10:00–10:15 Break

Contributed Papers

9:45

4aUW6. A comparison of models for the reflection loss of acoustic waves from a smooth water/sediment interface. Marcia J. Isakson and Andrew Worley (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

Reflection-loss measurements have been inverted using a composite poro-elastic model to produce realistic sediment properties. This type of inversion could provide a noninvasive method of characterizing sediments. However, it is not well understood how measurement variability affects the outcome of the inversion. When measurement variability is taken into account, other models may be able to predict reasonable sediment properties. In this study, several models will be considered for their ability to produce realistic sediment parameters and their fit to the data. Models considered will be the visco-elastic model, the Biot/Stoll model, the effective density fluid model, the Buckingham model, and the composite Biot squirt flow/shear model. Each model will be inverted using a simulated annealing algorithm with OASR as the forward model. Special emphasis will be placed on inverting within the statistical nature of the data. [Work supported by ONR, Ocean Acoustics.]

10:15

4aUW7. Buried target detection with a synthetic aperture sonar. John Piper (Coastal Systems Station, Code R21, Panama City, FL 32407-7001, piperje@ncsc.navy.mil)

Synthetic aperture sonar (SAS) systems can provide the high resolution and high signal-to-noise ratios that are important in detecting buried targets. Previous experiments with the Coastal Systems Station SAS have shown good results for above critical grazing angle targets and mixed results for below critical angle targets. Interest in characterizing the environmental conditions, which can allow detection at sub-critical angles, has prompted additional SAS testing. This paper describes a buried target experiment that included two 1.5-meter by 0.5-meter cylindrical targets,

one 35-cm silicon-fluid-filled sphere, and two 58-cm air-filled spheres. These targets were buried under sandy sediment in 17 meters of water approximately 1 mile offshore from Panama City Beach, Florida. Above critical angle and below critical angle runs were made during November 2001 and June 2002. Environmental conditions and the seafloor ripple structure were reported by divers and, during the 2002 experiment, measured by the Applied Physics Laboratory/University of Washington with their second-generation *in situ* measurement of porosity (IMP2) instrumentation.

10:30

4aUW8. Measurements of subcritical grazing angle detection of targets buried under a rippled sand interface. J. L. Lopes, C. L. Nesbitt, R. Lim (Coastal Systems Station, Code R21, 6703 W. Hwy. 98, Panama City, FL 32407-7001), D. Tang, K. L. Williams, and E. I. Thorsos (Univ. of Washington, Seattle, WA 98105-6698)

A series of controlled measurements were conducted to investigate shallow grazing angle acoustic detection of targets buried in sand having a rippled sediment–water interface. The measurements were performed in a 13.7-m deep, 110-m long, 80-m-wide test-pool with a 1.5-m layer of sand on the bottom. A silicone oil filled target sphere was buried under a rippled interface with contours formed by scraping the sand with a machined rake moved along a guide frame. Broadband, broad beam transducers were placed onto the shaft of a tilting motor. The transducers and tilting motor were attached to an elevated rail that enabled this assembly to be translated horizontally, permitting acquired data to be processed using synthetic aperture sonar techniques. Acoustic backscatter data were acquired at subcritical grazing angles in the frequency range of 10 to 50 kHz for various ripple wavelengths and heights. For each bottom configuration, the ripple profile over the buried target was measured using the In-situ Measurement of Porosity 2 (IMP2) system. Measurement results are presented that illustrate target detection via ripple scattering. The characteristics of the target return are found to depend sensitively on the ripple height and wavelength. [Work supported by ONR.]

10:45

4aUW9. Model/data comparisons of subcritical grazing angle detection of targets buried under a rippled sand interface. Kevin Williams, Eric Thorsos, Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, williams@apl.washington.edu), Raymond Lim, and Joe Lopes (Coastal Systems Station, Code R21, Panama City, FL 32407-7001)

Subcritical grazing angle target detection results for a sphere buried in Coastal System Station's Facility 383 pond are compared to predictions from two different models. Both models use perturbation theory to calculate the transmission across the rough interface due to ripple scattering. One is a simple sonar equation model that uses a first-order expression [Eq. (16), Jackson *et al.*, IEEE J. Ocean. Eng. **27**, 346–361 (2002)] to calculate the penetration, taking into account the measured ripple profile and the target strength estimate for the buried sphere. The other model uses the penetrating field to calculate target backscattering via a T-matrix formalism adapted to account for the specified sinusoidal roughness. These models have been combined with a reverberation model to predict detection SNR. The reverberation model uses a power law fit to the sand/water interface roughness spectrum determined from measured roughness profiles. [Work supported by ONR.]

11:00

4aUW10. Exploiting sediment acoustics properties for subcritical buried target detection using a synthetic aperture sonar aboard the AUV Reliant. Kerry W. Commander, Jose E. Fernandez, John E. Piper, and John S. Stroud (Coastal Systems Station, Code R21, Panama City, FL 32407-7001, commanderkw@ncsc.navy.mil)

The Coastal Systems Station High Frequency/ Low Frequency Synthetic Aperture Sonar (HF/LFSAS) was recently integrated into a Bluefin Robotics, Inc. Odyssey III Family autonomous underwater vehicle (AUV). This vehicle, designated Reliant, has been used in several at-sea experiments to determine the feasibility of using bottom roughness to enhance detection of buried targets at subcritical grazing angles. Previously, the HF/LFSAS was deployed on a towed 21-in. vehicle and made some remarkable buried target detections at subcritical angles. However, due to the complexities associated with using a towed platform, only a limited amount of data was obtained during these exercises. This sparse data set made it difficult to quantify the enhancing effects of bottom roughness on buried target detection. An extensive run matrix for buried target detection was completed this year using the Reliant AUV. The primary variables in the experiments were ripple orientation, standoff range, and grazing angle. Results from these sea tests are presented and compared to theoretical models.

11:15

4aUW11. Variation of sea-bed backscattering strength due to Bragg scattering. K. W. Commander, J. L. Lopes, and R. Lim (Coastal Systems Station, Code R21, 6703 W. Hwy. 98, Panama City, FL 32407-7001)

Bragg scattering from a rippled sea-bed may lead to reduced SNRs from proud targets, which adversely affects sonar performance. To quantify this effect, acoustic backscatter from a rippled bottom is investigated in the frequency range of 2–10 kHz by conducting a laboratory-type experiment to measure the reverberation levels from a bottom with a rippled interface. The experiment was performed in a 13.7-m deep, 110-m long, 80-m wide test pool that has 1.5 m of sand covering the bottom. In this experiment a calibrated parametric sonar was attached to a tower that was fitted with horizontal pan and vertical tilt motors. The rippled bottom was artificially formed with the aid of a sand scraper that consists of a frame and a rake that glides along the frame [Lopes *et al.*, ''Shallow Grazing Angle Sonar Detection of Targets Buried Under a Rippled Sand Interface,'' Oceans 2002 MTS/IEEE, pp. 461–467]. The bottom backscatter measurements are compared to predictions of a model in which the interface roughness is represented by a Gaussian spectrum centered on the imposed ripple frequency band.

11:30

4aUW12. Mechanical loading of a spherical hydrophone embedded in a sediment. Steven G. Kargl (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

The free-field calibration of the response of a hydrophone is typically performed in water and it is often assumed that the hydrophone is insensitive to loading when immersed in other materials. Hydrophones are routinely deployed in sandy sediments to measure the penetration of an acoustic field from a sonar. Models of the mechanical response of a spherical hydrophone will be discussed. Space is split into three concentric regions and an infinite external region. The innermost region is a vacuum, the inner spherical shell is an active element (e.g., a piezoelectric ceramic), and the outer shell is an impedance matching material. The active element is modeled as a homogenous elastic material such that its piezoelectric properties are neglected. The external medium can be either a fluid, elastic, or a poroelastic medium. The mechanical impedance shows that the loading causes a shift in the resonance frequency of the hydrophone as well as a change in the Q of the resonance. The sensitivity of the hydrophone will also be discussed. [Work supported by ONR.]

11:45

4aUW13. Measurements of bottom reflection loss in the East China Sea. Jee Woong Choi and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Measurements of bottom reflection loss were conducted in 105 m of water in the East China Sea (ECS) at location 29 39N, 126 49E as part of the Asian Seas International Acoustic Experiment (ASIAEX), in the spring of 2001. Sources in the frequency range 2–20 kHz were located at depths 25 m and 50 m, and receivers were deployed at a range of 500 m and at of depths 26 m and 52 m. Bottom reflection loss measurements are obtained as a function of frequency and, to a limited extent, grazing angle. Experimental values are compared with model calculations based on independently measured geoacoustic and bottom roughness parameters. The frequency dependence of sediment attenuation, influence of bottom roughness, and role of the critical angle will be discussed in the context of these measurements. [Work supported by ONR, ASIAEX program.]

Meeting of the Standards Committee Plenary Group

to be held jointly with the

ANSI-Accredited U.S. Technical Advisory Group (TAG) Meeting for: ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics

P. D. Schomer, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43 Acoustics
2117 Robert Drive, Champaign, Illinois 61821

H. E. von Gierke, Vice Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43
1325 Meadow Lane, Yellow Springs, Ohio 45387

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

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

S12	9:45 a.m. to 12:00 noon
S1	1:45 p.m. to 3:15 p.m.
S3	3:30 p.m. to 5:00 p.m.

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees. The ANSI-Accredited U.S. Technical Advisory Group (TAGs) for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, whose membership consists of members of S1 and S3, and other persons not necessarily members of these Committees, will meet during the Standards Plenary meeting. *The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting*. There will be a report on the interface of S1 and S3 activities with those of ISO/TC 43 and IEC/TC 29 including plans for future meetings of ISO/TC 43 and IEC/TC 29.

Members of S2 Mechanical Vibration and Shock (and U.S. TAG for ISO/TC 108 and five of its Subcommittees, SC1, SC2, SC3, SC5, and SC6) are also encouraged to attend the Standards Committee Plenary Group meeting even though the S2 meeting will take place one day earlier, on Wednesday, 30 April 2003, at 9:00 a.m.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3, and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. TAG
ISO		
P. D. Schomer, Chair H. E. von Gierke, Vice Chair	**ISO/TC 43** Acoustics	S1 and S3
P. D. Schomer, Chair H. E. von Gierke, Vice Chair	**ISO/TC 43/SC 1** Noise	S12
D. J. Evans, Chair	**ISO/TC 108** Mechanical Vibration and Shock	S2
R. Eshleman, Acting Chair	**ISO/TC 108/SC 1** Balancing, including Balancing Machines	S2
A. F. Kilcullen, Chair R. Eshleman, Vice Chair	**ISO/TC 108/SC2** Measurement and Evaluation of Mechanical Vibration and Shock as Applied to Machines, Vehicles and Structures	S2
D. J. Evans, Chair	**ISO/TC 108/SC3** Use and Calibration of Vibration and Shock Measuring Instruments	S2
D. D. Reynolds, Chair	**ISO/TC 108/SC4** Human Exposure to Mechanical Vibration and Shock	S3
R. L. Eshleman, Chair R. F. Taddeo, Vice Chair	**ISO/TC 108/SC5** Condition Monitoring and Diagnostics of Machines	S2
G. Booth, Chair	**ISO/TC 108/SC6** Vibration and Shock Generating Systems	S2
IEC		
V. Nedzelnitsky, U.S. TA	**IEC/TC 29** Electroacoustics	S1 and S3

Meeting of Accredited Standards Committee (ASC) S12 Noise

to be held jointly with the

ANSI-Accredited U.S. Technical Advisory Group (TAG) Meetings for: ISO/TC 43/SC 1, Noise

R. D. Hellweg, Chair S12
Compaq Computer Corporation, Acoustics Lab, Mechanical Engineering Group, MR01-3/03, 200 Forest Street, Marlborough, Massachusetts 01752

P. D. Schomer, Chair S12, and Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC 1, Noise
2117 Robert Drive, Champaign, Illinois 61821

H. E. von Gierke, Vice Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC 1, Noise
1325 Meadow Lane, Yellow Springs, Ohio 45387

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. There will be a report on the interface of S12 activities with those of ISO/TC 43/SC 1 Noise, including plans for future meetings of ISO/TC 43/SC 1. The Technical Advisory Group for ISO/TC 43/SC 1 consists of members of S12 and other persons not necessarily members of the Committee. Open discussion of committee reports is encouraged.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control, including biological safety, tolerance and comfort and physical acoustics as related to environmental and occupational noise.

THURSDAY AFTERNOON, 1 MAY 2003 ROOMS 103/104, 1:15 TO 5:00 P.M.

Session 4pAA

Architectural Acoustics, Noise and Education in Acoustics: Teaching Architectural Acoustics to Non-Acousticians

Rendell R. Torres, Chair
Architectural Acoustics Program, Rensselear Polytechnic Institute, School of Architecture, Greene Building, 110 8th Street, Troy, New York 12180-3590

Chair's Introduction—1:15

Invited Papers

1:20

4pAA1. Teaching room acoustics as a product sound quality issue. Mendel Kleiner and Daniel Vastfjall (Dept. of Appl. Acoust., Chalmers Univ. of Technol., S-41296 Gothenburg, Sweden)

The department of Applied Acoustics teaches engineering and architect students at Chalmers University of Technology. The teaching of room acoustics to architectural students has been under constant development under several years and is now based on the study of room acoustics as a product sound quality issue. Various listening sessions using binaural sound recording and reproduction is used to focus students' learning on simple, easy to remember concepts. Computer modeling using ray tracing software and auralization is also used extensively as a tool to demonstrate concepts in addition to other software for simple sound generation and manipulation. Sound in general is the focus of an interdisciplinary course for students from Chalmers as well as from a school of art, a school of design, and a school of music which offers particular challenges and which is almost all listening based.

1:35

4pAA2. Acoustics class at Berklee College of Music. Anthony K. Hoover (Cavanaugh Tocci Assoc. and Berklee College, 327F Boston Post Rd., Sudbury, MA 01776)

Berklee College of Music (in Boston) was developing its outstanding Music Technologies Division, and understood the need for a comprehensive class on acoustics. The result was a three-credit-hour class, offered twice per year, covering the fundamentals, architectural acoustics (outdoors, indoors, and transmission), vibration isolation, hearing and psychoacoustics, and more. One outgrowth was the Acoustical Society at Berklee, with presentations by local and visiting ASA members, yearly visits to an anechoic chamber, special studio sessions, tours, and joint meetings with professional societies. Over 2000 students have completed and performed well in the class. The author's favorite measure of success is the growing number of students who have chosen a career in acoustics. This paper will summarize and discuss this class.

1:50

4pAA3. Studio-based courses in architectural acoustics. Gary W. Siebein (School of Architecture, Univ. of Florida, P.O. Box 115702, Gainesville, FL 32611-5702, gsiebein@siebeinacoustic.com)

Examples of 4 nontraditional, elective courses offered to students in the Master of Architecture, Master of Science and Doctor of Philosophy degree programs in architecture are presented. Listening to Buildings involves experiential exercises in architectural acoustics for undergraduate students in a required large lecture course to form links between what people hear in a room and the architectural shapes and materials of the room that affects what is heard. The Applied Acoustics Design Lab takes students through each of the major components of an actual consulting project. Student work includes room acoustic design, sound isolation design, sound system design and HVAC noise control design for an actual building project. An Acoustical History of Theaters and Concert Halls teaches fundamentals of computer-based acoustic modeling and has students compare the physical design and aural simulations of a historic and a contemporary performance space. Graduate design studio in acoustics examines sound and silence as potential form-givers in architecture and as generative ideas for the basis of architectural interventions in complex sites and programs. Interior and exterior soundscapes and phenomenological philosophy form the background research for the studio.

2:05

4pAA4. Using acoustic measurement and analysis computer programs to teach architectural acoustics to architectural students. Robert C. Coffeen (School of Architecture and Urban Design, Marvin Hall, The Univ. of Kansas, Lawrence, KS 66045)

Explaining the concepts of sound amplitude and frequency to architectural students is difficult unless suitable visual and audible presentations are used. Computer programs designed for professional sound measurement and evaluation, and computer programs designed for acoustic analysis and auralization provide tools that can be employed to bring to life the basic sound concepts that must be understood by those students who will later be responsible for designing building spaces where hearing conditions are important.

2:20

4pAA5. Teaching noise control to architectural engineers. Ralph T. Muehleisen (Civil Environ. and Architectural Eng., Univ. of Colorado, 428 UCB, Boulder, CO 80309, ralph.muehleisen@colorado.edu)

Architectural engineers have to deal with acoustics in a variety of situations. HVAC engineers, structural engineers, safety engineers, lighting engineers, and construction engineers all need to be aware of noise, vibration, sound transmission, and general room acoustics. In this talk, the teaching of acoustics to architectural engineers is presented. In particular, the talk will focus on how to teach useful building noise control and architectural acoustics to architectural engineers with no previous background in acoustics and who may be less mathematically adept than other engineers or physicists.

2:35

4pAA6. A real challenge: Teaching acoustics to architecture students. Daniel R. Raichel (CUNY Graduate Ctr. and Douglas Eilar & Assoc., Fort Collins, CO 80526, raichel@juno.com)

The key to instilling the fundamentals of acoustics in architecture students is to arouse their interest. Because so many of the students are interested in music and high-fidelity equipment, it does not take much to ignite their interest in acoustics, particularly when they come to realize that perfectly good equipment can be undermined by poor room acoustics. Because they generally are not comfortable with mathematics, having had received perhaps no more than one or two semesters of introductory calculus; they need to be spoon-fed mathematics, even to the point of reviewing logarithmic manipulations, which are normally taught in secondary schools. The purpose of teaching acoustics to architects is not to make them acoustic experts, per se, but to make them appreciative of the effect of room acoustics and to understand that they must work hand-in-hand with acousticians when they design listening spaces that range in size from small classrooms to lecture halls to large concert halls. A regular acoustics text, such as that by Beranek, or Kinsler and Frey, or Raichel would be beyond the scope of an architectural course, but a text written especially for nonscience majors (such as that by Apfel) should and did serve admirably.

2:50

4pAA7. Teaching architectural acoustics to students of engineering versus students of architecture. Lily M. Wang (Architectural Eng. Prog., Univ. of Nebraska–Lincoln, 200B PKI, Omaha, NE 68182-0681, lwang4@unl.edu)

The author currently teaches architectural acoustics at the University of Nebraska to students in the College of Engineering and to students in the College of Architecture. These two groups have different backgrounds and focuses: for example, engineers are more mathematically prepared, while architects are more visually oriented. However, the instructor's goals for the members of both groups are similar, namely that they (1) master the fundamentals of good architectural acoustic design and (2) learn to respect and communicate well with each other during the design process, leading to more successful projects. This presentation will demonstrate the similarities and differences in the methods that are used by the author to address these two groups.

3:05–3:15 Break

3:15

4pAA8. One approach to architectural acoustics in education. J. Christopher Jaffe (Jaffe Holden Acoust., Inc., 114A Washington St., Norwalk, CT 06854)

In the fall of 1997, Dean Alan Balfour of the School of Architecture at the Rennselaer Polytechnic Institute asked me to introduce an undergraduate 14 credit certificate course entitled "Sonics in Architecture." Subsequently, the program was expanded to include a Master's Degree in Building Science. This paper discusses the trials and tribulations of building a scientific program in a liberal arts school. In addition, the problem of acquiring the research funds needed to provide tuition assistance for graduate students in Architectural Acoustics is reviewed. Information on the curriculum developed for both the lecture and laboratory courses is provided. I will also share my concerns regarding the teaching methods currently prevalent in many schools of architecture today, and how building science professionals might assist in addressing these issues.

3:30

4pAA9. The acoustics teaching drawdown at architectural schools in America: How to prepare learning materials for the new curricula. M. David Egan (Clemson Univ. and Egan Acoust., Box 365, Anderson, SC 29622)

In the early 1990s, the Association of Collegiate Schools of Architecture (ACSA) directory listed more than forty faculty members as teaching a course on acoustics. Today at the more than 120 ACSA full-member schools, there are only eight faculty members listed as teaching acoustics. Clearly the focus at most architecture schools in North America does not include the thorough study of science and technology. Courses in acoustics, illumination, HVAC systems, and fire protection are no longer included in curricula of most schools. Unlike studies in law and medicine, each school controls its own curriculum and what it calls its degree [A. O. Dean, Arch. Record (August 2002), pp. 84–92]. Learning materials have been developed to aid architecture professors who now must cover acoustics in the design studio. These materials, based on more than three decades of experience teaching and writing seven books on building technologies, also are intended to support self-study by students and design professionals. Examples of self-study exercises and hands-on exercises will be described [M. D. Egan, *Arch. Acoustics Workbook* (2000)]. A checklist of "writing principles" for the preparation of educational materials in architectural acoustics also will be presented.

Contributed Papers

3:45

4pAA10. Architectural acoustics to non acousticians: Prediction auralization in design and consulting. Quinsan Ciao (Dept. of Architecture, Ball State Univ., Muncie, IN 47306)

Acoustic theories and principals are not necessarily familiar to most of the parties closely related to architectural acoustic issues. For example, architects involved in building designs, occupant communicates involved in space improvements are often the clients of architectural acoustic consultants. On one hand they are responsible to make knowledgeable decisions, on the other hand they do not have time or interest to be systematically trained in acoustics principals and theories. Properly educating such clients is one of the challenges constantly faced by architectural acoustic consultants. Traditionally, consultants are tormented between two different approaches. One is to deeply educate clients in architectural acoustics. The other is to convince the clients to trust and leave the matter to experts. Neither has been satisfactory. The former is insufficient and inefficient to establish appreciation of practical results, while the lattter is resisted for lack of transparency. Thanks to the latest development of computing technologies, a new approach is available, which is to demonstrate the acoustic effects in audible media through auralization and prediction in conjunction with explanation of simple principles. This paper discusses this approach of client education in practices with case study examples.

4:00

4pAA11. The use of lattice gas method in explaining architectural acoustics principles. Sung Y. Yoon and Dana M. Smith (Dept. of Architecture, Architectural Acoust., Greene, RPI, Troy, NY 12180)

Many concepts in architectural acoustics are based on physical phenomena, such as sound propagation, reflection, etc. In explaining those phenomena, it is necessary to employ the principles of physical acoustics. Unfortunately, the methods of physical acoustics are not easily approachable unless one is familiar with mathematical physics techniques, partial differential equations, for example. The lattice gas method, based on the cellular automata concept, can be an alternative to avoid the use of complicated mathematical equations. A relatively simple code can be used to generate computer simulations. The existing models for sound propagation by Sudo and Sparrow, are examined. Models for other phenomena involving architectural acoustics are suggested.

4:15

4pAA12. Teaching acoustics courses online. Courtney B. Burroughs (Appl. Res. Lab., Penn State Univ., State College, PA 16804)

Three graduate-level courses in noise control engineering have been taught by the Graduate Program in Acoustics through the Penn State World Campus for more than four years. In each course, students receive

(1) a study guide which gives an overview of the course, the software, and assignments, and (2) a CD which contains the content of the course with animations embedded in the text. Because all communications are online, students have been able to take these courses asynchronously on their own schedule with minimum disruptions in their home and work schedules. Students work on group projects, known as collaborative learning activities, and conduct simulated measurements using virtual instruments. Individual learning activities include study questions, written problems, computer coding, and interactive demonstrations. The first course has been offered four times and several students have completed all three courses. Overviews of the elements of these online courses that have been successful and the elements that have presented problems, along with a discussion of adjustments that we have been able to make and those that we would like to make to improve the courses, will be presented.

4:30–5:00

Panel Discussion

THURSDAY AFTERNOON, 1 MAY 2003 ROOM 201, 1:10 TO 2:45 P.M.

Session 4pABa

Animal Bioacoustics: Nature's Orchestra—Acoustics of Singing and Calling Animals II

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

Chair's Introduction—1:10

Contributed Papers

1:15

4pABa1. Nonlinear phenomena in northern mockingbird (*Mimus polyglottos*) vocalizations: Acoustics and physiology. Sue Anne Zollinger, Tobias Riede, and Roderick A. Suthers (Indiana Univ., 1001 E. Third St., Bloomington, IN 47405)

The bird sound source (syrinx), like the mammalian larynx, has been discovered to behave as a nonlinear system of coupled oscillators. Modeling approaches as well as excised syrinx experiments show that the syrinx can be considered a nonlinear sound source. Until now, only few acoustic data were available showing to what extent a birdsong species' vocal repertoire actually contains nonlinear phenomena, like subharmonics, frequency jumps, biphonation and deterministic chaos. The passerine syrinx, in contrast to the mammalian larynx, consists of two sound sources, one situated in each of the bronchi that are independently controlled. Bronchial airflow measurements showed that individual mockingbirds use each side of the syrinx independently. However, there are interactions of both sound sources, which have not yet been investigated acoustically/spectrographically. The present study shows how many of the spectrographically observed nonlinear phenomena are attributed to the interaction of the two sound sources, i.e., nonlinear phenomena, while both sound sources show positive airflow values, compared to single sound source utterances, i.e., positive flow through one sound source only. Nonlinear phenomena could be attributed to the interaction of the two sound sources (earlier referred to as ''two-voice-phenomenon'') but were also found to be produced by a single sound source.

1:30

4pABa2. Nonlinear acoustics in the pant-hoot vocalization of common chimpanzees (*Pan troglodytes*). Tobias Riede (Indiana Univ., 315 Jordan Hall, Bloomington, IN 47405), Adam Clark Arcadi, and Michael J. Owren (Cornell Univ., Ithaca, NY)

Pant-hoots produced by chimpanzees are multi-call vocalizations. While predominantly harmonically structured, pant-hoots can exhibit acoustic complexity that has recently been found to result from inherent nonlinearity in the vocal-fold dynamics. This complexity reflects abrupt shifts between qualitatively distinct vibration patterns (known as modes), which include but are not limited to simple, synchronous movements by the two vocal folds. Studies with humans in particular have shown that as the amplitude and vibration rate increase, vocal-fold action becomes increasingly susceptible to higher-order synchronizations, desynchronized movements, and irregular behavior. We examined the occurrence of these sorts of nonlinear phenomena in pant-hoots, contrasting quieter and lower-pitched introduction components with loud and high-pitched climax calls in the same sounds. Spectrographic evidence revealed four classic kinds of nonlinear phenomena, including discrete frequency jumps, subharmonics, biphonation, and deterministic chaos. While these events were virtually never found in the introduction, they occurred in more than half of the climax calls. Biphonation was by far the most common. Individual callers varied in the degree to which their climax calls exhibited nonlinear phenomena, but we are consistent in showing more biphonation than any of the other forms. These outcomes demonstrate that understanding these calls requisitely requires an understanding of such events.

1:45

4pABa3. Intensity discrimination as a function of level and frequency in three species of birds. Amanda M. Lauer, Kirsten Poling, and Robert J. Dooling (Dept. of Psych., Univ. of Maryland, College Park, MD 20742, alauer@psyc.umd.edu)

Many studies have examined frequency discrimination in birds, but there has not been as complete a description of avian intensity discrimination abilities. Birds appear to be slightly less sensitive to changes in intensity than humans and other mammals; however, few studies have systematically looked at the effects of both frequency and presentation level on intensity discrimination in birds. Here we describe intensity discrimination as a function of frequency and sensation level in two small songbird species, the canary (*Serinus canarius*), the zebra finch (*Taeniopygia guttata*), and a nonsongbird species, the budgerigar (*Melopsittacus undulatus*). Intensity difference limens (DLIs) for pure tones were obtained from birds using standard operant conditioning procedures and the Method of Constant Stimuli. DLIs ranged from approximately 2–6 dB, which are slightly larger than the DLIs reported in mammals. For all three species, DLIs become smaller with increasing presentation level, but show little effect across frequency for a given level. These results are consistent with previous reports in other species. [Work supported by NIH DC01372 to RJD and DC05450 to AML.]

2:00

4pABa4. Automatic type classification and speaker identification of african elephant (*Loxodonta africana*) vocalizations. Patrick J. Clemins and Michael T. Johnson (Elec. and Computer Eng. Dept., Marquette Univ., P.O. Box 1881, Milwaukee, WI 53201-1881, Patrick.Clemins@marquette.edu)

This paper presents a system for automatically classifying African elephant vocalizations based on systems used for human speech recognition and speaker identification. The experiments are performed on vocalizations collected from captive elephants in a naturalistic environment. Features used for classification include Mel-Frequency Cepstral Coefficients (MFCCs) and log energy which are the most common features used in human speech processing. Since African elephants use lower frequencies than humans in their vocalizations, the MFCCs are computed using a shifted Mel-Frequency filter bank to emphasize the infrasound range of the frequency spectrum. In addition to these features, the use of less traditional features such as those based on fundamental frequency and the phase of the frequency spectrum is also considered. A Hidden Markov Model with Gaussian mixture state probabilities is used to model each type of vocalization. Vocalizations are classified based on type, speaker and estrous cycle. Experiments on continuous call type recognition, which can classify multiple vocalizations in the same utterance, are also performed. The long-term goal of this research is to develop a universal analysis framework and robust feature set for animal vocalizations that can be applied to many species.

2:15

4pABa5. Sound localization of aerial broadband noise in pinnipeds. Marla M. Holt, Ronald J. Schusterman, David Kastak, and Brandon L. Southall (Long Marine Lab., Univ. of California, Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060)

Pinnipeds (seals, sea lions, and walruses) emit broadband calls on land as part of their communication system in order to coordinate their reproductive activities. How well do they localize these types of signals? In this study, the aerial sound localization acuities of a harbor seal (*Phoca vitulina*), a California sea lion (*Zalophus californianus*), and a northern elephant seal (*Mirounga angustirostris*) were measured in the horizontal plane with a broadband white noise stimulus. Testing was conducted in a hemi-anechoic chamber using a left/right forced choice procedure to measure the minimum audible angle (MAA) for each subject. MAAs were defined as half the angular separation of two sound sources relative to a subject's midline that corresponded to 75% correct discrimination. MAAs were 3.6, 4.2, and 4.7 deg for the harbor seal, California sea lion, and northern elephant seal, respectively. These results demonstrate that these pinniped species had sound localization abilities comparable to the domestic cat and rhesus macaques. The acuity differences between our subjects were small, were not predicted by head size, and therefore likely reflect the relatively acute abilities of other pinniped species to localize aerial broadband signals.

2:30

4pABa6. Evoked potential measurement of the masked hearing threshold of a Pacific white-sided dolphin (*Lagenorhynchus obliquidens*). Whitlow W. L. Au (Marine Mammal Res. Prog., Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734), Thomas Jeanette, A. Western (Western Illinois Univ./RIRUC, Moline, IL 61265), and Kenneth M. Rameriz (John G. Shedd Aquarium, Chicago, IL 60605)

The masked hearing threshold of a Pacific white-sided dolphin (*Lagenorhynchus obliquidens*) was determined by measuring the animal's auditory brainstem response (ABR). The dolphin was trained to wear surface-contact electrodes embedded in suction cups and to swim into a hoop centered at 1 m below the water surface facing a sound projector 5 m away. Broadband transient signals with center frequencies of 8, 16, 32, 64, 80, and 100 kHz were used as the stimuli. ABR signals were measured by digitizing the electrode signals in 32 point blocks at a sampling rate of 20 kHz. Five hundred blocks were averaged in order to obtain an ABR. The response latency for suprathreshold threshold signals was approximately 1.9 ms with the highest peak-to-peak ABR amplitude of approximately 2.8 uV occurring for a signal frequency of 64 kHz. The spectrum of the ABR signal was similar to that of Tursiops truncatus, with a major peak at 1120 Hz and a secondary peak at 664 Hz. Threshold was determined by progressively reducing the amplitude of the stimulus until an evoked potential could not be detected. The energy signal-to-noise ratio within an integration window at threshold varied between 1 and 8 dB.

THURSDAY AFTERNOON, 1 MAY 2003 ROOM 201, 3:00 TO 5:05 P.M.

Session 4pABb

Animal Bioacoustics: General Topics in Animal Bioacoustics

Elizabeth Brittan-Powell, Chair

Psychology Department, University of Maryland, Biology-Psychology Building, College Park, Maryland 20742

Chair's Introduction—3:00

Contributed Papers

3:05

4pABb1. Toward standardization of noise exposure in animal bioacoustics. Robert Burkard (Ctr. for Hearing & Deafness, Univ. at Buffalo, 215 Parker Hall, Buffalo, NY 14214)

In the United States, the American National Standards Institute (ANSI) Accredited Standards Committee on Bioacoustics S3 develops voluntary standards related to psychological and physiological acoustics, including areas such as general acoustics, vibration, and shock. Several years ago, an S3 working group (WG) was formed, WG90 Animal Bioacoustics, with the goal of developing guidelines and standards in the area of animal bioacoustics, including relevant terminology, procedures to quantify noise exposure, and the effects of noise exposure on animals. The scope of this working group involves both terrestrial and marine animals, and their environments. Although ANSI Standards are voluntary, the development of accepted sound measurement procedures is needed before any legislation can be implemented that is intended to protect these animals from sound exposures detrimental to their long-term survival. To date, no standards have emerged from the Animal Bioacoustics working group. The purpose of this presentation is to provide an overview of the ANSI standards development process, briefly review some of standards that have been developed that are intended to quantify human exposure to sound, as well as describe the standards that need to be developed in order to protect both terrestrial and marine animals from excessive sound exposure.

3:20

4pABb2. Ocean noise and marine mammals: A summary report of the U.S. National Research Council Committee on Potential Impacts of Ambient Noise in the Ocean on Marine Mammals. George V. Frisk (Appl. Oceanogr. Phys. and Eng. Dept., M.S. #11, Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, gfrisk@whoi.edu)

In 2000, the U.S. National Ocean Partnership Program, with leadership by the Office of Naval Research, requested that the U.S. National Academies examine the potential effects of noise in the sea on marine mammals. A committee of 11 international experts was convened to study this controversial issue. The committee included marine mammals specialists, acousticians, bioacousticians, a geophysical exploration expert, and an expert in vessel engineering. Specifically, the committee was asked to evaluate the human and natural contributions to ocean noise and describe the long-term trends in noise levels, especially from human activities. In the report, the committee outlines the research needed to evaluate the impacts of marine noise from various sources (natural, commercial, naval, and acoustic-based ocean research) on marine mammal species, especially in biologically sensitive areas. The study reviews and identifies gaps in existing marine noise databases and recommends research needed to develop a model of ocean noise that incorporates temporal, spatial, and frequency-dependent variables. The findings and research recommendations of the committee will be presented.

3:35

4pABb3. Acoustic signal detection of manatee calls. Christopher Niezrecki, Richard Phillips, Michael Meyer (Dept. of Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL 32611-6250), and Diedrich O. Beusse (College of Veterinary Medicine, Univ. of Florida, Gainesville, FL 32610-0126)

The West Indian manatee (trichechus manatus latirostris) has become endangered partly because of a growing number of collisions with boats. A system to warn boaters of the presence of manatees, that can signal to boaters that manatees are present in the immediate vicinity, could potentially reduce these boat collisions. In order to identify the presence of manatees, acoustic methods are employed. Within this paper, three different detection algorithms are used to detect the calls of the West Indian manatee. The detection systems are tested in the laboratory using simulated manatee vocalizations from an audio compact disc. The detection method that provides the best overall performance is able to correctly identify $\approx$96% of the manatee vocalizations. However the system also results in a false positive rate of $\approx$16%. The results of this work may ultimately lead to the development of a manatee warning system that can warn boaters of the presence of manatees.

3:50

4pABb4. Localization of dolphin whistles through frequency domain beamforming using a narrow aperture audio/video array. Keenan R. Ball (Woods Hole Oceanogr. Inst., Dept. 4., M.S. 18, Woods Hole, MA 02543) and John R. Buck (UMass Dartmouth, North Dartmouth, MA 02747)

Correlating the acoustic and physical behavior of marine mammals is an ongoing challenge for scientists studying the links between acoustic communication and social behavior of these animals. This talk describes a system to record and correlate the physical and acoustical behavior of dolphins. A sparse, short baseline audio/video array consisting of 16 hydrophones and an underwater camera was constructed in a cross configuration to measure the acoustic signals of vocalizing dolphins. The bearings of vocalizing dolphins were estimated using the broadband frequency domain beamforming algorithm for sparse arrays to suppress grating lobes of Thode *et al.* [J. Acoust. Soc. Am. **107** (2000)]. The estimated bearings from the acoustic signals were then converted to video image coordinates and a marker was placed on the video image. The system was calibrated both at an indoor tank and from an outdoor dock at UMass Dartmouth prior to field tests in a natural lagoon at the Dolphin Connection on Duck Key, FL. These tests confirmed that the system worked well within the limits of underwater visibility by consistently placing the marker on or near the whistling or echolocating dolphin. [Work supported by NSF Ocean Sciences.]

4:05

4pABb5. Automated detection of sperm whale sounds as a function of abrupt changes in sound intensity. Christopher D. Walker (Dept. of Computer Sci., Univ. of Southern Mississippi, Hattiesburg, MS 39406, cdwalker@ocean.otr.usm.edu), Grayson H. Rayborn, Benjamin A. Brack, Stan A. Kuczaj, and Robin L. Paulos (Univ. of Southern Mississippi, Hattiesburg, MS 39406)

An algorithm designed to detect abrupt changes in sound intensity was developed and used to identify and count sperm whale vocalizations and to measure boat noise. The algorithm is a MATLAB routine that counts the number of occurrences for which the change in intensity level exceeds a threshold. The algorithm also permits the setting of a ''dead time'' interval to prevent the counting of multiple pulses within a single sperm whale click. This algorithm was used to analyze digitally sampled recordings of ambient noise obtained from the Gulf of Mexico using near bottom mounted EARS buoys deployed as part of the Littoral Acoustic Demonstration Center experiment. Because the background in these data varied slowly, the result of the application of the algorithm was automated detection of sperm whale clicks and creaks with results that agreed well with those obtained by trained human listeners. [Research supported by ONR.]

4:20

4pABb6. Behavior of dusky dolphins foraging on the deep-scattering layer in Kaikoura Canyon, New Zealand. Kelly Benoit-Bird (Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734), Bernd Wursig, and Cynthia McFadden (Texas A&M Univ., College Station, TX 77843-2258)

Little is known of foraging habits of sound-scattering layer consumers. A 200-kHz echosounder was used to survey dusky dolphins and the sound-scattering layer in winter 2002, in Kaikoura Canyon, New Zealand. Visual observations of dolphin surfacings occurred 84% of the time that dolphins were acoustically detected, confirming identifications from the acoustic data. Dusky dolphins were within the layer at 2000 h (about 1.5 h after dusk), within 125 m of the surface. As the layer rose to within 30 m of the surface at 0100 h, the observed depth of dolphins decreased presumably as the dolphins followed the vertical migration of their prey. The mean depth of dolphins was within the scattering layer except when the top of the layer was deeper than 125 m. Dusky dolphins often forage within large groups. Acoustically identified subgroups of coordinated animals ranged from 1 to 5 dolphins. Subgroup size varied with time of night,

minimum depth of the scattering layer, and the variance of the food resource. The largest subgroups occurred when the scattering layer was closest to the surface, and when the layer was most heterogeneous. Time, depth of layer, and layer variance contributed significantly to predicting foraging dusky dolphin subgroup size.

4:35

4pABb7. Head sinuses, melon, and jaws of bottlenose dolphins, *Tursiops truncatus*, observed with computed tomography structural and single photon emission computed tomography functional imaging. Sam Ridgway, Dorian Houser, James J. Finneran, Don Carder, William Van Bonn, Cynthia Smith (Navy Marine Mammal Prog., 235 PLBS, SSC SD, 53560 Hull, San Diego, CA 92152), Carl Hoh, Jacqueline Corbeil, and Robert Mattrey (School of Medicine, Univ. of California, San Diego, CA 92093)

The head sinuses, melon, and lower jaws of dolphins have been studied extensively with various methods including radiography, chemical analysis, and imaging of dead specimens. Here we report the first structural and functional imaging of live dolphins. Two animals were imaged, one male and one female. Computed tomography (CT) revealed extensive air cavities posterior and medial to the ear as well as between the ear and sound-producing nasal structures. Single photon emission computed tomography (SPECT) employing 50 mCi of the intravenously injected ligand technetium [Tc-99m] biscisate (Neurolite) revealed extensive and uptake in the core of the melon as well as near the pan bone area of the lower jaw. Count density on SPECT images was four times greater in melon as in the surrounding tissue and blubber layer suggesting that the melon is an active rather than a passive tissue. Since the dolphin temporal bone is not attached to the skull except by fibrous suspensions, the air cavities medial and posterior to the ear as well as the abutment of the temporal bone, to the acoustic fat bodies of each lower jaw, should be considered in modeling the mechanism of sound transmission from the environment to the dolphin ear.

4:50

4pABb8. Concentric scheme of monkey auditory cortex. Hiroko Kosaki, Richard C. Saunders, and Mortimer Mishkin (Neuropsychology Lab., NIMH, 49/1B80, 49 Convent Dr., Bethesda, MD 20892)

The cytoarchitecture of the rhesus monkey's auditory cortex was examined using immunocytochemical staining with parvalbumin, calbindin-D28K, and SMI32, as well as staining for cytochrome oxidase (CO). The results suggest that Kaas and Hackett's scheme of the auditory cortices can be extended to include five concentric rings surrounding an inner core. The inner core, containing areas *A*1 and *R*, is the most densely stained with parvalbumin and CO and can be separated on the basis of laminar patterns of SMI32 staining into lateral and medial subdivisions. From the inner core to the fifth (outermost) ring, parvalbumin staining gradually decreases and calbindin staining gradually increases. The first ring corresponds to Kaas and Hackett's auditory belt, and the second, to their parabelt. SMI32 staining revealed a clear border between these two. Rings 2 through 5 extend laterally into the dorsal bank of the superior temporal sulcus. The results also suggest that the rostral tip of the outermost ring adjoins the rostroventral part of the insula (area Pro) and the temporal pole, while the caudal tip adjoins the ventral part of area 7*a*.

THURSDAY AFTERNOON, 1 MAY 2003 ROOM 208, 1:00 TO 4:00 P.M.

Session 4pBB

Biomedical Ultrasound/Bioresponse to Vibration, Signal Processing in Acoustics and Physical Acoustics: High-Intensity Focused Ultrasound (HIFU) and Imaged-Guided Therapy II

J. Brian Fowlkes, Chair

Department of Radiology, University of Michigan Medical Center, Kresge III R3315, 200 Zina Pitcher Place, Ann Arbor, Michigan 48109-0553

Invited Papers

1:00

4pBB1. Detection theory applied to high intensity focused ultrasound (HIFU) treatment evaluation. Narendra Sanghvi, Adam Wunderlich, Ralf Seip, Jahangir Tavakkoli (Focus Surgery, Inc., Indianapolis, IN 46226), Kris Dines (XDATA Corp., Indianapolis, IN 46226), Michael Baily, and Lawrence Crum (Univ. of Washington, Seatle, WA)

The aim of this work is to develop a HIFU treatment evaluation algorithm based on 1-D pulse/echo (P/E) ultrasound data taken during HIFU exposures. The algorithm is applicable to large treatment volumes resulting from several overlapping elementary exposures. Treatments consisted of multiple HIFU exposures with an on-time of 3 seconds each, spaced 3 mm apart, and an off-time of 6 seconds in between HIFU exposures. The HIFU was paused for approximately 70 milliseconds every 0.5 seconds, while P/E data was acquired along the beam axis, using a confocal imaging transducer. Data was collected from multiple *in vitro* and *in vivo* tissue treatments, including shams. The cumulative energy change in the P/E data was found for every HIFU exposure, as a function of depth. Subsequently, a likelihood ratio test with a fixed false alarm rate was used to derive a positive or negative lesion creation decision for that position. For false alarm rates less than 5%, positive treatment outcomes were consistently detected for better than 90% of the HIFU exposures. In addition, the algorithm outcome correlated to the applied HIFU intensity level. Lesion formation was therefore successfully detected as a function of dosage. [Work supported by NIH SBIR Grant 2 R 44 CA 83244-02.]

1:25

4pBB2. Nonlinear pulse-echo imaging methods for HIFU-induced lesion visualization. Emad Ebbini (ECE Dept., Univ. of Minnesota, Minneapolis, MN 55455)

We have recently investigated the use of two different nonlinear imaging methods for visualization of HIFU-induced lesions. The hypothesis is that bubble activity at the location of the HIFU beam can be observed using diagnostic ultrasound with proper post-beamforming signal processing of the rf data. In particular, pulse inversion (PI) and quadratic imaging techniques have been investigated for this purpose. Results from over 100 *ex vivo* tissue experiments clearly demonstrate the superiority of the nonlinear imaging techiques over conventional B-scan imaging in terms of accurate mapping of lesion boundaries. We are currently examining imaging data to determine the use of nonlinear imaging methods for quantitative assessment of tissue damage. In this paper, we will describe a new experimental procedure for determining the existence of bubble activity associated with different levels of exposure to HIFU, from underexposure to overexposure conditions. Images from conventional B-scan will be compared and contrasted with quadratic and pulse inversion images for various levels of exposure. These results continue to support the hypothesis that bubble activity is extremely important in both lesion formation and lesion visualization using ultrasonic techniques.

1:50

4pBB3. Factors affecting radiation force method for monitoring therapeutic ultrasonic lesions. Frederic L. Lizzi, Robert Muratore, Samuel Mikaelian, Paul Lee, and S. Kaisar Alam (Riverside Res. Inst., 158 William St., New York, NY 10038)

Investigations are being conducted to develop a method using motion induced by radiation force to monitor HIFU lesions, by virtue of their increased stiffness. A therapeutic transducer, periodically excited at subthreshold levels, generates the radiation force: a collinear diagnostic transducer monitors the degree and time-course of induced motion. Lesions are detected by changes in pre- and post-treatment motion patterns. *In vitro* experiments and computer simulations have been performed to clarify the roles of several phenomena so that optimized systems can be designed for practical on-line applications. Lesion attenuation has been found to be a key factor since it directly affects radiation force. Lesions with near-normal attenuation exhibit less motion than normal, because of increased stiffness. Lesions with significantly higher attenuation can exhibit increased motion, due to increased force, but they demonstrate spatial patterns differing from normal. The profile of the push beam was found to affect the magnitude, spatial pattern, and time course of induced motion. The diagnostic transducer's bandwidth and beam profile were shown to affect the precision and sensitivity of motion characterization. Quantitative findings from these studies are being combined with thermal studies to design monitoring systems for specific applications.

2:15–2:30 Break

Contributed Papers

2:30

4pBB4. Study of a scanning HIFU therapy protocol, Part I: Theory and simulations. Steven G. Kargl and Marilee A. Andrew (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Over the last several years, many researchers have compared model predictions of isolated thermal lesions, caused by acoustic fields generated from (spherically) focused transducers, to thermal lesions created in tissue phantoms, *in vitro* soft tissue samples, and *in vivo* soft tissue samples. The models typically couple a nonlinear acoustic field from a focused transducer to the bio-heat transfer equation (BHTE). Recent experiments in polyacrylamide gel phantoms and excised bovine liver samples have demonstrated the possible deposition of thermal lesions in a scanning mode. An initial modeling effort to predict scanned thermal lesions will be discussed. The nonlinear acoustic field from a spherically focused transducer is predicted by a time–domain solution of the Khokhlov–Zabolotskaya–Kuznetsov equation. This field is then used as a heat source in the BHTE. The importance of scan rate, acoustic frequency, and time-averaged intensity will be investigated. [Work supported by USAMRMC.]

2:45

4pBB5. Fast calculations of the exact nearfield produced by a circular piston. Robert J. McGough (Dept. of Elec. and Computer Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48824-1226) and Thaddeus V. Samulski (Duke Univ. Medical Ctr.)

The computation time and the numerical error are both reduced substantially with an improved technique that calculates the nearfield pressure produced by a circular piston. A comparison of simulation results shows that, once the errors for all results are normalized, this new technique is at least 3 to 4 times faster than the impulse response approach of Oberhettinger and Stepanishen. Further improvements are obtained when the computational grid is divided into sectors defined by the angle between the coordinates of each field point and the normal evaluated at the center of the circular piston. The sector number is then converted through a linear function to a value that specifies the sampling of the integrand. When continuous wave excitations are applied to a simulated piston with a radius of five wavelengths, the resulting computations verify that this sectoring scheme reduces the computation time by an additional factor of 2 without increasing the maximum error value. These fast calculations are especially useful in numerical simulations for thermal therapy, where pressure fields are computed in large volumes for arrays with multiple sources.

3:00

4pBB6. Study of a scanning HIFU therapy protocol, Part II: Experiment and results. Marilee A. Andrew, Peter Kaczkowski, Bryan W. Cunitz, Andrew A. Brayman, and Steven G. Kargl (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Instrumentation and protocols for creating scanned HIFU lesions in freshly excised bovine liver were developed in order to study the *in vitro* HIFU dose response and validate models. Computer-control of the HIFU transducer and 3-axis positioning system provided precise spatial placement of the thermal lesions. Scan speeds were selected in the range of 1 to 8 mm/s, and the applied electrical power was varied from 20 to 60 W. These parameters were chosen to hold the thermal dose constant. A total of six valid scans of 15 mm length were created in each sample; a 3.5 MHz single-element, spherically focused transducer was used. Treated samples were frozen, then sliced in 1.27 mm increments. Digital photographs of slices were downloaded to computer for image processing and analysis. Lesion characteristics, including the depth within the tissue, axial length, and radial width, were computed. Results were compared with those generated from modified KZK and BHTE models, and include a comparison of the statistical variation in the across-scan lesion radial width. [Work supported by USAMRMC.]

3:15

4pBB7. Estimating "true" HIFU-induced temperature changes using thermocouples. Xinmai Yang, Ronald A. Roy, and R. Glynn Holt (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, xmyang@bu.edu)

An easy and economical way to measure temperature rises in tissue and tissue-mimicking phantoms is with imbedded thermocouples. However, a thermocouple undergoing HIFU exposure is subject to the formation of a viscous boundary layer, leading to an additional local temperature rise not experienced by the rest of the focal region. The resulting temperature measurement is then not indicative of the average undisturbed temperature in the region. This is the so-called thermocouple artifact (we will argue against the use of the word artifact). Motivated by our bubble-enhanced heating experiments, we present a simple method for estimating HIFU-induced temperature changes. The influence of the thermocouple is modeled by an effective increase in the local sound absorption coefficient. The effective absorption coefficient is estimated by measuring the increased rate of heating brought on by the thermocouple. The temperature rise in the medium adjacent to the thermocouple is then predicted by incorporating the effective local absorption into the acoustic source term in the 3D heat conduction equation. Comparisons between simulation and experiment and between the reported method and other methods for thermocouple-artifact correction are presented. [Work supported by the U.S. Army.]

3:30

4pBB8. Monitoring evolution of HIFU-induced lesions with backscattered ultrasound. Ajay Anand and Peter J. Kaczkowski (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 40th St. NE, Seattle, WA 98105, ajaya@apl.washington.edu)

Backscattered radio frequency (rf) data from a modified commercial ultrasound scanner were collected in a series of *in vitro* experiments in which high intensity focused ultrasound (HIFU) was used to create lesions in freshly excised bovine liver tissue. Two signal processing approaches were used to visualize the temporal evolution of lesion formation. First, apparent tissue motion due to temperature rise was detected using cross-correlation techniques. Results indicate that differential processing of travel time can provide temperature change information throughout the therapy delivery phase and after HIFU has been turned off, over a relatively large spatial region. Second, changes in the frequency spectrum of rf echoes due to changes in the scattering properties of the heated region were observed well before the appearance of hyper-echogenic spots in the focal zone. Furthermore, the increase in attenuation in the lesion zone changes the measured backscatter spectrum from regions distal to it along the imaging beam. Both effects were visualized using spectral processing and display techniques that provide a color spatial map of these features for the clinician. Our results demonstrate potential for these ultrasound-based techniques in targeting and monitoring of HIFU therapy, and perhaps post-treatment visualization of HIFU-induced lesions.

3:45

4pBB9. Acoustic field of a spherically curved wedge transducer. Jeffrey A. Ketterling (Riverside Res. Inst., 156 William St., New York, NY 10038)

High-intensity focused ultrasound (HIFU) applications typically utilize spherically curved transducers, often with modifications to the surface electrode. These modifications, such as rectangular cut-outs, can normally be modeled as a collection of annular rings and wedges. The acoustic field of an annular ring has been well characterized, but a wedge shape has not. We present a model of the acoustic field of a wedge transducer based on the spatial impulse response (SIR) method. A wedge transducer is created by making a cut normal to a spherical cap and keeping the section with the smaller surface area. Analytic expressions for the SIR are derived for an arbitrary point P in space. To find the time domain pressure at P, the SIR is convolved with the time derivative of the wedge surface velocity. Numerical examples of SIRs and the acoustic pressure field are given for a wedge formed from a spherical cap with a geometric focus of 9 cm and an outer diameter of 8 cm. For a single frequency drive of 4.7 MHz, calculations of pressure are compared to hydrophone measurements of a HIFU transducer with equivalent geometry. The results show an excellent agreement between theory and experiment.

THURSDAY AFTERNOON, 1 MAY 2003 ROOMS 110/111, 1:30 TO 3:15 P.M.

Session 4pEA

Engineering Acoustics, Signal Processing in Acoustics, Physical Acoustics, Structural Acoustics and Vibration: Nondestructive Evaluation and Material Characterization

Sally J. Pardue, Chair

Mechanical Engineering, Tennessee Technological University, 115 West 10th Street, Box 5014, Cookeville, Tennessee 38505

Contributed Papers

1:30

4pEA1. Towards a laser-vibrometry technique for measuring railroad rail stress. Vesna Damljanović and Richard L. Weaver (Dept. of Theoret. and Appl. Mech., Univ. of Illinois, 104 S. Wright St., Urbana, IL 61801, damljano@uiuc.edu)

There is a broad consensus on the need for cost-effective and reliable methods for measurement of axial stress in railroad rails. Constrained thermal expansions and contractions induce large forces which in turn lead to bucklings and rail breaks. We are developing a new method for such measurements, based on the decrease (increase) of effective dynamic flexural rigidity under the influence of compressive (tensile) loads. Scanned laser vibrometry measurements of vibration fields at a prescribed frequency, followed by a comparison with guided wave theory for the complex cross section of the rail, allows the extraction of flexural wavenumbers. These wavenumbers are in turn related to contained load and to the rail's intrinsic rigidity. Here we report the ongoing status of this work, describing the results of extensive laboratory measurements on unloaded rail, and test-bay measurements on rails subjected to compressive loads to 100 kip.

1:45

4pEA2. Analysis of digitized on-site ultrasonic state measurements. Sissay Hailu, Dov Hazony (Dept. of EECS, CWRU, Cleveland, OH 44106, dxh2@po.cwru.edu), and Yehonathan Hazony (Boston Univ., Boston, MA 02115)

Of concern is noninvasive *gage-less* monitoring fatigue experiments as well as detection and tracking of cracks, surface, and bulk. Accordingly, a digital signal decomposition method is developed for the analysis of sharp, time-limited ultrasonic pulses traveling through a tensile specimen on-site. The method extracts from the propagating pulses, respective elongation, and diameter reduction due to stress, associated with signal delays and changes in amplitude and shape. The analytical process is based on a *nonlinear multivariate regression analysis*. When applied to the current experiments, the method extracts delay variations accurate to the order of ± 1 ps and quantified signal-deformation data. These results relate to a tensile-stress experiment under a five-cycle strain-cycle experiment, extending some of the cycles well into the plastic regime of a stainless steel specimen. The three-section cylindrical specimen consists of two gripper heads and a slender cylindrical section 6 mm in diameter and 20 mm long. Transducers are implanted on both specimen faces along the main axis [Mostafa *et al.*, Int. J. Fract. **85**, 99–109 (1997)]. [Work supported by NASA.]

2:00

4pEA3. Nonlinear acoustic spectroscopy applied to damage detection of elastic structures. Gerard Vanderborck (Thales Underwater Systems, 525 route des Dolines, BP 157, 06903 Sophia Antipolis Cedex, France, gerard.vanderborck@fr.thalesgroup.com) and Michel Lagier (Actea, 06183 Juan les Pins, France)

The strong linkage between damage and nonlinear elasticity of materials leads to much research in the field of nonlinear response of mechanical structures. The aim of this work is to demonstrate the relevance of nonlinear acoustic spectroscopy for damage detection in elastic structures. The proposed method consists of the observation of linear and nonlinear responses to harmonic excitations of the structure: the damage detection and localization are obtained from an array of sensors. The nonlinear response amplitude is proportional to the local nonlinearity (failure nonlinear elastic response). A dual frequency method has been demonstrated for cracked beam excited by two force sources located at different points; the theoretical analysis comes from the Green function method applied to each overtone displacement field, the crack elastic response is handled by a nonlinear force/displacement behavior. Small scale experiments are presented for aluminum and concrete beams and the experimental results confirm the capability of nonlinear acoustic spectroscopy for detection and monitoring of damaged structures. [Work supported by European Community.]

2:15

4pEA4. Linear and nonlinear angle beam ultrasonic spectroscopy to evaluate adhesive layers. Laszlo Adler, Bin Xie, Vadim Iakovlev (Adler Consultants, Inc., 1275 Kinnear Rd., Columbus, OH 43212, ladler1@aol.com), and Stanislav Rokhlin (Ohio State Univ., Columbus, OH 43210)

A new technique for quantitative evaluation of adhesive bond integrity of metals and composites called angle beam ultrasonic spectroscopy was developed. The novelty of this approach is that it utilizes both normal and obliquely incident ultrasonic beams on the bond line simultaneously and measures the frequency response of the reflected ultrasonic signals. From the frequency dependence of the reflection coefficients, elastic moduli, attenuation coefficients, thickness and density of the adhesive layer can be measured. The method was further enhanced by adding a low-frequency dynamic load acting as a nonlinear (parametric) enhancement. The low-frequency excitation parameters are set to make the maximum stress distribution coincide with the bond line. In addition to the bond layer properties, interfacial spring constants are also obtained by using the nonlinear method. Experimental results on adhesive samples will be presented and discussed with model prediction.

2:30

4pEA5. Influence of acoustic memory on ultrasonic attenuation. Michael McPherson, Igor Ostrovskii, and M. A. Breazeale (Natl. Ctr. for Physical Acoust., Coliseum Dr., University, MS 38677)

Recently a new physical phenomenon of acoustic memory has been described [Phys. Rev. Lett., **89**, 155506-1/3 (2002)]. In this lecture, we show for the first time the connection between acoustic memory and ultrasonic attenuation. Different results are given showing attenuation as a function of temperature, of ultrasound amplitude, and of frequency. We used [001] $LiNbO_3$ single-crystal samples and longitudinal acoustic waves over the frequency range from 10 to 32 MHz. Strong correlation exists between acoustic memory amplitude and the measured coefficient of ultrasonic attenuation. Acoustic memory causes strong variation in attenuation measurements. In this light, we discuss the known difficulties in practical attenuation measurements in ferroelectrics. We mention an unusual temperature dependence, including a strong hysteresis effect. The frequency and amplitude dependence of attenuation are also discussed.

2:45

4pEA6. A comparison of single crystal versus ceramic piezoelectric materials for acoustic applications. James F. Tressler (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, tressler@pa.nrl.navy.mil)

For nearly fifty years, piezoelectric ceramics (primarily from the PZT family) have been the materials of choice as the active elements in sound projectors and receivers, medical ultrasound probes, etc. There is currently great interest in the materials community in the use of newly discovered single crystal relaxor ferroelectric materials as a replacement in applications that currently utilize piezoelectric ceramics. The salient features of single crystal piezoelectrics are strains an order of magnitude larger than are achievable in PZT and electromechanical coupling coefficients on the order of 90 percent. This presentation will provide a more thorough comparison of the physical and piezoelectric properties of single crystal piezoelectrics versus conventional PZT piezoceramics. This will hopefully provide the transducer designer with a better understanding of the pros and cons in the use of single crystals. In addition, a brief review of single crystal synthesis procedures will be described. Finally, some acoustic devices utilizing both single crystals and piezoceramics will be compared. [Work supported by DARPA and ONR Code 321-TS.]

3:00

4pEA7. Acoustical investigations of borate glasses containing oxides of some transition elements and ferric oxide dopants. Surjit Singh Bhatti and Kanwar Jit Singh (Appl. Phys. Dept., Guru Nanak Dev Univ., Amritsar-143005, India, kanwarjitsinghapd@yahoo.com)

Glass samples of manganese oxide borate and zinc oxide borate (with and without ferric oxide doping) have been prepared to study their acoustical, mechanical, and thermal behavior as function of composition. Sound velocities and attenuation measurements in these glass systems at 1,2, and 5 MHz give elastic moduli, Poissons ratio, micro-hardness, acoustic impedance, internal friction, thermal expansion coefficient and Debye and softening temperatures. Structural changes involve boron anomaly, field strengths of cations, difference in ionic radii, and charge state of iron. Makishima–Mackenzie (theoretical model) and IR and NGR techniques confirm the conclusions arrived at. The network modifier (NWM) is varied from 25 to 45 mol % for manganese oxide borate and from 15 to 40 mol % with 10 mol % doping of ferric oxide. For zinc oxide borate glasses, it varies from 26 to 34 mol % and with 10 mol % of ferric oxide, its variation is from 15 to 35 mol %. Impact of doping by ferric oxide on the properties of these glass systems have been investigated.

Session 4pPA

Physical Acoustics: Atmospheric and Seismic Propagation

Vladimir E. Ostashev, Chair
Environmental Technology Laboratory, NOAA, 325 Broadway, Boulder, Colorado 80303

Contributed Papers

1:30

4pPA1. Predicting sound propagation in the atmosphere using an artificial neural network. Michael Mungiole (Army Res. Lab., Attn: AMSRL-CI-EE, 2800 Powder Mill Rd., Adelphi, MD 20783-1197, mmungiole@arl.army.mil) and D. Keith Wilson (Eng. R&D Ctr., U.S. Army CRREL, Hanover, NH 03755-1290)

The spectral content of acoustic signals is dramatically altered as sound propagates through the atmosphere due to refraction from vertical gradients in wind and temperature, ground interactions, and other effects. Parabolic equation (PE) techniques have been successfully used to calculate numerical solutions for many of these effects. The PE models generally produce accurate attenuation values, but execution time is excessive for many applications where near real-time results are required. To obtain sound level attenuation predictions more quickly, we developed an artificial neural network for a range of heights and source/receiver horizontal separations. The PE and boundary conditions were modified to obtain a nondimensional representation, which resulted in seven parameters required to specify all input combinations. This model version was then used to train the neural network, using a range of values for each parameter. The standard deviation of the errors (propagation model minus neural network simulations) was generally within 2 dB for the training data set containing approximately 15 000 cases. The smaller test data sets resulted in errors having standard deviations of approximately 1 dB. The neural network was a good predictor of the sound propagation model results except for small values of the nondimensional ground impedance parameter.

1:45

4pPA2. On a connection between amplitude fluctuations, phase fluctuations, and processing gain. Ronald A. Wagstaff (1 Coliseum Dr., NCPA, Univ. of Mississippi, Univ., MS 38677)

Weston *et al.* [Philos. Trans. R. Soc. London, Ser. A **265**, 595 (1969)] show the remarkable agreement between the spectra for the coefficients of variation of amplitude and phase, i.e., amplitude and phase fluctuations, for a hydrophone in shallow water. Wagstaff [J. Acoust. Soc. Am. **112**, 2422 (2002)] showed the functional similarity between phase angles and pseudo-phase angles for an outdoor microphone. Pseudo-phase angles are amplitude fluctuations that have been scaled appropriately to have a similar functionality in temporally coherent signal processing as phase angles. The concepts of the two previously mentioned references are merged to exploit the similarity of phase fluctuations and amplitude fluctuations to achieve multiplicative pseudo-coherent gain. Gains in excess of 20 log(N) have been achieved (N is the number of samples averaged). 10 log(N) is considered ideal for vector averaging. It is seldom achieved, because of coherent attenuation and cancellation associated with the use of real phase angles. Results are included for wind noise in outdoor measurements. [Work supported by U.S. Army Armament Research Development and Engineering Center.]

2:00

4pPA3. Cramer–Rao lower bounds on the angle-of-arrival estimates for a wave propagating in a random medium: Geometric acoustics regime. Sandra L. Collier (U.S. Army Res. Lab., AMRSL-CI-EE, 2800 Powder Mill Rd., Adelphi, MD 20783-1197, scollier@arl.army.mil) and D. Keith Wilson (Eng. Res. and Development Ctr., U.S. Army Cold Regions Res. and Eng. Lab, Hanover, NH 03755-1290)

In the geometric acoustics regime, a propagating wave is weakly diffracted and weakly scattered by the medium. The variance of the real component of the signal is much less than the variance of the imaginary component, thus the signal may not be modeled as a complex Gaussian random variable (whose real and imaginary components have equal variance), as is often done in the Rytov extension region, where both scattering and diffraction are strong. A statistical model for a signal in the geometric acoustics regime has been previously developed [S. L. Collier and D. K. Wilson, J. Acoust. Soc. Am. **111**, 2379 (2002)] and its properties have been further investigated [S. L. Collier and D. K. Wilson, ASA April 2003 Meeting on Signal Processing (submitted)]. This statistical model is applied here to an acoustic wave propagating in a random medium with fluctuations described by von Kármán's spectrum. Additive white Gaussian noise is also considered. The correlation functions of the phase and log-amplitude fluctuations for a von Kármán spectrum are derived in the geometric acoustics limit. The Cramer–Rao lower bounds (CRLBs) on the angle-of-arrival estimates are calculated assuming multiple unknown parameters. The range dependence of the CRLBs is studied in detail.

2:15

4pPA4. Equations for direct numerical simulation of sound propagation in a moving atmosphere. Vladimir E. Ostashev (Environ. Technol. Lab., 325 Broadway, Boulder, CO 80305), Lanbo Liu, D. Keith Wilson, Mark L. Moran (U.S. Army Cold Regions Res. and Eng. Lab., Hanover, NH 03755), David F. Aldridge (Sandia Natl. Labs., Albuquerque, NM 87185), and David Marlin (U.S. Army Res. Lab., White Sands Missile Range, NM 88002)

Most previous analytical and numerical studies of sound propagation in a moving atmosphere have been based on wave equations for the sound pressure and on various parabolic approximations to the wave equations. However, these equations cannot be used as starting equations for recently proposed direct numerical simulation (DNS) of sound propagation outdoor since such starting equations should be first-order differential equations with respect to time. In the present paper, we derive two closed sets of the first-order differential equations for the sound pressure and fluctuations in medium velocity and density due to a propagating sound wave. These sets can be used as starting equations for DNS of sound propagation in a moving atmosphere. The ranges of applicability of these sets are studied by comparing them with the equations for the sound pressure used previously. Note that both sets can also be employed for analytical studies of sound propagation in a moving atmosphere. Examples of the use of these sets for DNS and analytical studies of sound propagation in a moving

atmosphere are presented. [Work partially supported by a DoD High-Performance Computing Modernization Office grant and U.S. Army Research Office Grant No. DAAG19-01-1-0640.]

2:30

4pPA5. On the use of modal expansions to model broadband propagation in the nighttime boundary layer and other downward refracting atmospheres over lossy ground planes. Roger Waxler (N.C.P.A., Univ. of Mississippi, 1 Coliseum Dr., P.O. Box 1848, University, MS 38677, rwax@olemiss.edu)

A modal expansion for long-range, low-frequency propagation in downward refracting atmospheres over complex impedance planes has been developed. This modal expansion decomposes the sound field in the frequency domain into a ducted part (the modal sum) and an upwardly propagating part (expressed as an integral over a continuum). One may simulate the propagation of a pulse released from a point source by multiplying the Fourier transform of the pulse shape by the frequency domain Greens function and inverting the Fourier transform. At long distances from the source one expects that the sound field is adequately described by the modal sum alone. For sound speeds which are asymptotically (with height) constant this is indeed the case. However, the nighttime boundary layer is characterized by a sound speed which is downward refracting near the ground, increasing up to an inversion point, above which it decreases slowly becoming slightly upward refracting. In this case there are certain exceptional frequencies for which the continuum integral contains a resonant component which propagates to large horizontal distances and must be added to the modal sum.

2:45–3:00 Break

3:00

4pPA6. An analytic model for acoustic scattering from an impedance cylinder placed normal to an impedance plane. Michelle E. Swearingen and David C. Swanson (Penn State Univ., P.O. Box 30, State College, PA 16802, michelle.swearingen@pobox.com)

An analytic model for the scattering of a spherical wave off an infinite right cylinder embedded perpendicularly in a ground surface is developed in cylindrical coordinates. The model is developed to simulate a single tree and is developed as a first piece to creating a model for estimating attenuation in a forest based on scattering from individual tree trunks. Comparisons are made to the plane wave case, the transparent cylinder case, the transparent ground case, and the rigid ground case as a method of theoretically verifying the model. Agreement is excellent for all benchmark cases. Model sensitivity to five parameters is determined, which aids in error analysis, particularly when comparing the model results to experimental data, and offers insight into the inner workings of the model. An experiment was performed to collect real-world data on scattering from a cylinder normal to a ground surface. The data from the experiment is analyzed with a transfer function method into frequency and impulse responses. The model results are compared to the experimental data. Agreement is good below 1500 Hz, and poor for higher frequencies.

3:15

4pPA7. Examining surface sealing and crusting using the acoustic to seismic transfer function. Del Leary, Craig J. Hickey, James M. Sabatier (NCPA—Univ. of Mississippi, Oxford, MS 38677), and David A. DiCarlo (USDA—ARS Natl. Sedimentation Lab.)

Soil sealing is examined by measuring the acoustic to seismic (A/S) transfer function. An A/S transfer function is a swept sine measurement using a suspended loud speaker to impinge acoustic energy from the air onto a soil sample. A laser Doppler vibrometer (LDV) is used to obtain the surface particle velocity as a measure of the seismic energy that has been transferred into the soil. This technique is noncontact and therefore allows successive measurements to be taken in time as the surface crust is formed. Soil samples are rained upon then allowed to dry forming a crust or seal that changes both the stiffness and hydraulic properties of the surface layer. Neshoba soils tested show a quantifiable decrease in the seismic energy transferred, as well as an increase in correlation between successive trials, as the crust forms. Additional measurements done with a submerged transducer show an even greater decrease in surface velocity. Current studies also include an acoustic reflection technique to measure changes in the hydraulic properties. [Work supported by USDA ARS.]

3:30

4pPA8. The measurements of the compressional wave velocity of soils during unconsolidated-undrained triaxial testing. Zhiqu Lu, C. J. Hickey, and J. M. Sabatier (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, Coliseum Dr., University, MS 38677)

In this study, a conventional triaxial cell was modified to measure the compressional wave velocity during a triaxial test. Two air-dry remolded soils taken from the counties in Sharkey and Neshoba, MS, were chosen for the study. Unconsolidated-undrained triaxial tests with pore pressure measurement were carried out. Soil samples were isotropically consolidated up to three different cell pressures, axially compressed with the axial strain up to 22%, and subjected to the unload–reload stress path cycles before and after soil failures. The velocity of the compressional wave in the axial direction as a function of the axial strain was measured along with the measurement of the stress–strain response. A comparison of the load-deformation behavior with the load-acoustic velocity behavior was made. The variation of the acoustic velocity with the effective stresses during the isotropic loading, normally consolidated compression and unload–reload stress path cycles were examined. Several empirical expressions of the compressional wave velocity in terms of the effective stresses and the over-consolidation ratio were proposed and examined with the measured data.

3:45

4pPA9. Coordinate determination of the low-flying object by the acoustical method. Igor L. Oboznenko (Kyiv Polytechnic Univ., Kyiv 03057, Ukraine) and Tran H. Dat (Nagoya Univ., Nagoya 464-8603, Japan)

The acoustical method of passive sonar is proposed for determination of the object flying at low altitudes. The method is based on sound refraction into the lower relatively to the object medium. The Rayleigh waves are used with the speed far higher than the low-flying noisy target. Besides the spherical wave, the side wave with attenuation in depth is reflected to the receiver. The latter can be clearly detected on the depth of the surface half-wave that in reality for the infralow frequency amounts up to hundreds of meters. The location finding of the object is obtained via creation of the static fan of the polar pattern with the specified angular resolution of each radar view. At the horizontal range (100 km), depth of immersion multiple of N receivers (horizontal collinear equidistant antenna) and the height of the flying object 3–10 m, in the wave field the correction to the ray acoustics, which is very significant in its amplitude, is done. To count horizontal range to the object, a vectorial (three components) displacement (or vibrating speed) receiver, placed into the center of the radar array is used. The model experiments in water match the calculations.

Session 4pSA

Structural Acoustics and Vibration and Musical Acoustics: Structural Acoustics of Musical Instruments

Jeffrey S. Vipperman, Chair
Department of Mechanical Engineering, University of Pittsburgh, 648 Bennedum Hall, 3700 O'Hara, Pittsburgh, Pennsylvania 15261

Chair's Introduction—1:30

Invited Papers

1:40

4pSA1. Structural acoustics for the violin. George Bissinger (Phys. Dept., East Carolina Univ., Greenville, NC 27858) and Earl G. Williams (Naval Res. Lab., Washington, DC 20375-5350)

The violin is a long-standing problem in acoustics due to the complexity of its shape, the interactivity of its various modes of response, and the inherent complications of coupling these to the acoustic field, independent of any quality aspects. By combining zero-mass loading modal analysis of the violin with simultaneous acoustic measurements over a sphere in an anechoic chamber, major acoustic energy sources were identified among the coterie of corpus (top+back+ribs) modes and interior cavity (via two ports, or f-holes) modes up to 4 kHz. Direct radiation from substructures such as the neck-fingerboard, bridge, and tailpiece was not significant. For each mode the total damping and the radiation efficiency and damping were computed, along with a simple top-back radiation directivity ratio using rms radiativities. The fraction-of-vibrational-energy-radiated was estimated from the radiation-to-total-damping ratio. A new addition to the modal violin acoustics investigations was the application of near-field patch holography to help determine the relative contributions of corpus and port radiation to overall violin radiativity. [Work supported by NSF and ONR.]

2:00

4pSA2. Analysis of the radiation from the violin f-holes using patch near field acoustical holography. Earl G. Williams (Naval Res. Lab., Washington, DC 20375, williams@pa.nrl.navy.mil) and George Bissinger (East Carolina Univ., Greenville, NC 27858)

Although differential pressure measurements offer a direct means to understand the energy flow from the f-holes of the violin they have been performed only at discrete frequencies over relatively limited portions of the acoustic field, and none have ever covered an entire f-hole over a broad frequency region. Application of recently developed near field acoustical holography (NAH) patch processing techniques to 108-node planar rectangular grid microphone data provides a powerful tool to understand the flow of acoustic energy from the f-holes up to 4 kHz. The grid covered each f-hole as well as a small portion of the violin top-plate and provided the necessary spatial resolution to allow isolation of only f-hole aperture radiation in the NAH processing. The projected radiativity in the far field at 1.2 m from just the f-holes was compared with prior microphone measurements in an anechoic chamber over an entire sphere around the violin. As expected the lowest cavity mode $A0$ was the major radiator at the frequencies below all the corpus modes. Surprisingly the first corpus bending modes appear to radiate strongly through the f-holes also. [Work supported by ONR and NSF.]

2:20

4pSA3. A structural dynamics and experimental investigation of the American five-string banjo. Joe Dickey (Ctr. for Non Destructive Evaluation, Johns Hopkins Univ., Baltimore, MD 21218) and Ray Wakland (Penn State Univ., University Park, PA 16804)

The American five-string banjo is unique among musical instruments in that many significant parameters that effect tone are easily adjusted. This is probably why so many banjo players fiddle with their banjo. The instrument is a combination of canonical vibrating systems, i.e., plucked strings that drive a circular, radiating, membrane. A structural dynamics model and experiment are used to characterize the sound and relate changes in sound to setup parameters. Three figures-of-merit, FOMs, are defined; they are loudness, brightness, and decay rate of the sound. The effects of a number of parameters on the FOMs are investigated analytically and experimentally. Among these are the loss factor and tension of the membrane, mass of the bridge, and the location on the string of the excitation. It is noted that the calculated effects of the changes agree with generally accepted setup practices.

2:40

4pSA4. The application of smart structures toward feedback suppression in amplified acoustic guitars. Steven F. Griffin (3112 Dakota St. NE, Albuquerque, NM 87110), Steven A. Lane (Airforce Res. Lab., Kirtland AFB, NM), and Robert L. Clark (Duke Univ., Durham, NC)

Smart structures technology can be applied to amplified acoustic guitars to prevent instability resulting from acoustic feedback. This work presents a coupled model of the guitar dynamics and the acoustic feedback mechanism, and explains how a simple control loop using a piezoceramic actuator can be used to reduce the effects of acoustic feedback. In addition to model simulations, experimental results using a real system and a simple controller are presented. The results show that a significantly higher (7-dB) guitar output can be achieved before instability, without detrimentally affecting the amplified and un-amplified guitar response.

3:00

4pSA5. SEA applications to wind instruments. Peter L. Hoekje (Dept. of Phys., Baldwin–Wallace College, Berea, OH 44017, phoekje@bw.edu)

The behavior of wind instruments, including brass instruments, is primarily determined by the shapes of their air columns, and their interaction with the sound generation mechanism. However, the influence of the surrounding body of the instrument has been a matter of some debate, and papers exploring this question have been published since the early years of the J. Acoust Soc. Am. An apparent correlation between instrument material and playing behavior is disputed by arguments that the structure is stiff and massive compared to the air inside, and that many of the apparent effects are linked to machining differences among materials. The complexity of the instrument body makes this problem well suited for Statistical Energy Analysis (SEA), which treats the air column and the external structure as coupled statistical subsystems that share energy. For trumpets and trombones, the power radiated from the structural vibrations is about 40 dB lower than the energy radiated directly from the air column, with an enhancement at high frequencies due in part to the increasing modal density of the three dimensional structure. The coupling to the structural vibrations themselves from the player's lips and from the air vibrations are similar to each other in magnitude.

3:20–3:35 Break

3:35

4pSA6. Modal analysis and sound radiation from Caribbean steelpans. Andrew Morrison (Dept. of Phys., Northern Illinois Univ., DeKalb, IL 60115)

The Caribbean steelpan is one of the most interesting acoustic musical instruments invented in the last century. Although simple in design, the acoustic properties of the steelpan are surprisingly complicated. Holographic interferometry was used to determine the modes of vibration of a double second and a low tenor steelpan. Sound intensity measurements were taken to explore the relationship between the modes of vibration and the radiated sound field. The pan was placed in an anechoic chamber, and selected notes were excited electromagnetically with a swept sinusoid signal. A two-microphone probe was used to gather sound intensity measurements. Sound intensity maps and animations were constructed for the first three harmonics.

Contributed Papers

3:55

4pSA7. The effects of bell vibrations on the acoustic spectrum of the trumpet. Thomas R. Moore (Dept. of Phys., Rollins College, Winter Park, FL 32789, tmoore@rollins.edu)

The acoustic spectrum of a modern trumpet with the bell section heavily damped has been compared to the spectrum of the same instrument with the bell section left free to vibrate. The amplitude of vibration of the metal was measured in both cases and was shown to be significantly different between the two sets of measurements. Artificial lips were used to ensure consistency between trials. A significant change in the acoustic spectrum between the two cases is found, with the variation being largest in the lower harmonics where the relative power may change by as much as a factor of 2. It is shown that the changes can be explained by a variation in the viscous boundary layer that is attributable to the vibrating walls of the bell. [Work supported by a grant from the Jessie Ball duPont Fund.]

4:10

4pSA8. Structural dynamics of concert harps with wooden and composite material soundboards. Melinda Carney and Thomas J. Royston (Univ. of Illinois at Chicago, 842 W. Taylor St. MC 251, Chicago, IL 60607, troyston@uic.edu)

The replacement of a Sitka spruce grand concert harp soundboard with a carbon fiber-reinforced plastic soundboard could provide improved durability and long-term stability. Experimental vibratory studies on concert harps with wooden soundboards are reviewed. A computational finite-element model is used to identify critical material properties by matching its predictions to the experimental data. With the material properties identified in the finite-element model, the lay-up of the composite soundboard is created using matching criteria based on research of wood replacements for violin top plates. The composite lay-up is then incorporated into the finite-element model, verifying that the dynamic response closely approximates that of the wooden soundboard. The identification technique and composite replacement design process may be applicable to other musical instruments, as well as other nonmusical, wooden plate structures.

4:25

4pSA9. An experimental study of changes in the impulse response of a wood plate that is subject to vibrational stimulus. Jared Grogan, Michael Braunstein, and Andrew Piacsek (Dept. of Phys., Central Washington Univ., Ellensburg, WA 98926)

It is a well-known dictum among players of stringed instruments that the tone of a new instrument improves with playing and that a fine instrument needs to be played if it is to maintain its optimum sound quality. This process is sometimes referred to as ''playing in'' an instrument. There is scant mention in the scientific literature, however, of a quantitative analysis of this phenomenon. As a first step in rigorously testing this hypothesis, measurements were made of tap tones of rectangular pieces of thin spruce before and after they were subjected to vibrational stimulus. Four spruce rectangles (20x28 cm) were cut from a single sheet obtained from a luthier supplier; three of these were stimulated at different amplitudes, while the fourth was a control plate. The stimulus (provided by a harmonically

4p THU. PM

driven guitar string connected to the plate via a bridge) lasted approximately 10 weeks, during which time tap tones of all four plates were periodically recorded. Spectrograms of the tap tones are compared among the plates and over time. A preliminary analysis of the data does not reveal any significant changes in the acoustic response of the plates.

4:40

4pSA10. Effect of body shape on vibration of electric guitars. Daniel A. Russell, Wesley S. Haveman, Willis Broden, and N. Pontus Weibull (Sci. & Math. Dept., Kettering Univ., 1700 W. Third Ave., Flint, MI 48504, drussell@kettering.edu)

The body vibrations of an electric guitar are typically ignored since the string vibrations are converted to sound through the use of a magnetic pickup. However, vibrations in the neck have been shown to cause dead spots at certain fret positions [H. Fleischer, J. Acoust. Soc. Am. **105**, 1330 (1999)]. In this paper we compare the vibrational mode shapes and frequencies of three electric guitars with different body shapes. Two guitars are solid-body electrics: one with a body shape which is symmetric about the neck axis (Epiphone Coronet) and the other which is not (Gibson Explorer). Mode shapes and frequencies are considerably different for the body, though neck vibrations are more closely related. The third guitar is an arched top hollow-body electric (Gibson ES-335). For this guitar, the top and back plates and the air cavities may also contribute to the guitar sound quality. Mode shapes and frequencies are determined from experimental modal analysis using an impact hammer and accelerometer.

4:55

4pSA11. On the scaling of free-reed organ pipes: A comparison of 19th-century theories to modern nonlinear models. Jonas Braasch (Institut für Kommunikationsakustik, Ruhr-Universität Bochum, 44780 Bochum, Germany)

After their introduction in the 19th century, the scaling of free reed organ pipes was often done on the basis of investigations by W. Weber, J. G. Töpfer, and F. Haas. Especially, Töpfer is often named as the founder of mathematical/physical oriented scaling. Unfortunately, it was impossible in the 19th century to derive a number of measures (e.g., optimal lengths of a conical pipe and the boot) theoretically. In this presentation it is demonstrated how recent theories can be used to derive some of those measures. For this purpose, a one-dimensional physical model was developed. The model uses the reed generator developed by Tarnopolsky *et al.* [J. Acoust. Soc. Am. **108**, 400–406 (2001)] and a wave-guide model for the resonator so as to determine the optimal resonator and boot dimensions. Wrong assumptions that were made in the 19th century will be also addressed. For example, organ builders frequently used the solution of Weber to determine the optimal resonator length. The solution of Weber was aimed to make the reed pipe's frequency independent from temperature and blowing pressure, but what the organ builders would have needed is a pipe whose frequency is mainly determined by the resonator length, so that the pipes would detune with the flue pipes, when the temperature changes.

THURSDAY AFTERNOON, 1 MAY 2003 — ROOM 205, 2:00 TO 4:40 P.M.

Session 4pSC

Speech Communication and Psychological and Physiological Acoustics: The Funding Game—Strategies for Different Federal Agencies for Behavioral Sciences

Astrid Schmidt-Nielsen, Cochair
Office of Naval Research, Code 342, 800 North Quincy Street, Arlington, Virginia 22217-5660

Patrice S. Beddor, Cochair
University of Michigan, 1076 Frieze Building, Ann Arbor, Michigan 48109-1285

Chair's Introduction—2:00

Invited Papers

2:05

4pSC1. Merit-based funding at the National Science Foundation. Mary P. Harper (Natl. Sci. Foundation, 4201 Wilson Blvd., Arlington, VA 22230, mharper@nsf.gov)

National Science Foundation (NSF) invests in the best ideas from the most capable people as determined by competitive merit review. This talk will first summarize how NSF evaluates proposals for research and education projects in speech communication. The review timeline, as well as the merit review criteria of intellectual merit and broader impacts, will be discussed. Finally, a summary of current and future NSF funding opportunities will be provided.

2:20

4pSC2. NSF funding opportunities in the behavioral and cognitive sciences. Philip Rubin (Natl. Sci. Foundation, 4201 Wilson Blvd., Ste. 995, Arlington, VA 22230, prubin@nsf.gov)

This presentation will provide a brief overview of existing and emerging funding opportunities at the National Science Foundation (NSF). The Division of Behavioral and Cognitive Sciences (BCS) at the NSF supports research to develop and advance scientific knowledge focusing on human cognition, language, social behavior and culture, as well as research on the interactions between human societies and the physical environment. BCS programs consider proposals that fall squarely within disciplines, but they also encourage and support interdisciplinary projects, which are evaluated through joint review among Programs in BCS, as well as a joint review with programs in other Divisions, and NSF-wide multi-disciplinary panels, as appropriate. All programs in BCS consider proposals for

research projects, conferences, and workshops. Some programs also consider proposals for doctoral dissertation improvement assistance, the acquisition of specialized research and computing equipment, group international travel, and large-scale data collection. BCS participates in special initiatives and competitions on a number of topics, including cognitive neuroscience, children's research, human origins, and the development of infrastructure to improve data resources, data archives, collaboratories, and centers.

2:30

4pSC3. Seeking NIH funding: Defining the process. Lana Shekim (Natl. Institutes of Health, Bethesda, MD 20892)

The presentation will provide a brief introduction to the National Institute on Deafness and other Communication Disorders (NIDCD) with emphasis on the Voice and Speech program in the Division of Extramural Research. The process of seeking NIH funding will be outlined and a number of funding mechanisms will be described. The peer review process and the time course of a grant application will be highlighted.

2:40

4pSC4. Training camp: The quest to become a new National Institutes of Health (NIH)-funded independent investigator. Daniel A. Sklare (SPB/DER, NIDCD, NIH, EPS-400C, 6120 Executive Blvd., MSC-7180, Rockville, MD 20892-7180)

This presentation will provide information on the research training and career development programs of the National Institute on Deafness and Other Communication Disorders (NIDCD). The predoctoral and postdoctoral fellowship (F30, F31, F32) programs and the research career development awards for clinically trained individuals (K08/K23) and for individuals trained in the quantitative sciences and in engineering (K25) will be highlighted. In addition, the role of the NIDCD Small Grant (R03) in transitioning postdoctoral-level investigators to research independence will be underscored.

2:50

4pSC5. Office of Naval Research funding strategies. Astrid Schmidt-Nielsen (ONR, Code 342, 800 N. Quincy St., Arlington, VA 22217-5660, schmida@onr.navy.mil)

The best strategies for obtaining military research funding can be quite different from NIH and NSF. Funding opportunities at the Office of Naval Research will be discussed. Different types of Navy/military funding will be described, and strategies for obtaining various types of funding, including the Young Investigator Program, will be discussed.

3:00

4pSC6. Air Force program in hearing research. Willard D. Larkin (Air Force Office of Sci. Res., AFOSR/NL, Rm. 713, 4015 Wilson Blvd., Arlington, VA 22203-1954, willard.larkin@afosr.af.mil)

The Air Force Office of Scientific Research (AFOSR) manages all the basic science activity for the U.S. Air Force. Research is supported in the Air Force Laboratory, and grants are provided to universities and research institutions. This presentation will outline the scope of AFOSR's support for basic research in areas germane to human hearing. Topics of current interest include the complex of problems associated with spatial hearing, the psychophysical and perceptual foundations for the design of auditory displays, problems of auditory information overload, and neurologically informed computational models of auditory perceptual processing. To develop better methods of hearing protection, AFOSR also supports efforts to understand the biophysical basis of cochlear excitation via bone conduction. Opportunities to contribute to these and related topics will be described.

3:10–3:40
Panel Discussion

3:40–4:40
Breakout Sessions

Session 4pUW

Underwater Acoustics and Acoustical Oceanography: High Frequency Sediment Acoustics II and Ambient Noise

Eric I. Thorsos, Chair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, Washington 98105-5598

Contributed Papers

1:00

4pUW1. *In situ* measurement of geoacoustic sediment properties: An example from the ONR Mine Burial Program, Martha's Vineyard Coastal Observatory. Barbara J. Kraft, Larry A. Mayer (Ctr. for Coastal and Ocean Mapping, 24 Colovos Rd., Durham, NH 03824), Peter G. Simpkin (IKB Technologies Ltd., Bedford, NS B4B 1B4, Canada), and John A. Goff (Univ. of Texas Inst. for Geophys., Austin, TX 78759)

In support of the Office of Naval Research's Mine Burial Program (MBP), *in situ* acoustic and resistivity measurements were obtained using ISSAP, a device developed and built by the Center for Coastal and Ocean Mapping. One of the field areas selected for the MBP experiments is the WHOI coastal observatory based off Martha's Vineyard. This area is an active natural laboratory that will provide an ideal environment for testing and observing mine migration and burial patterns due to temporal seabed processes. Seawater and surficial sediment measurements of compressional wave sound speed, attenuation, and resistivity were obtained at 87 stations. The ISSAP instrument used four transducer probes arranged in a square pattern giving acoustic path lengths of 30 and 20 cm with a maximum insertion depth of 15 cm. Transducers operated at a frequency of 65 kHz. The received acoustic signal was sampled at a frequency of 5 MHz. A measurement cycle was completed by transmitting 10 pulses on each of the five paths and repeating three times for a total 150 measurements. Resistivity measurements were obtained from two probes mounted on ISSAP following completion of the acoustic measurements. [Research supported by ONR Grant Nos. N00014-00-1-0821 and N00014-02-1-0138.]

1:15

4pUW2. Development of an impedance tube technique for *in-situ* classification of marine sediments. Preston S. Wilson, Ronald A. Roy, and William M. Carey (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215)

The removal of samples or the insertion of measuring devices into the ocean bottom can alter the acoustic behavior of these sedimentary materials. A less invasive method for the acoustic characterization of marine sediments in the 1 to 10 kHz frequency range has been investigated. A water-filled impedance tube has been used to measure the complex reflection coefficient at the surface of a sediment layer at the bottom of a water tank. The acoustic properties of the sediment were obtained as a function of frequency by an inversion process. The system is modeled numerically and an effective fluid description of the sediment is used. The difference between the measured and predicted complex reflection coefficient is minimized through variation of the sediment sound speed and attenuation in the model. Results will be presented for different sediments. [Work supported by U. S. Navy ONR.]

1:30

4pUW3. High-frequency seafloor studies using buried directional receivers. Anthony P. Lyons (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804-0030, apl2@psu.edu) and John C. Osler (Defence R&D Canada Atlantic, Dartmouth, NS B2Y 3Z7, Canada)

Knowledge of the acoustic arrival angle can be useful for studying penetration mechanisms and for estimating sediment sound speed dispersion. The arrival angle of acoustic energy penetrating the sediment water interface, however, is difficult to measure using sparsely distributed pressure sensors. The arrival angle, as well as amplitude information, can be unambiguously obtained by measuring particle motion with directional receivers. An experiment was conducted off of Elba Island in 1999 to assess the feasibility of using of high-frequency accelerometers to measure directionality. Measurements of acoustic penetration onto a sandy seafloor were obtained over wide frequency range (2.5 to 29 kHz) using the accelerometers. Results from the experiment demonstrating this novel measurement technique will be presented, as will issues concerning accelerometer design and calibration.

1:45

4pUW4. Use of a buried array to characterize sediment volume scattering. Darrell R. Jackson, Kevin L. Williams, Anatoliy N. Ivakin, and Eric I. Thorsos (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

During the SAX99 experiment, a buried array was used to measure the sound field due to sources placed in the water column. Measurements were made over the frequency range 11–50 kHz. The sound field at the array showed a ''coda'' that was longer than suggested by simulations of rough-interface scattering. Simulations have been used to show that scattering by sediment volume heterogeneities is the likely cause of the coda. The volume scattering strength has been estimated by fitting a point scatterer model to the data, and the strongest discrete scatterers have been located and characterized by means of backpropagation. When the measured properties of the volume scatterers are inserted in backscattering models, the predicted backscattering strengths fall somewhat below measured values, indicating that roughness scattering is dominant at this (sandy) site. [Work supported by ONR.]

2:00

4pUW5. Acoustic imaging of small water-lain sand deposits. Max Deffenbaugh, Neal L. Adair, David C. J. D. Hoyal, and David E. Giffin (ExxonMobil Upstream Res. Co., P.O. Box 2189, Houston, TX 77252)

Reduced-scale physical modeling of depositional systems, like submarine fans, river deltas, and point bars, provides insight into the formation and internal structure of the full-scale systems which may become economic hydrocarbon reservoirs. Turbid water with controlled sediment concentration and flow velocity is discharged into a 3 m$\times$5 m tank of still water to create deposits up to typically 10 cm thick. These deposits are imaged by a pencil-beam high-frequency (7 MHz) acoustic system to cap-

ture the evolution of deposit topography and by a broad-beam lower-frequency system (150 kHz) to image an internal structure. An x–y positioning system moves the transducer to create detailed 3D images. At 7 MHz, the deposit surface is ''rough,'' so significant backscattered energy is detected even for non-normal incidence. This, together with the narrow beamwidth, allows the deposit elevation directly below the sensor to be measured independent of the local slope. The deposit surface is ''smooth'' to the 150 kHz system, so reflections come only from points of normal incidence. This makes imaging more complicated, but the lower frequencies penetrate the deposit and reveal some internal structure. Images from both systems will be shown and compared.

2:15

4pUW6. Porosity and grain size dependence of the longitudinal wave velocity of water-saturated beach sand. Masao Kimura and Masahiro Noguchi (Dept. of Geo-Environ. Technol., Tokai Univ., 3-20-1 Orido, Shimizu, Shizuoka 424-8610, Japan, mkimura@scc.u-tokai.ac.jp)

The longitudinal wave velocity of water-saturated sand is dependent on the porosity. The data which show the relationship between the velocity and the porosity are dispersed [E. L. Hamilton and R. T. Bachman, J. Acoust. Soc. Am. **72**, 1891–1904 (1982)]. It seems that this dispersion is due to the grain size, the standard deviation of the grain size, and the grain shape. In this study, to investigate the dispersion, the longitudinal wave velocities, the porosities, and the grain sizes of many kinds of water-saturated beach sands are measured. The relationships between the velocity, the porosity, and the grain size are obtained. From these results, it is seen that the velocity of the water-saturated beach sand with the same porosity varies with the grain size. That is, the velocity of the water-saturated beach sand with the same porosity increases, as the grain size increases. It is considered that the frame bulk modulus of the water-saturated beach sand with the same porosity varies with the grain size.

2:30

4pUW7. Tomographic measurements of sandy sediment sound speed and attenuation. Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

During the SAX99, a tomographic system called the Acoustic Imager (AI) was developed to measure sound speed, attenuation and their variability in sandy sediments. It consists of 60 transducers arranged in a ring with a diameter of 29 cm placed in the horizontal plane. When the ring is pressed into the sediment at a certain depth, a set of sound speed and attenuation measurements is made. Taking data at different depths, a 3-D view of the sediment sound speed and attenuation can be obtained. In this paper, two kinds of measurements are reported. The first is measurements of sound speed at different locations over a depth of 10 cm with measurements taken every two cm. The second is a measurement at one location and one depth over a period of 48 hours to examine temporal variation of sediments. [Work supported by ONR.]

2:45

4pUW8. High frequency scattering, attenuation and permeability in sands. Tokuo Yamamoto (Geoacoustic Lab., Appl. Marine Phys. Div. RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149), Junichi Sakakibara (Geoacoustic Lab., Taito-ku, Tokyo 111-0051, Japan), and Yoshiyuki Mohri (Natl. Inst. for Rural Eng., Tsukuba, Japan)

Imaging the permeability within sediments is one of the important problems that is not well investigated in geoacoustics. Acoustic wave energy scattered from velocity and density fluctuations within sediments increases the attenuation. However, only the intrinsic attenuation due to the Biot mechanism is related to permeability. We investigate this problem using the crosswell acoustic tomography and simple propagation measurements in water saturated sands in a large tank and a pond. The range of frequency is from 10 to 100 kHz. The sands used in the experiments are of uniform grain sizes of 0.05 to 0.5 mm. We find that the wave propagation data follows the Biot theory. The apparent attenuation due to scattering is proportional to the velocity fluctuations and can be significant. For accurate imaging of permeability in sand volume, the removal of the apparent attenuation due to volume scattering from total attenuation is very important. [Work supported by ONR Code 321OA, NIRE, and MEST.]

3:00–3:15 Break

3:15

4pUW9. High frequency bottom scattering: Theory and experiment. Tokuo Yamamoto, Keiichi Ohkawa, and Haruhiko Yamaoka (Geoacoustic Lab., Appl. Marine Phys. Div. RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149)

The exact solution of the Biot theory for the transmission of an acoustic wave at the fluid–porous seabed interface is obtained in closed form. For grazing angles larger than 5 degrees, the transmission effect on volume scattering is *ca*. 20 dB larger than that of a fluid–fluid bottom model. If grain size and porosity is known, the Biot model calculates the attenuation. A Biot scattering model is constructed by replacing the transmission coefficients and attenuation of the fluid–fluid bottom model by Yamamoto (1996). During the Shallow Water Acoustic Technology (SWAT) 2000 experiments, the 5.5 kHz 3D backscattering data from the sandy bottom at the SS site were obtained using a self-contained UM 32 channel bi-linear hydrophone array rigidly mounted on a 5 m high rigid metal tower. The Biot model extracted, sound velocity, aspect ratio, and backscatter model parameters. The Biot model is also applied to the 15 kHz 3D backscattering data from a clay bottom. For this case, the difference between the Biot model and the fluid–fluid bottom was small. It is concluded that the Exact Biot model should be used when dealing with the high frequency bottom backscattering or forward scattering from the sandy bottom. [Work supported by ONR Code 321OA.]

3:30

4pUW10. Assessing the sediment volume contribution to scattering in SAX99 sediment: Sound-speed fluctuations. Kevin B. Briggs (Naval Res. Lab., Seafloor Sci. Branch, Stennis Space Center, MS 39529-5004) and Dajun Tang (Univ. of Washington, Seattle, WA 98105)

A simulation experiment using Monte Carlo realizations has been undertaken based on real variations in sound-speed and bulk density data from SAX99 sand sediment. In this investigation, sound-speed profiles measured from cores are used to estimate the sound-speed power spectra and sound-speed correlation lengths. Due to the manner in which sound speed is measured from 6.1-cm-diam cores at 1-cm intervals, measured sound speed is an average value across the core. Thus, the values of the measurements as well as the estimates of the correlation lengths may be a function of the sampling interval and estimation of correlation lengths may be distorted. After realizations are generated from the sound-speed power spectrum and correlation length, the virtual core is sampled to obtain a sound-speed profile and then the power spectrum is re-estimated. Because actual sound speed is known from the original data in the simulation, the difference between the parameters used to generate the simulation and the virtual parameters is a measure of the distortion by averaging or smoothing. In addition, the cross-correlation between sediment bulk density and sound-speed fluctuations in the same cores is shown. [Work sponsored by ONR.]

3:45

4pUW11. Models of discrete scattering in marine sediments. Anatoliy N. Ivakin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ivakin@apl.washington.edu)

Acoustic scattering in marine sediments with discrete inclusions, such as shell fragments and rocks, is described in terms of the scattering cross section per unit sediment volume and the scattering cross section per unit seabed area. These cross sections are expressed through the individual scattering functions of discrete targets and statistical distributions of their parameters (size, shape, orientation, material, etc). Some simplified approaches for description of the individual scattering from irregular inclusions are considered. Rigorously, individual scattering functions should take into account the influence of the water–sediment interface. Effects which result from this influence are discussed. The role of the size distribution function is analyzed. Two types of distributions are considered. The first one is a narrow distribution with one dominating scale. The second type is a wide or multiscale distribution, for example, described by a power-law function. The frequency-angular dependencies of the seabed scattering strength for various models of discrete scattering are calculated and discussed. [Work supported by ONR, Ocean Acoustics.]

4:00

4pUW12. Modeling the sensitivity of sediment backscattering measurements to the underlying variability of the sediment bulk environmental characterizations and inhomogeneity spectra. Kevin D. LePage (Naval Res. Lab., Code 7144, 4555 Overlook Ave. SW, Washington, DC 20375), Charles W. Holland (The Penn State Univ., State College, PA 16804), and Henrik Schmidt (MIT, Cambridge, MA 02139)

A significant part of the uncertainty in reverberation predictions for shallow-water environments lies in the variability of the scattering strength associated with the bottom sediments. In this paper we evaluate how variability in the background properties of these sediments, and the characterizations of the inhomogeneity distributions within these sediments is propagated through to uncertainty in the sediment scattering strength. We also address the uncertainty in the measurement problem, i.e., how precise is the determination of the sediment bulk and inhomogeneity properties from local measurements of the time-angle evolution of bottom backscatter. We take as an example an environmental characterization obtained for a site on the Malta Plateau, south of Sicily. Here both the reflection properties of the seafloor and the time-angle evolution of the backscatter have been used to generate a notional environmental characterization of the bulk and inhomogeneity spectra of the bottom. We use the OASES volume scattering model to address the sensitivity and uncertainty issues surrounding the interpretation of this data set and to address the implications on the resulting reverberation uncertainty. [Work supported by the ONR Capturing Uncertainty DRI.]

4:15

4pUW13. Scattering of plane evanescent waves by buried cylinders: Modeling the coupling to guided waves and resonances. Philip L. Marston (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

The coupling of sound to buried targets can be associated with acoustic evanescent waves when the sea bottom is smooth. To understand the excitation of guided waves on buried fluid cylinders and shells by acoustic evanescent waves and the associated target resonances, the two-dimensional partial wave series for the scattering is found for normal incidence in an unbounded medium. The shell formulation uses the simplifications of thin-shell dynamics. The expansion of the incident wave becomes a double summation with products of modified and ordinary Bessel functions [P. L. Marston, J. Acoust. Soc. Am. **111**, 2378 (2002)]. Unlike the case of an ordinary incident wave, the counterpropagating partial waves of the same angular order have unequal magnitudes when the incident wave is evanescent. This is a consequence of the exponential dependence of the incident wave amplitude on depth. Some consequences of this imbalance of partial-wave amplitudes are given by modifying previous ray theory for the scattering [P. L. Marston and N. H. Sun, J. Acoust. Soc. Am. **97**, 777–783 (1995)]. The exponential dependence of the scattering on the location of a scatterer was previously demonstrated in air [T. J. Matula and P. L. Marston, J. Acoust. Soc. Am. **93**, 1192–1195 (1993)].

4:30

4pUW14. Ambient noise analysis of underwater acoustic data. Mark A. Snyder, Pete Orlin, Annette Schulte (Naval Oceanogr. Office, Stennis Space Ctr., MS 39522), and Joal Newcomb (Naval Res. Lab., Stennis Space Ctr., MS 39529)

The Littoral Acoustic Demonstration Center (LADC) deployed three Environmental Acoustic Recording System (EARS) buoys in the northern Gulf of Mexico during the summers of 2001 and 2002. The buoys recorded frequencies up to 5859 Hz continuously for 36 days in 2001 and for 72 days in 2002. The acoustic signals recorded include sperm whale vocalizations, seismic airguns, and shipping traffic. The variability of the ambient noise is analyzed using spectrograms, time series, and statistical measurements. Variations in ambient noise before, during, and after tropical storm/hurricane passage are also investigated.

4:45

4pUW15. Long-term noise statistics from the Gulf of Mexico. Anthony I. Eller (Sci. Applications Intl., Inc., 1710 Saic Dr., McLean, VA 22102, anthony.i.eller@saic.com), George E. Ioup, Juliette W. Ioup, and James P. Larue (Univ. of New Orleans, New Orleans, LA 70148)

Long-term, omnidirectional acoustic noise measurements were conducted in the northeastern Gulf of Mexico during the summer of 2001. These efforts were a part of the Littoral Acoustic Demonstration Center project, Phase I. Initial looks at the noise time series, processed in standard one-third-octave bands from 10 to 5000 Hz, show noise levels that differ substantially from customary deep-water noise spectra. Contributing factors to this highly dynamic noise environment are an abundance of marine mammal emissions and various industrial noises. Results presented here address long-term temporal variability, temporal coherence times, the fluctuation spectrum, and coherence of fluctuations across the frequency spectrum. [Research supported by ONR.]

5:00

4pUW16. Underwater sound from the whale's point of view. Paul T. Arveson (Balanced Scorecard Inst., 6902 Breezewood Terrace, Rockville, MD 20852)

There have been numerous reports in the recent literature of apparently stressful effects on marine mammals due to sonar experiments. But another man-made source—the radiated noise from ships—contributes significantly to the ocean ambient, nearly everywhere and all the time. The technical basis for this talk is a set of accurate and detailed measurements of the radiated noise of a typical cargoship [P. Arveson and D. Vendittis, "Radiated noise characteristics of a large cargo ship," J. Acoust. Soc. Am. (2000)]. However, the talk will be a popular-level demonstration and a (necessarily) fictitious narrative of acoustical experiences from a humpback whale's point of view. Room acoustics permitting, the audience should be able to gain an experiential insight into the environmental impact of shipping noise on the life and habits of these creatures.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

G. S. K. Wong, Chair S1
Institute for National Measurement Standards, National Research Council, Montreal Road, Bldg. M36, Ottawa, Ontario K1A OR6, Canada

J. Seiler, Vice Chair S1
U.S. Department of Labor, Mine Safety and Health Admin., P.O. Box 18233, Bldg. 38, Cochrans Mill Road, Pittsburgh, Pennsylvania 15236

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound, etc. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged. *People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 8:00 a.m. on Thursday, 1 May 2003.*

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance, and comfort.

THURSDAY AFTERNOON, 1 MAY 2002 ROOM 207 3:30 TO 5:00 P.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

to be held jointly with the

ANSI-Acredited U.S. Technical Advisory Group (TAG) Meeting for: ISO/TC 108/SC 4 Human Exposure to Mechanical Vibration and Shock

R. F. Burkard, Chair S3
Hearing Research Laboratory, State University of New York at Buffalo, 215 Parker Hall, Buffalo, New York 14214

C. Champlin, Vice Chair S3
University of Texas, Department of Communication Sciences & Disorders, CMA 2-200, Austin, Texas 78712

D. D. Reynolds, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 4, Human Exposure to Mechanical Vibration and Shock
3939 Briar Crest Court, Las Vegas, Nevada 89120

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. There will be a report on the interface of S3 activities with those of ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock, including plans for future meetings of ISO/TC108/SC 4. The US Technical Advisory Group for TC 108/SC 4 consists of members of S3 and other persons not necessarily members of this Committee. Open discussion of committee reports is encouraged. *People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 8:00 a.m. on Thursday, 1 May 2003.*

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics, shock, and vibration which pertain to biological safety, tolerance, and comfort.