

Session 4aAA

Architectural Acoustics: Absorption, Modeling and Case Studies

Christopher Brooks, Chair

Orpheus Acoustics, 925 Virginia Avenue, Lancaster, Pennsylvania 17603

Chair's Introduction—8:30

Contributed Papers

8:35

4aAA1. Sound absorption characteristics of open brick/block absorbing walls. Hideki Tachibana, Shin'ichi Sakamoto (Inst. of Industrial Sci., Univ. of Tokyo, Roppongi 7-22-1, Minato-Ku, Tokyo 106, Japan), Hikari Mukai (Miji Univ., Japan), and Tong-Jun Cho (Tokyo Univ., Tokyo, Japan)

Brick and block walls with openings and backing air space with porous materials are often used in auditoria as a sound absorbing treatment. These kinds of walls have a frequency-selective sound absorption property in low frequencies, which is usually explained as the Helmholtz resonator consisting of the air volume in the part of the opening and the backing cavity. In addition, these types of wall often have different peaks of sound absorption coefficient in higher frequencies (around 1000 Hz, in the case of 10-cm thickness of the brick/block). In this study, this sound absorption effect has been investigated by theoretical consideration, reverberation room measurement of a real construction, and a $\frac{1}{10}$ scale model experiment, and it has been clarified that the sound absorption is caused by the effect of open-pipe resonance which happens in the openings. In addition to these studies, the effects of the Helmholtz resonance and open-pipe resonance have been visualized by numerical analysis and physical experiment using the Kundt's dust tube method.

8:50

4aAA2. Sound absorption by seats in concert halls, occupied and unoccupied, and average sound absorption by the interior surfaces of halls—II. Leo L. Beranek (975 Memorial Dr., Ste. 804, Cambridge, MA 02138) and Takayuki Hidaka (Takenaka R&D Labs., Chiba 270-13, Japan)

The average sound absorption coefficients for the interior surfaces of eight halls, in which all finishes were completed before installation of seats, were determined from reverberation measurements. When the chairs were installed, their absorption coefficients, both occupied and unoccupied, were also determined from RT data. Five of these halls have lightly upholstered seats, one has medium upholstered seats, and two have heavily upholstered seats. The data are compared with the 15-hall data of Chap. 5 in Beranek [*Concert and Opera Halls: How They Sound*, Acoustical Society of America (1996)]. A survey of the chair design in those 15 halls has produced a correlation between chair construction and the words: "lightly," "medium," and "heavily" upholstered. The new data permit more accurate estimation of reverberation times as a function of frequency than previously found by comparing the details of a hall-in-planning with the constructions and seats of these eight halls.

9:05

4aAA3. Computation of impulse response for stage-house geometries with use of edge-diffraction models. Rendell Torres, Mendel Kleiner, and Peter Svensson (Chalmers Rm. Acoust. Group, Dept. of Appl. Acoust., Chalmers Univ. of Tech., S-412 96 Gothenburg, Sweden, rendell@ta.chalmers.se)

A mathematical model following diffraction studies by Medwin [J. Acoust. Soc. Am. **72**, 1005–1013 (1982)] is developed to calculate the impulse response of a three-dimensional "stage house" geometry in a baffle. The model includes multiple diffraction components in addition to specular reflection, and it forms a more complete approach than current room impulse-response predictors based on geometrical acoustics. Thus it may also be suitable for more accurate auralization, particularly at low frequencies. A comparison is made with results from a "cone-tracing" computer model [B. Dalenbäck, J. Acoust. Soc. Am. **100**, 899–909 (1996)] and results from physical scale-model measurements. [Work supported by Axel & Margaret Axson Johnson's Foundation, Sweden.]

9:20

4aAA4. A critique on modeling absorbent materials from a numerical simulation perspective. Debashis Basu (State Univ. of New York at Buffalo, 212 Ketter Hall, Amherst, NY 14260)

Absorbent materials, poroelastic in nature, have very complex acoustic wave propagation characteristics in terms of the physics and mathematics defining the phenomenon. Even for a single layer of absorbent, the mathematics governing the physics of the material under acoustic flow conditions is quite daunting. A critical review of the progress in the theories (and inherent assumptions) is being presented. Proper engineering judgment might allow a more simplified numerical simulation to get an idea of the complex acoustical behavior. More often it is the lack of availability of proper data which drives the numerical simulation to be done under certain inexact approximations. The present article tries to lay down the theories and semi-empirical relations developed over the past few decades, and assumptions allowing a simplified model, which at least help in the acoustic characterization of a complex problem. With growing emphasis on a "better, safer, quieter" acoustic environment and mandatory noise control measured in different countries, there would be an ever increasing demand for effective acoustic design and absorbent materials would play a key role in several applications. As would be clearly apparent from the current article, the available "standard commercial" data on absorbent materials gives an estimate of what is happening but does *not* contain enough information to accurately predict the physics of the problem in certain situations.

9:35

4aAA5. Rain and hail simulation and sound isolation tests on a wood and metal roof assembly. Russell Cooper (Jaffe Holden Scarbrough Acoust., Inc., 114A Washington St., Norwalk, CT 06854)

A series of acoustic tests were performed on a proposed roof construction for a new concert hall at The University of North Texas to evaluate its sound transmission loss properties. The test was conducted to determine if the wood and metal deck construction would be sufficient to isolate nearby

highway traffic noise (airborne) and heavy rain and hail noise (structure-borne) from being heard in the concert hall. Four variations of the roof system were constructed and tested to determine the range of possible solutions. Simulation of rain and hail noise test methods and results are presented along with standard ASTM E-90 results and the recommended solution.

9:50

4aAA6. Factors influencing the perception of bass. John S. Bradley (Natl. Res. Council, Ottawa, ON K1A 0R6, Canada), Gilbert A. Soulodre (Commun. Res. Ctr., Ottawa K2H 8S2, Canada), and Scott Norcross (American Univ., Washington, DC 20016-8058)

It is commonly assumed that low-frequency reverberation time determines the perception of bass in concert halls. Laboratory experiments were conducted to evaluate whether this and other factors influence subjects' perceptions of the strength of bass sounds in simulated sound fields. The levels of the early and late arriving low-frequency sounds as well as low-frequency reverberation times were systematically varied. Ten subjects rated the strength of the bass content of the music for each sound field on a five point scale relative to a reference sound field. Both the levels of the early and late arriving low-frequency sound had significant effects on the judgments of the bass content. Low-frequency reverberation time was not significantly related to subjective ratings of bass. The direction of arrival of low-frequency sound had smaller effects on the assessments of the bass content in the sounds. The results suggest that increased low-frequency reverberation time is not important for increasing the sense of bass in concert hall sounds. Further, the low-frequency attenuations caused by the grazing incidence seat dip effect can be compensated for by overhead reflections to increase the sense of bass in concert halls.

10:05–10:20 Break

10:20

4aAA7. Design considerations for Boettcher Hall, a vertical surround symphonic facility. Christopher Jaffe, Gary Madaras (Jaffe Holden Scarbrough Acoust., Inc., 114A Washington St., Norwalk, CT 06854), and John S. Bradley (IRC, Natl. Res. Council, Ottawa K1A 0R6, Canada)

In the early 1960s, Dr. Leo Beranek correlated reflection patterns in concert halls with qualitative listening experiences. Over the years, practitioners and other researchers have confirmed the validity of his work, proving that the direction, amplitude, frequency composition, and arrival time of the reflections define quality, regardless of room geometry. This paper discusses methods for achieving qualitatively preferred listening experiences in vertical surround halls such as Boettcher Concert Hall in Denver, Colorado, and the difficulties that can arise when breaking the mold of a traditional shoe-box hall.

10:35

4aAA8. Acoustical characteristics of a 360-degree surround hall. John S. Bradley (IRC, Natl. Res. Council, Ottawa, ON K1A 0R6, Canada), Gary Madaras, and Chris Jaffe (Jaffe, Holden, Scarborough, Norwalk, CT 06854)

Boettcher Hall in Denver is a large multipurpose hall with audience seating completely surrounding the stage and has been described as a 360-deg surround hall. Extensive acoustical measurements were carried out to evaluate the characteristics of this hall configuration and to compare it to other halls of more conventional design. Impulse response measurements were made for the 45 combinations of three on-stage source positions and 15 receiver positions distributed throughout the audience seating area. The setup of the hall was representative of a typical unoccupied concert configuration. Measured parameters included octave band values of: reverberation times, early decay times, relative sound levels, and early/late sound ratios. Various measures of spatial impression were also measured including: lateral energy fractions, inter-aural cross correlations, and the newly proposed measure of listener envelopment, the late lateral sound

level. Hall average values as well as the spatial variation of acoustical measures are compared to data obtained from a number of other halls. [Work supported by the Concert Hall Research Group.]

10:50

4aAA9. Effects of an audience and variable architectural features on the acoustics of Boettcher Concert Hall. Gary Madaras (Jaffe Holden Scarbrough Acoust., Inc., 114A Washington St., Norwalk, CT 06854), John S. Bradley (IRC, Natl. Res. Council, Ottawa K1A 0R6, Canada), and Christopher Jaffe (Jaffe Holden Scarbrough Acoust., Inc., Norwalk, CT 06854)

Acoustic parameters (including reverberation time, early decay time, relative sound level, early to late sound ratios, lateral energy fraction, and interaural cross correlation) have been measured in Boettcher Concert Hall. All parameters were measured using different settings of variable architectural features such as acoustic drapes, variable height overhead stage reflectors, and movable stage towers. The monaural parameters were also measured with and without an audience. The effects of the variable features on the acoustic parameters are assessed. The effects of the audience on some of the acoustic parameters have been estimated using standard calculation methods. Calculated differences are compared to those actually measured in the hall. [Work supported by the Concert Hall Research Group.]

11:05

4aAA10. A comparative database of salient architectural features: Performance spaces and houses of worship. Christopher Brooks (Orpheus Acoust., 925 Virginia Ave., Lancaster, PA 17603)

A set of salient architectural features for the interior volumes of houses of worship and performance spaces was selected. These features include volume, height, length, width, number of audience seats, type of plan, and predominant material. The buildings were chosen to illustrate a broad range of types within these two categories; well-known examples are included as benchmarks. Data were taken from the literature and entered into a database. With a descriptive statistics program, these data can be inspected from various perspectives. The purpose of this study is to allow a designer to enter features of a proposed design, and by inspection to get a sense of where the design lies in relation to existing buildings. For instance, a proposed church may have important similarities with an existing concert hall. Future plans include access to drawings and acoustical measurements (if available) for each building included, and posting on the World Wide Web with forms for collecting new data.

11:20

4aAA11. Average acoustical characteristics of ten atrium spaces. J. S. Bradley (Acoust. Lab., Natl. Res. Council, Montreal Rd., Ottawa, ON K1A 0R6, Canada) and Y. K. Oh (Mokpo Natl. Univ., Muan, Chonnam, 534-729 Korea)

Extensive acoustical measurements were made in ten different atrium spaces varying in volume from 3 300 to 45 000 m³. Impulse responses were measured at between 18 and 71 combinations of source and receiver position in each atrium depending on the size of the atrium. The computer-based measurement system (RAMSoft-3) used a maximum length sequence signal and a fast Hadamard transform procedure to obtain 5-s-long impulse responses. Octave band values of decay times, sound levels, and energy ratios were calculated for each impulse response. The average values of each quantity in each atrium were used to characterize typical acoustical characteristics of untreated atria. These allow rough estimates of the acoustical conditions in other atria as a function of the volume of the spaces.

Session 4aABa**Animal Bioacoustics: Animal Communication**

Peter Tyack, Chair

*Department of Biology, MS#34, Woods Hole Oceanographic Institution, 45 Water Street, Woods Hole, Massachusetts 02543-1049***Chair's Introduction—8:40*****Invited Papers*****8:45****4aABa1. Vocal tract length and formant frequency dispersion correlate with body size in rhesus macaques.** W. Tecumseh Fitch (Harvard/MIT Speech and Hearing Sci. Prog., 33 Kirkland St., Rm. 1036, Cambridge, MA 02138)

Vocal tract length of 23 rhesus macaques (*Macaca mulatta*) was measured using radiographs and computer graphic techniques. LPC analysis of tape-recorded threat vocalizations was used to determine vocal tract resonance frequencies ("formants") for the same animals. A new acoustic variable is proposed, "formant dispersion," which theoretically depends upon vocal tract length. Formant dispersion is the averaged difference between successive formant frequencies, and was closely tied to both vocal tract length and body size. Despite the common claim that voice fundamental frequency (F_0) provides an acoustic indication of body size, repeated investigations have failed to support such a relationship in many vertebrate species including humans. Formant dispersion, unlike voice pitch, is proposed to be a reliable predictor of body size in macaques, and probably many other species.

9:10**4aABa2. Whistle matching in wild bottlenose dolphins.** Vincent M. Janik (School of Biological and Medical Sci., Bute Bldg., Univ. of St. Andrews, Fife KY16 9TS, UK, vj@st-and.ac.uk)

Bottlenose dolphins (*Tursiops truncatus*) produce individually distinctive signature whistles when they are in isolation. These whistles have been hypothesized to be important in individual recognition and group cohesion. However, captive individuals are known sometimes to imitate each other's signature whistles. Imitation could either lead to confusion over the caller's identity and act against individual recognition or could be used to address specific individuals in a communication network. In this study a hydrophone array with three transducers was used to investigate whether whistle copying also occurs in the wild. Recordings were made in a 500-m-wide channel in the Moray Firth, Scotland. Caller position was determined by comparing differences in the time of arrival of a sound at different hydrophones. Since signature whistles could not be determined by isolating individuals, exact whistle matching in vocal interactions was used to study copying. The results showed that whistle matching occurred in 17% of all whistle interactions. An analysis of the variability of matched whistles showed similar degrees of stereotypy as reported for signature whistles. Thus signature whistle matching seems to be an important feature of bottlenose dolphin communication in the wild. [This study was funded by a DAAD-Doktorandenstipendium HSPI/AUFE.]

9:35**4aABa3. First audiogram for marine mammals in the open ocean and at depth: Hearing and whistling by two white whales down to 30 atmospheres.** Sam Ridgway, Donald Carder, Rob Smith, Tricia Kamolnick, and Wesley Elsberry (NCCOSC RDTE DIV D3503, 49620 Beluga Rd., San Diego, CA 92152-6266)

In examining the potential impact of human-made sound on sea mammals, it was considered that whale hearing sensitivity might diminish with increasing ambient pressure. To test the effect of depth, two white whales made 885 dives to a platform at 5, 100, 200, or 300 m in the Pacific Ocean. At each stationing on the platform up to 12 min at a time, whales whistled when they heard a 500-ms tone from a hydrophone. With increasing depth, air density increase in the middle ear, sinuses, and nasal cavity changed each whale's whistle response, but did not attenuate hearing as it does in the aerial ear (of humans and other land mammals tested in pressure chambers) due to middle ear impedance changes. The findings support theories that sound is conducted through whale head tissues to the ear without the usual ear drum/ossicular chain amplification of the aerial middle ear. These first ever hearing tests in the open ocean demonstrate that whales hear as well at depth as near the surface, therefore, zones of influence for human-made sounds are just as great throughout the depths to which whales dive, or at least to 300 m. [Work supported by Office of Naval Research N0001496WK30349.]

Session 4aABb

Animal Bioacoustics: Bioacoustics Sensing of the Environment

Daniel R. Raichel, Chair

Department of Mechanical Engineering, Cooper Union, 51 Astor Place, New York, New York 10003

Invited Papers

10:15

4aABb1. The use of CF/FM sounds in bats. Yanling Lei (Cooper Union, 51 Astor Pl., New York, NY 10003) and Daniel R. Raichel (Cooper Union and CUNY Graduate School, New York, NY 10003)

The manner in which bats process their echolocation calls involves two main components: (1) producing and transmitting the signals; and (2) receiving and analyzing them [Fenton and Brock, *Bats*, Facts On File (1992)]. The high-frequency signals constituting these calls consist of constant frequency (CF) sounds and frequency modulated (FM) sounds. CF components were found to be used for prey detection and identification; FM components were found to be utilized to update distance information. The Doppler shift also helps in detection of targets and gauging the speed of moving targets. Thus, combining the pulses and the resultant echoes enables the bat to determine object distances, sizes, and shapes. Bats were found to have their own call frequencies and are insensitive to a wide range of potential jamming sounds, which allows them to clearly distinguish their own calls amidst environmental cacophony. Experiments have shown that only when entire CF/FM simulations were used did the bats perform discrimination tests badly [J. D. Altringham, *Bats: Biology and Behavior* (Oxford U.P., New York, 1996)]. This indicates that the echolocation process may be essentially a series processing of CF/FM signals rather than parallel processing of both types.

10:40

4aABb2. A multiple target acoustic scene representation model for bat echolocation signals. Min Liang and Mathew J. Palakal (Dept. of Comput. Sci., Indiana Univ.–Purdue Univ., 723 W. Michigan St., SL 280, Indianapolis, IN 46202)

Determination of the distance to targets from echo delay is an important aspect of echolocation in bats. Neurophysiological studies suggest that the tonotopically organized delay-sensitive neurons (DSN) found in the cortex of bat encode target distance. These studies also indicate that in the tonotopic area, the DSNs in each active ensemble share the same range acuity and the acuity systematically improves over a period of time. A biologically plausible model is suggested here to understand how the bat assembles responses to echoes from multiple targets. This model consists of three stages in cascade: (1) an auditory periphery system (APS); (2) a self-organization neural network (SONN) and; (3) a cortical representation neural network (CRNN). The inputs of APS are the pulse-echo (PE) signals and the outputs are the time-frequency spike trains of PE. SONN self-organizes to the spike trains of PE in frequency index. CRNN has the same architecture of SONN but with a specific neuron response function, $y(T, \delta, f) = I(\delta, f)\Phi[a_1 T(\delta - a_2^n \beta_1)/a_2^n]\Phi(f - \beta_2)$, to achieve the multi-resolution. In this function, $\Phi(\cdot)$ is the negative second derivative of a Gaussian function, $I(\delta, f)$ is the spike amplitude of frequency component f arriving at time δ , T is the time onset of echoes after vocalization, a_1 and $a_2 > 1$ are constants, n is a negative integer, β_1 is the minimum best delay, and β_2 is the best frequency. The response maps of CRNN at different T form the multi-resolution representation of the acoustic scene. Simulations are carried out for the proposed model using phantom targets and the results are compared to neurophysiological findings. [Work supported by NSF.]

Contributed Papers

11:05

4aABb3. Acoustic flow in echo amplitudes and spectra: a viable concept for obstacle avoidance in CF-bats? Rolf Müller and Hans-Ulrich Schnitzler (Dept. of Animal Physiol., Univ. of Tübingen, Morgenstelle 28, D-72076 Tübingen, Germany, rolf.mueller@uni-tuebingen.de)

Bats are capable of obstacle avoidance based solely on echolocation information. When doing so, species being classified as CF (constant frequency) bats allocate most signal energy to rather long, narrow-band portions of their echolocation pulses. These signal components may subserve obstacle avoidance by means of information conveyed by proportional changes in amplitude (due to geometric attenuation, absorption, compound directivity of emitter and receiver) and frequency content (due to Doppler shifts) of the echoes along a trajectory. It can be demonstrated that a combination of these two putative sensory variables allows, in principal, for a metrical reconstruction of target position relative to the animal's flight vector within a hemifield. Thus the contained information seemingly suffices for mediating appropriate avoidance maneuvers. CF bats' unique

narrow-band signal design is matched by a likewise peculiar layout of their auditory system. In particular, the existence of an auditory fovea featuring extremely high filter qualities may be hypothesized to aid estimation of proportional changes in frequency. In order to establish the extent to which the informational content of the signal parameters under consideration may be accessible in the auditory system's primal sketch, the properties of a gammatone filter bank fitted to physiologically determined filter qualities are explored.

11:20

4aABb4. Measurement of the echo location signals of the Atlantic Spotted Dolphin *Stenella frontalis* in the waters off the Grand Bahamas. Whitlow W. L. Au (Hawaii Inst. of Marine Biology, P.O. Box 1106, Kailua, HI 96734) and Denise L. Herzing (Florida Atlantic Univ., Boca Raton, FL 33431)

A line array of three hydrophones with a video camera attached to the array was used to measure the echo location signals of wild Atlantic Spotted Dolphins. The separation distance between hydrophones in the array

was 30 cm. The array was attached to a float that supported an amplifier–line driver assembly with the signals sent via a 76-m multi-conductor cable back to the support boat. The float and array assembly was oriented by a swimmer. The echo location signals from the hydrophone were digitized simultaneously at a sample rate of 500 kHz. Twenty files of echo location click trains were collected with the quality of the data varying from poor (files with lots of whistles and off-axis signals) to very good. The on-axis signals typically had a bimodal spectrum with a low-frequency peak at 45–60 Hz and a high-frequency peak at 120–140 kHz. Peak-to-peak source levels up to 210 dB *re*: 1 μ Pa were measured. The rms bandwidth varied between 32 and 46 kHz, with a cluster around 40 kHz.

11:35

4aABb5. Fish resonance absorption spectroscopy. Orest I. Diachok (Naval Res. Lab., Washington, DC 20375)

Measurements of the absorption coefficient were made at 12 km between 0.6 and 5.0 kHz at a shallow water (83-m) site in the Gulf of Lion in September, 1995. At night absorption lines due to dispersed sardines at 1.3 kHz at 25 m and sardine schools at 1.5 kHz at 60 m were evident. One hour before sunrise the resonance frequency, *f*, of dispersed sardines increased with time, becoming 2.7 kHz at sunrise, as dispersed sardines descended to 65 m; and in the following 15 minutes *f* decreased to 1.7 kHz, as dispersed sardines formed schools, as evidenced by echo sounder records. The signal loss at 2.7 kHz at sunrise, attributed to dispersed sardines at 65 m, was 35 dB; the signal loss at 1.7 kHz during daytime, attributed to sardine schools at 65 m, was 15 dB. The shift in *f* and the decrease in absorptivity that occurred during the transition are in accord with theory: Close proximity between fish in school at 65 m causes both *f*

and the scattering cross section to diminish (Feuillade, 1996). The latter is proportional to the extinction cross section which, together with *n* (number per unit volume), control the absorptivity. Estimation of *n* from absorption spectroscopy measurements, and the hypothesis that marine mammals might exploit this phenomenon, will be discussed.

11:50

4aABb6. Ultrasound-mediated transdermal delivery of tagging compounds into migratory fish. Joseph A. Clark (CDNSWC, Code 734, Bethesda, MD 20084 and COMB, Ste. 236, Columbus Ctr., 701 E. Pratt St., Baltimore, MD 21202), Jane A. Young (CDNSWC, Bethesda, MD 20084), Amrit N. Bart, and Yonathan Zohar (COMB, Baltimore, MD 21202)

There is a need for noninvasive, low-cost, efficient methods of tagging large numbers of aquatic animals. For example, improved tagging methods are needed to follow the migratory patterns of some endangered salmonoid species. This talk reports on an investigation of the feasibility of using cavitation levels of ultrasound to mediate the delivery of tagging compounds into fish. Two tagging compounds are investigated: calcein (Fluoroxon) and oxytetracycline-hydrochloride (Terramycin). Both of these compounds are presently used as tagging agents. However, injection methods of delivery are not efficient for cases where large numbers of wild fish are involved, and if they are placed directly into the aquatic environment of fish the compounds diffuse too slowly through the skin to adequately tag the fish before creating other health problems for them. A system for generating controlled doses of cavitation level ultrasound to schools of fish will be described and measurements of the enhancement in delivery of the tagging compounds to the fish will be reported.

THURSDAY MORNING, 19 JUNE 1997

ROOM O, 8:00 A.M. TO 12:00 NOON

Session 4aBV

Bioresponse to Vibration/Biomedical Ultrasound and Physical Acoustics: Medical Ultrasound

Christy K. Holland, Chair

Department of Radiology, University of Cincinnati, ML 742, 234 Goodman Street, Cincinnati, Ohio 45267-0742

Contributed Papers

8:00

4aBV1. Numerical modeling of biomedically applied transient high-energy sound pulses. Eckard Steiger (Inst. fuer Hochstfrequenztechnik und Elektronik/Akustik, Univ. of Karlsruhe, Kaiserstrasse 12, 76131 Karlsruhe, Germany)

A finite difference in time-domain (FDTD) discretization is applied to simulate finite-amplitude sound pulse propagation in biomedical applications. A numerical model requires a method of accounting for correct representation, especially of shorter wave components. These are generated by nonlinear steepening during the propagation. Therefore, the FDTD discretization has to be of low-dispersive and low-dissipative nature. A high-order scheme suitable for the computation of pulsed sound including weak shocks is applied to model a Storz SL 10 shock-wave lithotripter. The SL 10's electromagnetic source generates cylindrically diverging pulsed pressure waves. By reflection at a hemi-paraboidal brass reflector these waves are focused and steepening of the wave profiles causes weak shock-wave generation in the focal region. A high-order method to model curved boundaries (given by the reflector) is presented. Wave profiles predicted are compared with measured ones. Excellent agreements demonstrate the proposed method to be a flexible tool to simulate biomedically applied transient high-energy sound.

8:15

4aBV2. Acoustic and cavitation fields of a pressure release ellipsoidal reflector. Michael R. Bailey (Appl. Phys. Lab., Univ. of Washington, 1013 N.E. 40th St., Seattle, WA 98105)

In an electrohydraulic lithotripter (device to treat kidney stones), an underwater spark at the near focus of a rigid ellipsoidal reflector produces acoustic cavitation at the far focus. A reflector made of pressure release foam, however, dramatically reduces cavitation with little change in acoustic pressure amplitude ($\sim +35$ MPa, ~ -10 MPa). The pressure waveform at the focus of a rigid reflector is a positive spike followed by a negative tail. The negative tail sets bubbles into a growth-and-collapse cycle hundreds of ms long. With a pressure release reflector, a negative phase precedes a positive spike. Bubble growth induced by the negative phase is quickly ($<10\mu$ s) snuffed by the spike, and collapse is weak. Cavitation was measured by (1) listening for bubble collapse and (2) recording bubble-induced pits on aluminum foil placed *along* the reflector axis. Pit size was used as an experimental measure of cavitation intensity. Intensity was reduced 3–10 times by the pressure release reflector. Results are in agreement with calculations from the Gilmore equation. Comparing the bioeffects of the two reflectors provides a way to assess cavitation's role in lithotripsy. [Work supported by ONR, ARL:UT IR & D, and NIH.]

8:30

4aBV3. Localized detection of cavitation generated by lithotripsy shock waves. Robin O. Cleveland (Appl. Phys. Lab., Univ. of Washington, 1013 N.E. 40th St., Seattle, WA 98105) and Oleg A. Sapozhnikov (Moscow State Univ., Moscow 119899, Russia)

Extracorporeal shock wave lithotripters generate intense cavitation which plays a role in both kidney stone destruction and tissue damage. The cavitation can be detected by listening for the characteristic “double-bang” acoustic emission, radiated by the bubbles during their rapid expansion and collapse phases. Previous workers have used a single focused transducer to detect the double-bang signature. However, the focal spot of the transducers is typically 5 cm long and individual cavitation events can not be localized. A system has been developed that uses two confocal transducers (1 MHz, 20 cm radius of curvature, 10 cm diameter) to detect cavitation events in a localized volume. The system can be run in two modes: passive or active. In the passive mode both transducers listen to acoustic emissions generated by the cavitation field. In the active mode one transducer sends a tone and the second transducer listens to the sound scattered by any bubbles present in the focal region. In both cases the region to which the detectors are sensitive is approximately a sphere 5 mm in diameter. The passive system shows bubble rebound times far longer than predicted by theory for a single bubble. Scattering indicates that bubbles persist in the focal region for at least 6 ms. [Work supported by NIH.]

8:45

4aBV4. Subharmonic imaging with gas-filled microbubbles. William T. Shi, Flemming Forsberg, and Barry B. Goldberg (Dept. of Radiol., Thomas Jefferson Univ., Philadelphia, PA 19107)

The nonlinear properties of gaseous contrast agents can be used to create new ultrasound imaging methods, such as harmonic and subharmonic imaging. Those methods are capable of separating the echoes from the contrast bubbles in blood and those from surrounding tissue. Harmonic images are produced by transmitting sound pulses at one frequency, but receiving echoes at double that frequency, while subharmonic images receive echoes at half that frequency. Experiments revealed that the subharmonic component of echoes from some contrast agents, at relatively high acoustic pressure, was greater than the second-harmonic component. Since harmonic imaging suffers from a reduction in image contrast, due to second-harmonic generation during ultrasound propagation, as well as second-harmonic leakage from the transmitted signal, this finding makes subharmonic imaging an attractive alternative to harmonic imaging. Harmonic response measurements with a pulse-echo arrangement and subharmonic images obtained with a modified scanner will be presented. The acoustic pressure threshold for subharmonic generation was found to be quite different for contrast agents with different compositions. The optimal transmitted ultrasound pulses and the reception strategy for subharmonic imaging will also be discussed.

9:00

4aBV5. A new imaging science test object for performance measurements of ultrasonic imaging systems. Daniel Phillips, Stephen McAleavey, and Kevin J. Parker (Rochester Ctr. for Biomedical Ultrasound, Dept. of Elec. Eng., Univ. of Rochester, Rochester, NY 14627)

A novel test object for ultrasound imaging has been developed. The test object, or phantom, is manufactured with thin film techniques allowing precise placement of “digital scatterers” which can produce sophisticated test targets, similar to those that are widely used in imaging science to evaluate displays, printers, and electronic imaging systems. This type of test object can be used to determine parameters such as the MTF and resolution limits, and can facilitate the detection of imaging phenomena and artifacts such as spatial warping and spatial frequency aliasing. These test objects can be used in conjunction with standard performance evalu-

ation methods and devices to augment and enhance their capabilities and range of application. In addition, by imparting an appropriate motion to the target, it is possible to provide unique test stimuli for pulsed and continuous wave as well as color flow and color energy Doppler ultrasound systems. Results are presented which demonstrate how the ability to make precise line targets allowed the assessment of direction related resolution and spatial frequency response by inspection, thus facilitating the implementation of a simple evaluation procedure.

9:15

4aBV6. Abstract withdrawn.

9:30

4aBV7. Echosignal peculiarities in biological mediums. V. Genis (Biomed. Eng. and Sci. Inst., Drexel Univ., Philadelphia, PA 19104), E. I. Oboznenko, and I. L. Oboznenko (Kiev Polytech. Inst., Kiev 252056, Ukraine)

A modified method of Born’s approximation is proposed. The essence of the method is that in real situations, the incident wave gains the object’s speed, but not the medium’s speed; which follows from the Fresnel formula. A solution algorithm for the weak-reflecting bio-organ’s scattering problem is proposed in this task. The algorithm is based on a more precise definition of Born’s approximation. Numerical and experimental research shows that the pathologies can have dimensions of about 1 mm, which can be identified according to echosignal spectrums. It is shown that norm and pathologies, like pectoral glands, are distinguished. If the pathology is less dense, there will be potent destruction in the echosignal spectrum. If pathology is denser, there will be less potent destruction. The proposed method permits one to classify the pathologies in bio-tissues more subtly.

9:45

4aBV8. Measurements of differential scattering cross section using a ring transducer. T. Douglas Mast (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804), Tomas T. Jansson (Lund Univ., 221 00 Lund, Sweden), and Robert C. Waag (Univ. of Rochester, Rochester, NY 14627)

A method for the measurement of intrinsic scattering object properties is presented. This procedure is based on the measurement of the scattered acoustic field as a function of scattering angle and frequency. Measurements are normalized using analytically determined expressions for emitter and detector beams resulting from a combination of unfocused linear elements arranged in a circular configuration. The spatial effects of finite emitter pulse length and detector gate length are represented by a convolution formula valid for narrow-band signals and long receiver gates. The method includes correction for target absorption as well as measurement of the directly transmitted acoustic power in the free field, yielding the average differential scattering cross section per unit volume. Measured results for two phantoms consisting of glass microspheres embedded in agar show good relative agreement with theory for varying angle, frequency, and

4a THU. AM

phantom properties. For the phantoms employed, scattering effects, rather than increased absorption, are shown to account for most of the difference in transmission loss between pure agar and agar with glass spheres.

10:00–10:15 Break

10:15

4aBV9. Evaluation of a three-dimensional fractal model of ultrasound scattering in liver. Daniel Phillips, Robert Cramblitt, and Kevin J. Parker (Rochester Ctr. for Biomedical Ultrasound, Dept. of Elec. Eng., Univ. of Rochester, Rochester, NY 14627)

It is postulated that a major source of scattering in liver is the collagen rich structure of the vessels that comprise the hepatic vasculature system. This collagen rich hepatic vessel structure manifests itself as the branching pattern of the vessels in three dimensions. Past study of these branching patterns in other organs such as the lung has resulted in data and models which have ascribed a fractal description to their organization. A computer based model has been developed of the portal vasculature system of the human liver based on anatomical, physiological data and precepts as well as a fractal methodology. A thin, planar slice of the model is used to represent sites of ultrasound scattering. A simple model is utilized to calculate the ultrasound scattering from the cross sections of the thin-shelled cylinders used to represent the collagen component of the vascular walls. Simulation of the radio frequency scattered waves is carried out along with envelope detection to allow analysis of the first and second order statistics. The resultant statistics are analyzed in light of the parameters utilized to define the model and their variation.

10:30

4aBV10. Ultrasonic backscattering of blood in simple oscillatory flow. Yu-Hong Lin (Graduate Program in Acoust., Penn State Univ., University Park, PA 16802) and K. Kirk Shung (Penn State Univ., University Park, PA 16802)

Ultrasonic backscattering of porcine whole blood (hematocrit at 40%) was measured by a 10-MHz catheter mounted transducer in a mock flow loop at different flow rates controlled by a roller pump and a pulsatile pump. The amplitude of the oscillatory flow was fixed at 40 cm/s and the baseline velocity was set at three different levels, 0, 20, and 40 cm/s, to change the shear rate range. As the flow rate increases from 0 to 40 cm/s to 20 to 60 cm/s, the peak of cyclic variation of the backscattering signal leads more of the peak of the mean flow velocity. At 40 to 80 cm/s, the magnitude of cyclic variation drops from 4 dB to 1.5 dB. It was found that at the two lower flow rate ranges the backscattering signal increases with shear rate if the mean shear rate is below 150 l/s. When the shear rate further increases, the backscattering signal decreases. The wave form of the cyclic variation is also altered as the baseline shear rate changes. Results show that, in simple oscillatory flow, the amplitude and the timing of the peak of ultrasonic backscattering from whole blood are affected mainly by the shear rate which mediates red cell aggregation.

10:45

4aBV11. Assessment of turbulent flow in arteries using Doppler ultrasound. John S. Stroud, Christy K. Holland (Dept. of Radiol., Univ. of Cincinnati, Cincinnati, OH 45267-0579), Peter J. Disimile, and Curt W. Fox (Univ. of Cincinnati, Cincinnati, OH 45267-0070)

The detection of atherosclerotic-related diseases in the early stages could potentially impact medical treatment in a cost-effective manner. Atherosclerosis is a common arterial disorder characterized by deposition of cholesterol, lipids, and cellular debris in the inner layers of large- and medium-sized arteries. As atherosclerotic deposition progresses, an area of

narrowing, or stenosis, forms, resulting in reduced circulation to organs and other tissues supplied by the artery. When the stenoses become severe in critical arteries (e.g., the carotids or the coronaries) the result can be a catastrophic stroke or heart attack. In the early stages of the disease, the hemodynamics downstream of a stenosis become disturbed. The resulting flow disturbances produce characteristic features in Doppler ultrasound scans which are of diagnostic value. A technique has been investigated for extracting the streamwise turbulence intensity, or normalized velocity variance, using Doppler ultrasound in an arterial flow model. The model consists of a pump and 1.3-cm i.d. polyurethane tubing which is optically and acoustically transparent. Correlation between Doppler ultrasound and laser Doppler anemometry (LDA) measurements was examined in laminar as well as turbulent flow. Continuous flow has been employed initially and the work will be extended to pulsatile flow to replicate cardiac output.

11:00

4aBV12. Ultrasonic wavefront distortion caused by human abdominal wall layers. Laura M. Hinkelman (Dept. of Meteorol., Penn State Univ., University Park, PA 16802), T. Douglas Mast (Penn State Univ., University Park, PA 16802), Michael J. Orr, and Robert C. Waag (Univ. of Rochester, Rochester, NY 14627)

The relative importance of the fat and muscle layers of the human abdominal wall in producing ultrasonic wavefront distortion has been assessed based on direct measurements. Specimens employed included 6 whole abdominal wall specimens and 12 partial specimens obtained by dividing each whole specimen into a fat and a muscle layer. In the measurements, a hemispheric transducer transmitted a 3.75-MHz ultrasonic pulse through a tissue section maintained at 37°. The received wavefront was measured by a linear array translated in the elevation direction to form a two-dimensional aperture with overall dimensions 92.16×46.08 mm² and a measurement spot size of 0.72×1.44 mm². Differences in arrival time and energy level between the measured waveforms and computed references that account for geometric delay and spreading were calculated. After correction for the effects of geometry, the received waveforms were synthetically focused at 180 mm. The characteristics of the distortion produced by each specimen and the quality of the resulting foci were analyzed and compared. The results indicate that both fat and muscle layers contribute significantly to the distortion of ultrasonic beams by the abdominal wall. However, the spatial characteristics of the distortion produced by fat and muscle layers differ substantially, and the total distortion produced by the abdominal wall is not equivalent to the sum of the distortion produced by the layers.

11:15

4aBV13. Infrasound method for bone mass measurements. Dimitri Donskoy (Davidson Lab., Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030)

Studies of application of acoustic energy for noninvasive skeletal diagnostics have shown feasibility and demonstrated the advantages of utilizing acoustic techniques for bone mass measurements. Unlike conventional radiological techniques, acoustic techniques emit no radiation, are low cost, and utilize equipment which is portable and easy to operate. Although a significant number of acoustic tests have been performed, these techniques have not been used as bone diagnostic tools in clinical applications because of the difficulties in the interpretation of measurements. Thus, ultrasound velocity and attenuation depend on density as well as on certain other properties of bone. The effect of soft tissues creates additional difficulties in the interpretation and use of these techniques. The proposed infrasound method involves measurements of the "rigid body" resonance of a tibia or ulna with the use of specially designed arrangement, which greatly simplifies a biomechanical model of tibia (ulna). This makes the

measurement and its interpretation much more accurate. Numerical analysis of the developed model showed that the mass of bone and its loss can be accurately measured and these measurements do not depend on variation in bone flexibility and soft tissue parameters. An experimental setup was arranged and *in vivo* measurements were conducted and proved the basic concept of the proposed method. [Work supported by NASA.]

11:30

4aBV14. Methods of integral equations in tasks of bio-organs sound scattering. V. Genis (Biomed. Eng. and Sci. Inst., Drexel Univ., PA 19104), E. I. Oboznenko, and I. L. Oboznenko (Kiev Polytech. Inst., Kiev, 252056, Ukraine)

Ultrasound scattering on biological, arbitrary-form structures is considered. Methods of integral equations by volume and surface integration are used in the task. The methods allow one to explore scattering from the surface and volume of arbitrary pathological insertions. A method of small perturbations is used. In the first approximation the weak-reflecting pathology is accepted. It is noted that pathology form and value appreciably affect echosignal and spectrum structure. In particular, echosignal spectrums for less dense and denser pathologies vitally are notable. The proposed method allows one to classify the pathologies of bio-organs. Sufficient agreement with theoretical models is shown experimentally on the weak-reflecting object (rubber).

11:45

4aBV15. Numerical calculations of scattering characteristics of inhomogeneities in biological tissue. Vladimir I. Genis (School of Biomed. Eng. Sci. and Health Systems, Drexel Univ., Philadelphia, PA 19104, genis@cbis.ece.drexel.edu), Igor L. Oboznenko (Polytechnic Univ., Kiev 252037, Ukraine), and Stanley Zietz (Drexel Univ., Philadelphia, PA)

Identification of inhomogeneities is one of the main problems in medical diagnosis, particularly in tissue characterization. Often the backscatter signal does not permit discrimination between inhomogeneities in the tissue. Despite high effectiveness of the *B*-mode diagnostic systems, at the present time ultrasonography does not provide a sufficient degree of diagnostic confidence in identification of pathology. Diffraction tomography promises to extend these observations quantitatively and provide information about tissue properties. The main goal of this work is to present a theoretical basis of the scattering process on inhomogeneities and provide some preliminary experimental results confirming theoretical investigations and numerical simulations. The method of integral equations for solving forward and inverse scattering problems is used, and scattering diagrams are presented. Experimental setup is based on one transmitting and seven receiving elements. Presented experimental results support numerical simulations.

THURSDAY MORNING, 19 JUNE 1997

ROOM S, 8:30 TO 11:45 A.M.

Session 4aEA

Engineering Acoustics: Modeling and Predictive Capabilities for Transducers I

David Yeager, Chair

Motorola, Acoustics Technology Laboratory, Room 2319, 8000 West Sunrise Road, Fort Lauderdale, Florida 33322

Chair's Introduction—8:30

Invited Papers

8:35

4aEA1. Design simulation of composite transducers. Wenwu Cao (Whitaker Ctr. for Med. Ultrasonic Transducer Eng., Dept. of Mathematics, and Mater. Res. Lab., Penn State Univ., University Park, PA 16802)

Many advanced transducers, such as those for medical ultrasound imaging, are made of composite materials. These composites contain both piezoelectric active and inactive constituents and therefore have a nonuniform surface vibration profile. This nonuniform surface vibration will in turn influence many characteristics of the acoustic beam pattern produced by the transducer, such as the focusing ability, sidelobes, and the near-field pressure distribution. These complex transducer systems cannot be modeled properly using equivalent circuit models due to the 3-D nature of the transducer vibration. A 3-D finite-element modeling scheme will be given in this presentation. The modeling contains two stages: First, a finite-element method is used to study the nonuniform structural vibrations of the composite transducer and obtain the surface velocity and displacement distributions; second, this surface velocity and displacement information will be fed to the Helmholtz integral to derive the transducer radiation pattern. It is also possible to compute directly the near-field pressure distribution using FEM. Accurate predictions on the transducer behavior are obtained. In addition, stress and electric field concentrations inside the transducer can be calculated. Comparison with experimental results will be shown via an animation.

8:55

4aEA2. Optimal digital compensation networks for acoustic transducers. Paul J. Titterton, Jr. (SRI Int., 333 Ravenswood Ave., Menlo Park, CA 94025)

A digital compensation network is designed which (1) minimizes deviation from the user-determined optimal transducer response, (2) limits the sensitivity to physical perturbations of that response, and (3) ensures actuator authority limitations for known and unknown-but-bounded input sequences. H^∞ , H^2 , l^∞ , and l^1 transfer function norms translate the design goals and specifications into a convex constrained optimization problem, which is solved using semidefinite programming. The methodology can be simply extended to treat transducer arrays and optimal beamforming. A mathematically modeled underwater acoustic transducer provides a numerical example for exploring the optimal trade-off between competing specifications. [This work is partially supported by SRI International.]

9:15

4aEA3. Capabilities and limitations of “state of the art” modeling techniques for use in the design of electroacoustic transducers. Carlos I. Beltran (Chicago Lawn High Technol., 3321 W. 62 St., Chicago, IL 60629)

Several modeling systems exist to aid in the design and engineering of electroacoustic transducers. Finite element analysis (FEA) can be used to model magnetic circuits, electromagnetic circuits, thermal transfer capacity, mechanical and dynamic properties of structural parts, vibrational modes of diaphragms and other moving parts, piezoelectrics, acoustic impedance, and acoustic radiation. Several computer programs are available to model electroacoustic systems and components via lumped parameter methods derived from electrical-mechanical and electrical-acoustical analogies. Lastly, general purpose mathematics programs and spreadsheets can be used for a wide variety of problems. The capabilities and limitations of these modeling techniques are discussed using real world examples.

9:35

4aEA4. Finite-element and boundary-element computer modeling for loudspeaker cone design. Roger E. Shively (Harman-Motive, Inc., 1201 S. Ohio St., Martinsville, IN 46151)

The use of finite-element and boundary-element computer modeling for the design and optimization of loudspeaker cones is demonstrated. First, the specifics of the analytical model and theory are summarized. The experimental setup, which utilizes laser vibrometry is described, and the correlation of analytical FEM and BEM models to experimental models and anechoic measurements for a test case are reviewed. Three case histories are then illustrated: (1) correlation of a high-frequency model, (2) a design case for maintaining a current loudspeaker packaging size with improved frequency response and distortion over the current performance, and (3) a design case for evaluating a new material for a loudspeaker's diaphragm.

9:55

4aEA5. Measurement and analysis of loudspeaker component nonlinearities at low frequencies. Thomas P. Heed (Oxford Int., Ltd., 4237 W. 42nd Pl., Chicago, IL 60632)

In the constant displacement region, several displacement-dependent quantities control the behavior of electro-dynamic transducers. This research uses a tensile/compression loadframe to measure the displacement-dependent stiffness, B1-Product, and mechanical losses of various driver components. The individual data sets are curve-fitted by power series expansions. The power series coefficients are used to predict the relative low-frequency total harmonic distortion (THD). Then, the measured components are built into loudspeakers, and the predictions are compared to actual THD measurements. Experimentally, this method is very sensitive. Additionally, the effective stiffness of a transducer's suspension is merely the sum of the power series coefficients representing the spider's and surround's stiffnesses. By comparing various spiders and cones, the effective stiffness of a suspension can be linearized, thereby reducing low-frequency THD. Finite-element analysis (FEA) simulation of nonlinear suspension behavior can facilitate such a linearization process. Lastly, power series coefficients can determine the relative variability of components (e.g., which of two spider designs is more variable). OEM manufacturers of transducers must maintain statistical process control of Q_{Tot} , resonance, and THD. Q_{Tot} , resonance, and THD are all dependent on displacement-dependent quantities. Therefore, reducing component variability is helpful in achieving an acceptable c_{pk} for Q_{Tot} , resonance, and THD.

10:15–10:30 Break

10:30

4aEA6. A simplified approach to a nonlinear loudspeaker modeling. Earl Geddes (GedLee and Assoc., 1388 Medinah Dr., Itasca, IL 60143)

Exact solutions of the nonlinear equations for a loudspeaker are extremely difficult to solve as they involve Volterra series expansions and higher-order system functions and impulse responses [A. R. Kaiser *et al.*, J. Audio Eng. Soc. **35**, 421–433]. Indeed, these complex methods are required for exact system simulations, however, they can be greatly simplified with a reasonable assumption about the system. By assuming that the distortion (or nonlinearity) is not so large that it causes an excessive amount of distortion, say about 20% THD, a vastly simplified approach can be developed. It is not always the case that this near-linear assumption can be made, but clearly, it should be accurate for higher performance units. This presentation will show how to use an algebra processor (Maple) to develop these simplified predictions given the nonlinearities of the parameters. These equations will then be used to show the effect that various loudspeaker enclosure designs have on the distortion of a given (typical) loudspeaker using a commercially available software package written by the author.

10:50

4aEA7. Loudspeaker drive-unit measurements at excursion. David Clark (DLC Design, 47677 Avante Dr., Wixom, MI 48393)

Loudspeaker drive units are conventionally modeled as linear devices. Evidence is presented that shows that this assumption leads to unacceptable modeling errors for practical conditions of use. This paper presents new techniques for accurate, nondestructive measurement of linear and nonlinear electromechanical parameters. These parameters are used in more complete models which improve accuracy of frequency response and distortion estimates. The measurement workstation uses pneumatic pressure to displace the diaphragm in order for position-dependent parameters to be measured over the full excursion range. Suspension compliance and mechanical damping are measured under conditions which simulate normal operation, thereby capturing the influence of creep, aerodynamic drag, and other second-order characteristics which are not measured separately at present. Moving mass and force factor

measurements utilize a magnetic position servo and gravity to obtain data without displacing the suspension. Conventionally, maximum diaphragm displacement, x_{\max} is calculated from voice-coil and magnet geometry. This ignores the magnetic fringe field and the effect of suspension nonlinearities. An improved value of x_{\max} is derived from the new measurement data.

11:10

4aEA8. Acoustical simulation of active noise control telephone earpiece system. John C. Baumhauer, Jr. (Lucent Technologies, Bell Labs, 6602 E. 75th St., Indianapolis, IN 46250)

A lumped-parameter equivalent circuit representation of an active noise control (ANC) feedback system has been developed for telephone handset earpiece application. Since the feedback controller must be stable to avoid "squealing" in the user's ear, its filter must work with known and robust transducer transfer functions. The earpiece loudspeaker (receiver) is the weak link. As opposed to the feedback electret microphone located between the earpiece and pinna, the moving-coil receiver's response undergoes a large, negative phase change with increasing frequency owing to multiple, highly coupled degrees-of-freedom. Moreover, the magnitude and phase of the receiver's transfer function are shown to depend on its electroacoustic parameter tolerances, such as cone stiffness, coil $B1$, acoustic damping, and handset porting. Another variation is due to the acoustic leak that occurs naturally between the earpiece and the ear. This is not just a function of the earpiece design, but also of the varying handset-to-ear force applied. Not only is low-frequency response in the ear lost with an increased leak, but the high gain feedback magnitude is reduced and the overall phase lag in the receiver becomes greater. Results with simulated ITU-T coupler equivalent circuits demonstrate these effects. A prototype ANC system is discussed.

Contributed Paper

11:30

4aEA9. A BEM approach to effectively modeling differential microphones. David Yeager (Acoust. Technol. Lab., Radio Products Group, Motorola Corp., Rm. 2319, 8000 W. Sunrise Blvd., Ft. Lauderdale, FL 33322)

A flat frequency response, reasonably good signal-to-noise ratio, mechanical robustness, and low cost make omnidirectional electret microphones ideal for telecommunications applications. Differential microphones offer the potential additional advantage of improving the acoustic signal-to-noise ratio by rejecting unwanted background noise. Background noise rejection results from two independent acoustic mechanisms—directional selectivity and proximity. Modeling the acoustic performance of a differential microphone requires accurately representing these mecha-

nisms, which means properly accounting for diffraction and damping effects. For these reasons, differential microphones are more sensitive to mounting conditions and housing geometry, and therefore, designing a differential microphone into a product can be a daunting task. This paper discusses some of the design principles that dictate noise rejection, and suggests methods for modeling differential microphones using numerical techniques such as the boundary element method (BEM). A coupled BEM model enables an acoustical interface between the microphone diaphragm and nearby cavities and ports. This is especially important near resonance. Damping plays a key role in microphone performance, and is an overriding factor in the directional behavior of a differential microphone. Techniques for representing these effects in a BEM model using SYSNOISE will be discussed. [SYSNOISE is a registered trademark of LMS Numerical Technologies, Belgium.]

THURSDAY MORNING, 19 JUNE 1997

ROOM D, 9:00 TO 11:15 A.M.

Session 4aMU

Musical Acoustics: Sound Generation in Musical Instruments

George Bissinger, Chair

Department of Physics, East Carolina University, Greenville, North Carolina 27858

Contributed Papers

9:00

4aMU1. Effects of reed cell geometry on the vibration frequency and spectrum of a free reed. James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402)

In a reed organ the free reed is mounted above the windchest and is typically surrounded by a reed cell cavity with small cross-sectional area. The length of this tubelike cavity varies from just slightly longer than the reed in some instruments to several centimeters longer in others. A series of measurements of the frequency and spectrum of the radiated sound have been made on reeds from American reed organs mounted on a laboratory windchest, using a simulated reed cell of varying dimensions. The presence of the reed cell cavity generally results in a lower frequency of vibration at the same blowing pressure and increased amplitudes of the higher harmonics relative to the fundamental, although some anomalous

effects are observed for longer reed cells. Some hysteresis effects are observed as the blowing pressure is continuously increased and decreased.

9:15

4aMU2. Acoustics of the khaen: The Laotian free-reed mouth organ. Casey A. Fetzer and James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402)

The khaen is constructed with free reeds mounted in bamboo pipe walls inside a carved wooden windchest. Each reed of the khaen sounds for both directions of air flow (inhaling and exhaling). The reed vibration is strongly coupled to the pipe resonance, and the reed will sound only if a small hole near the reed is closed, causing the resonant frequency of the pipe to be near that of the reed. For some examples of khaen made in northeastern Thailand, variations in frequency and sound spectrum with

blowing pressure have been studied, with both positive and negative blowing pressures considered. The relationship between frequency of reed vibration and pipe length has been studied to determine the range of pipe length over which the reed can be made to sound, as well as the amount of frequency shift associated with changes in resonance frequency of the pipe. A comparison is made with a previously reported result that the sounding frequency of the reed-tube combination is higher than the natural resonance frequencies of either the reed or the pipe taken alone. [L. E. R. Picken, C. J. Adkins, and T. F. Page, *Musica Asiatica* 4, 117–154 (1984).]

9:30

4aMU3. Applications of the 2 DOF network representation of violin cavity modes A0 and A1 to partial cavity volume and f-hole geometry changes in an Aluminum violin surrogate. George Bissinger (Dept. of Phys., E. Carolina Univ., Greenville, NC 27858, phbissin@ecuvm.cis.ecu.edu)

Lack of uniqueness in the circuit elements of the 2 DOF Network model [E. A. G. Shaw, *J. Acoust. Soc. Am.* 87, 398–410 (1990)] for the two lowest violin cavity modes A0 and A1 is one difficulty in its application, e.g., Shaw provided three possible parameter sets with substantial variation among them. To provide constraints on the range of parameters, global fitting of a large database of A0 and A1 frequencies, and their upper bout–lower bout pressure ratios—obtained by adding measured quantities of water (up to 250 cm³) to a flat-plated A1 violin-shaped cavity (volume = 1800 cm³) and varying its orientation—produced an optimized set of parameters close to Shaw’s preferred set. With this optimized set of parameters, it was possible to estimate: (1) the mutual inductance contribution to the f-hole inductance; (2) a volume dependence of the A0 mode frequency of $f \propto V^{-0.254}$ in good agreement with the experimental value of $V^{-0.269}$; and (3) predictions of the effect of sliding, rotating, and flipping the f holes.

9:45

4aMU4. Measurement of direct radiation from violin excited by force hammer impact at bridge. Machele Bailey and George Bissinger (Dept. of Phys., E. Carolina Univ., Greenville, NC 27858)

A violin suspended inside a small anechoic chamber was impacted at the bridge with a small force hammer. A rotatable, semicircular, equispaced seven-microphone array was used to measure the radiation in a 0–2 kHz range over a sphere of radius 30 cm (in the near field ($r < 1$) region for $f < 1100$ Hz) at 15° intervals (except in the neck-fingerboard region). The input-force-normalized microphone spectra were then analyzed to separate the contributions from each violin normal mode. On animation, most of the higher-frequency radiation patterns clearly showed apparent “phasiness” among portions of the radiation patterns, an effect primarily related to near-field effects and not the bridge impulse propagation delay through the violin corpus. The measured 3-D radiation patterns for each mode compared well with those calculated with a boundary element program using as input normal mode vibrational data from a prior modal analysis for this violin.

10:00

4aMU5. V-R model predictions of averaged radiation from a violin compared with spatial average of bridge force hammer-excited direct radiation. George A. Bissinger and Machele Bailey (Dept. of Phys., E. Carolina Univ., Greenville, NC 27858, phbissin@ecuvm.cis.ecu.edu)

Mechanical modal analysis data for a violin furnish two essential components for the vibration-radiation (V-R) model [G. Bissinger, *J. Acoust. Soc. Am.* 97, 3154–3164 (1995)] to reliably predict the averaged radiative properties of a bowed string instrument from its measured mechanical properties, viz., (1) how strongly a particular corpus mode is excited at the bridge and (2) normal mode frequency and shape data for boundary element acoustic radiation calculations of the radiation efficiency for each corpus normal mode. Combining both of these elements in the V-R model gives predictions of a spatially averaged acoustic radiation for a violin.

These were compared with similarly spatially averaged measurements of direct radiation from a violin excited by force-hammer impact at the bridge using a rotating semicircular ($r=30$ cm) seven-microphone array in a small anechoic chamber covering almost an entire sphere around the violin. The V-R model calculations for each mode compared quite well in general to the measured average radiation.

10:15

4aMU6. Important violin resonance frequencies. Oliver E. Rodgers (179 Kendal Dr., Kennett Square, PA 19348, olerodj@juno.com)

Experimental determination of the natural frequencies and nodal line patterns of those modes which produce sound was made of a test violin which had previously been analyzed by George Bissinger using an experimental modal analysis system [*J. Acoust. Soc. Am.* 97, 3154–3164 (1995)]. Excitation of the test violin was done by bowing the instrument and the analysis system was the CONQT software previously used by the author. Nodal lines were defined approximately with a small microphone which was held very closely to the plate surface to seek lines of minimum response. Comparison with the Bissinger results points out the very few vibration modes in the first 1000 Hz which contribute to the overall tone of the instrument. These modes are found among those displayed in Bissinger’s work. Only two modes of the air cavity and four modes of the gross mechanical system are important in this frequency range. The CONQT data suggest that tone quality is determined strongly by the five to seven prominent higher modes, which are complex motions of small areas of the plates, and/or by the ability of the lower modes to transmit sound at off-peak frequencies.

10:30

4aMU7. Vibrational modes of two violins. Mark Roberts and Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115)

Using electronic TV holography, as well as other methods for observing vibrational motion and sound radiation, the normal modes of vibration in two violins have been studied. The principal modes observed in a Hutchins violin showed fairly good agreement with those reported by Marshall [*J. Acoust. Soc. Am.* 77, 695–709 (1985)] for the same violin. The strongly radiating T_1 and C_3 modes appear to be doublets, and this phenomenon is discussed. Vibrational modes excited by a force applied to the bridge by internal sound pressure, and by the sound field of a loudspeaker are compared.

10:45

4aMU8. Subharmonic generation in cymbals at large amplitude. Charles Wilbur and Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115)

The aftersound in cymbals has been attributed to chaotic behavior at large amplitude, which provides a mechanism for the conversion of low-frequency energy to a rich assortment of modes of much higher frequency [N. H. Fletcher, “Nonlinear dynamics and chaos in musical instruments,” in *Complex Systems: from Biology to Computation*, edited by D. G. Green and T. Bossomaier (IOS, Amsterdam, 1993)]. When a cymbal is driven sinusoidally at an amplitude slightly lower than that leading to chaotic behavior, the cymbal sound includes harmonics of a fundamental having a fraction (often one-half or one-third) the drive frequency. Nonlinear behavior leading to subharmonic generation as well as chaotic behavior is discussed.

11:00

4aMU9. Investigation into the feasibility of a smart acoustic guitar. Steven F. Griffin (Air Force Phillips Lab., 3550 Aberdeen Ave. SE, Kirtland AFB, NM 87117-5776) and Sathya V. Hanagud (Georgia Inst. of Technol., Atlanta, GA 30332)

In 1990, the “Mendelssohn” Stradivarius violin sold for \$1,686,700. A good violin sells for around \$2000. What is it about the Stradivarius that makes it cost almost 1000 times as much? The structure and geometry of

the two instruments are similar, yet subtle differences in structural dynamics cause them to vibrate differently in response to a violinist's bow. This, in turn, causes differences in the sound produced by the two instruments, which ultimately determines quality and, to a large extent, price. If it were possible to force the less expensive violin to vibrate like the Stradivarius, the legendary sound would follow. This paper explores the potential for the use of active structural/acoustic control to obtain a desired acoustic response in an acoustic guitar. Detailed information on the desired acoustic

response of guitars is available, and experimental specimens are relatively easy to obtain. The model developed is an elastic plate with piezoceramic sensors and actuators backed by a rigid, vented cavity. The sensors and actuators are used with active feedback control to influence the vibration and acoustics of the guitar. The feasibility of favorably changing acoustic guitar vibration and acoustics is examined in detail, including analytical and experimental results.

THURSDAY MORNING, 19 JUNE 1997

ROOM G, 10:30 TO 11:30 A.M.

Session 4aNS

Noise: Progress Report and Discussion on the Continuing Activity on ASA's Role in Noise and Its Control

Bennett M. Brooks, Chair

Brooks Acoustics Corporation, P.O. Box 3322, Vernon, Connecticut 06066

The Technical Committee on Noise, through a subcommittee, is involved in outreach activities related to noise control. The details of a special session on meeting-room acoustics to be presented at the meeting of the Council of Engineering and Scientific Society Executives (CESSE) in Pittsburgh in July 1997 will be reviewed. Plans for a noise seminar to be held in San Diego will also be reviewed. The results of discussions with hotel industry engineers regarding meeting-room acoustics will be presented. Progress will be reported on a video of noise control demonstrations for use in classrooms. The committee is also involved in exploration of the possibilities of hands-on automated hearing testing in public spaces.

THURSDAY MORNING, 19 JUNE 1997

ROOM N, 9:00 TO 10:30 A.M.

Session 4aPA

Physical Acoustics: Porous Media

Craig J. Hickey, Chair

National Center for Physical Acoustics, University of Mississippi, Coliseum Drive, University, Mississippi 38677

Contributed Papers

9:00

4aPA1. Modification of Biot's theory of porous materials. H. Tavossi and B. R. Tittmann (Penn State Univ., Dept. of Eng. Sci. and Mech., University Park, PA 16802)

Acoustic waves in porous material are traditionally investigated by using Biot's theory to describe the elastic wave propagation in such materials. However, experimental results obtained by models of porous media made of cohesionless particles forming loose solid matrix with fluid-filled pores, contradict the above theory at high frequencies. These results show that the acoustic wave speed as a function of frequency and particle size goes through a maximum and then decreases with frequency until a cutoff frequency is reached, whereas Biot's theory predicts an asymptotic increase of propagation speed with frequency, without a wave cutoff frequency. The theoretical analysis and the experimental investigation are presented that modify and adapt Biot's theory to the cohesionless porous materials. The equations describing these modifications to the theory take into account the following effects: the contact surfaces between the solid particles, their number concentration, the size of the solid particles, the

depth at which they are situated, the external pressure, their random arrangement, and the propagation direction of the acoustic waves.

9:15

4aPA2. Coupled phase theory for the complex density of rigid-porous materials. Keith Attenborough and Jon Evans (Faculty of Technol., The Open Univ., Milton Keynes MK7 6AA, England, k.attenborough@open.ac.uk)

Despite the fact that most porous materials, of influence in building and urban acoustics, are either fibrous or granular in nature, theoretical models used to predict their acoustical behavior at audio frequencies are derived from an assumed *pore-based* microstructure. This leads to the need for pore-related parameters that are either difficult to measure or not measurable independently. An alternative is to start from the known size and shape distribution and properties of the constituent solid particles. One possible particle-based approach is coupled phase theory. This has been used extensively to predict sound propagation in suspensions and emul-

sions. By means of an expression for the interphase inertial force appropriate to spherical particles, it has been shown that “frameless” Biot theory is equivalent to coupled phase theory modified to include hydrodynamic interactions between the forces acting on individual particles. The two models differ only in their expression of the interphase viscous force. Extensions of the coupled phase approach that are necessary when considering air-filled granular and fibrous materials are discussed.

9:30

4aPA3. Dynamics of single bubbles in laboratory sediments. D. Felipe Gaitan and James M. Sabatier (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677, gaitan@olemiss.edu)

Increased interest in the acoustical properties of sediments has been motivated by the fact that in shallow-water environments the seabed is the dominant factor controlling propagation. With this in mind, the dynamics of a gas bubble surrounded by a poro-elastic medium in a laboratory sediment have been studied. Measurements of the effect of the surrounding medium on the resonance frequency and the quality factor of the bubble will be presented. [Work supported by ONR.]

9:45

4aPA4. On the dynamic drag coefficient for wave propagation through fluid-saturated highly porous open-cell foams. Robert E. Slade, Keith Walton (School of Mathematical Sci., Univ. of Bath, Bath BA2 7AY, UK), Colin A. Mead, Alan T. Parsons (Winfrith Technol. Ctr., Dorchester, UK), Roger J. Pinnington, and Anthony R. Briscoe (Univ. of Southampton, Southampton, UK)

This paper considers the damping of waves propagating in high porosity (97%) open-cell foams which are fully saturated by a viscous fluid. The technique of homogenization is applied on two spatial scales to the fluid-foam system. The resulting equations are found to be those of Biot [M. A. Biot, *J. Acoust. Soc. Am.* **28**, 168–178 (1956)] but reduce to the simpler form given by Chase [D. M. Chase, *J. Acoust. Soc. Am.* **65**, 1–8 (1979)] when particular properties of the solid foam matrix are taken into consideration. A number of important material parameters are identified, and of particular significance is the measure of drag between the solid and fluid components. The results of numerical fluid flow experiments are used to obtain estimates of the damping coefficient. Comparisons are made with measurements of the steady state and frequency-dependent damping coefficients, which are determined using specially constructed experimental rigs. [Published with the permission of the Controller of Her Britannic Majesty’s Stationary Office.]

10:00

4aPA5. The behavior of transient elastodynamic waves at a linear slip interface. Martin D. Verweij (Lab. of Electromagn. Res., Dept. of Elec. Eng., Ctr. for Technol. Geosci., Delft Univ. of Technol., Mekelweg 4, 2628 CD Delft, The Netherlands, m.d.verweij@ctg.tudelft.nl) and Christopher H. Chapman (Schlumberger Cambridge Res., Cambridge CB3 0EL, England)

A method is presented for the analysis of the transmitted and reflected transient elastodynamic wavefield at a fracture that may be modeled as a linear slip interface. This interface model implies that the traction is continuous at the fracture, while the displacement may show a jump discontinuity that is proportional to this local traction. To start with, for both the *SH* and the *P/SV* wave systems, transform domain expressions are determined for the transmitted and reflected wavefields at the interface. Subsequently, an almost completely analytical transformation back to the space-time domain is carried out. The method provides the exact time domain transmitted and reflected waves (including the body wave, interface wave, and head wave contributions) that are caused by a spatially curved, incident wavefield due to a point source. Various numerical results are given. For the *SH* case, the directionally dependent filter behavior of the fracture is shown. For the *P/SV* case, first the reflection, transmission, and conversion of body waves is presented. The next set of results involves the interface wave contribution and includes pictures of the particle movement on both sides of the fracture. Finally, the presence of headwaves in case of *SV*→*P* and *SV*→*SV* reflection is shown.

10:15

4aPA6. Estimating pile bearing capacity via acoustic and seismic measurements. Chung-Ham Yang (Dept. of Civil Eng., Penn State Univ., University Park, PA 16802, cxy122@psu.edu) and H. Randolph Thomas (Pennsylvania Transportation Inst., University Park, PA 16802)

Current dynamic methods estimate pile bearing capacity through the measurements of in-pile stress waves. Typically, accelerometers and strain gauges are mounted to the pile shaft. This on-pile instrumentation is expensive and interrupts the driving operation. The high cost and interruptive nature of the on-pile instrumentation imposes practical limits on the number of piles that can be tested. The limited amount of tested piles cannot reliably represent the remaining piles that might be driven through a variety of subsurface conditions. This research is aimed at developing a method for estimating pile bearing capacity through the acoustic and seismic measurements. The pile–soil system is modeled as a single-degree-of-freedom mass-spring-slider-dashpot system. Based on principles of vibration, the pile bearing capacity is analytically derived as a function of the vibrating mass, damped natural frequency, damping ratio, and soil quake. The damped natural frequency and the damping ratio are determined from ground waves and air pressure waves, and the vibrating mass and soil quake are estimated based upon *in-situ* soil properties. The measurement of ground waves needs no on-pile instrumentation and does not interrupt the driving operation. This enables every pile at a site to be tested.

Session 4aPP

Psychological and Physiological Acoustics: Masking and Temporal Effects (Poster Session)

Lawrence L. Feth, Chair

Department of Speech and Hearing Sciences, Ohio State University, 110 Pressey, 1070 Carmack Road, Columbus, Ohio 43210

Contributed Papers

All posters will be on display from 8:30 a.m. to 11:30 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:30 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m.

4aPP1. Growth of masking by harmonic complexes with high crest factors. Marjorie R. Leek and Van Summers (Army Audiol. & Speech Ctr., Walter Reed Army Med. Ctr., Washington, DC 20307-5001)

There is a strong link between the discrimination of small phase changes in highly peaked harmonic complexes and the sensation level of the stimuli. This relationship may stem from a differential ability across listeners to code dynamic amplitude changes within waveform periods. To test this hypothesis, growth of masking was measured for flat-amplitude harmonic complexes varying in phase structure and level. Maskers were constructed with odd harmonics in cosine phase and all even harmonics shifted in phase from 0 to 90 degrees. Masked thresholds were measured for a 5-ms Hanning-windowed probe centered either at the primary peak of the masker waveform or at a secondary peak near the midpoint of the period. Growth-of-masking functions when the probe is placed at the primary waveform peaks are generally linear. However, in normal-hearing listeners, masking of probes placed at the secondary peaks grows more nonlinearly. Reduced masking observed at low stimulus levels may be related to compression of the high-amplitude primary peaks which act as forward maskers. Differential forward-masking by the large primary peaks across listeners with normal hearing and those with hearing impairment will be related to the ability to discriminate complexes with differing phase structures. [Work supported by NIH.]

4aPP2. Psychometric functions for detecting an increment added to a continuous or gated pedestal. Huanping Dai and Donna L. Neff (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

Green *et al.* [J. Acoust. Soc. Am. **66**, 1051–1056 (1979)] reported that a psychometric function for detecting a 1000-Hz increment was two to three times steeper with a continuous rather than with a gated pedestal. Intrigued by their finding, the observations to frequencies of 250 and 4000 Hz were extended, and to signal types including tone in noise and noise in noise. Psychometric functions of three normal-hearing listeners were constructed from 2IFC adaptive tracks and fitted with a Gaussian function: $pc = \Phi(d'/\sqrt{2})$, where $d' = (x/\alpha)^\beta$, x is the signal level, α is the threshold, and β is the slope. In all conditions, slopes obtained with continuous pedestals were steeper than those obtained with gated pedestals. For tone-in-tone conditions [$x = 10 \log(1 + \Delta I/I)$], slopes differed by a factor of 2 at 250 and 1000 Hz, and by a factor of 4 at 4000 Hz. For tone-in-noise conditions [$x = E/N_0$], slopes differed by a factor of 2 across the three frequencies. For noise-in-noise conditions [$x = 10 \log(1 + S_0/N_0)$], the slopes differed by a factor of 1.5. Implications of these data for existing models of intensity coding will be discussed. [Work supported by NIH/NIDCD.]

4aPP3. Masking of pure tones by sinusoidally amplitude-modulated tonal maskers. Melanie J. Gregan, Sid P. Bacon, and Jungmee Lee (Psychoacoust. Lab., Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ 85287-1908, spb@asu.edu)

In experiment 1, masking patterns were obtained with an 80-dB SPL, 500-ms sinusoidally amplitude-modulated (SAM) masker ($m=1.0$). The 30-ms signal was centered at a masker peak or masker valley. Masker frequency (fm) was 750, 1350, or 2430 Hz; signal frequency (fs) was 0.8, 0.9, 0.95, 1.0, 1.2, 1.44, or 1.62 fm . Thresholds were generally higher for a signal at masker peak. The magnitude of this effect was governed by fs/fm , rather than by fs . In experiment 2, growth-of-masking functions ($fm=1350$ Hz, $fs=1.44 fm$) were measured for a SAM masker ($m=0.5, 0.75, \text{ or } 1.0$). These thresholds were compared with those obtained with unmodulated maskers in forward or simultaneous masking. The comparisons suggest that thresholds for a signal at a peak of a SAM masker are due to simultaneous masking, while those in a valley are due primarily to forward masking when $m=1.0$ or simultaneous masking when $m=0.5$ or 0.75 . Finally, the slope of the masking functions in simultaneous masking (unmodulated masker or signal at peak of SAM masker) changed from a slope greater than 2.0 to a slope of 1.0 at the highest levels; this will be discussed in terms of basilar membrane nonlinearity. [Work supported by NIDCD.]

4aPP4. Comparisons between tone-in-noise and phase sensitivity tasks. Brian Branstetter, Bruce G. Berg, Curt Southworth, and Lisa Khuu (Dept. of Cognit. Sci., Univ. of California, Irvine, CA 92612, bgberg@uci.edu)

According to critical band theory, phase sensitivity as a function of stimulus bandwidth and detection thresholds for a tone masked by noise are both indicators of the frequency resolution of peripheral auditory filters. The relation between these two measures is examined. For the phase sensitivity task, stimuli consist of three tones with constant intensity and randomly sampled phases. Two identical sounds are played on each trial together with a third having different randomly sampled phases. Listeners identify the odd sound. The frequency separation between tones is varied according to an adaptive staircase procedure which yields an estimate of the bandwidth at which the odd sound is identified with 71% accuracy. Estimates are obtained for center frequencies of 500, 1000, 2000, and 4000 Hz. For the second task, thresholds for 500, 1000, 2000, and 4000 Hz tones masked by broadband Gaussian noise are estimated. Results show that performance levels for different conditions within each task are highly correlated, whereas across-task correlations are either zero or slightly

negative. These findings demonstrate that predictions from critical band theory do not generalize to the level of individual listeners. [Work supported by ONR.]

4aPP5. Influence of masker phase structure on tone detection by normal-hearing and hearing-impaired listeners. Van Summers and Marjorie R. Leek (Army Audiol. and Speech Ctr., Walter Reed Army Medical Ctr., Washington, DC 20307-5001)

For normal-hearing listeners, harmonic complexes with equivalent power spectra but which differ in temporal waveform shape (i.e., in phase spectra) may also differ in their effectiveness as maskers. Summing harmonic components in positive-Schroeder phase generally produces a less effective masker than a negative-Schroeder complex [Kohlrausch and Sander, *J. Acoust. Soc. Am.* **97**, 1817–1829 (1995)]. This influence of phase structure on masking effectiveness may reflect nonlinear compressive processing in a healthy cochlea [Carlyon and Datta, *J. Acoust. Soc. Am.* **99**, 2542 (1996)]. This study measured detection thresholds of normal-hearing and hearing-impaired listeners for tones masked by complexes in positive- and negative-Schroeder phase. Masker stimuli contained all harmonics between 200 and 5000 Hz of a 100-Hz fundamental with harmonics set equal in amplitude. Three probe frequencies (1000, 2000, and 4000 Hz) and three probe levels (60, 70, and 80 dB SPL) were tested. For normal-hearing listeners, positive-Schroeder stimuli were consistently less effective maskers than negative-Schroeder complexes, particularly at the 60-dB probe level. Differences in masker phase structure had less effect at high presentation levels and for hearing-impaired listeners. The results are consistent with nonlinear compressive auditory processing by normal-hearing listeners which is reduced at high presentation levels and in the presence of sensorineural hearing loss. [Work supported by NIH.]

4aPP6. Psychophysical suppression as a function of signal frequency. Jungmee Lee and Sid P. Bacon (Psychoacoustics Lab., Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ 85287-1908)

Psychophysical suppression was investigated across signal frequency (250, 500, 1000, 2000, and 4000 Hz) in a forward-masking paradigm. Masker duration was 200 ms, signal duration was 20 or 40 ms, and signal delay was 0 or 20 ms. When using a noise masker (spectrum level of 40 dB), the amount of suppression was determined by subtracting threshold in the presence of a broadband masker from that in the presence of a critical band masker. When using a tonal masker (masker level of 50 dB SPL, suppressor level of 70 dB SPL, with the suppressor frequency being 1.2 times the masker/signal frequency), the amount of suppression was determined by subtracting the threshold in the presence of the masker plus suppressor from that in the presence of masker alone. For both masker types, the amount of suppression increased as signal frequency increased up to 1000 Hz, but then reached an asymptote or decreased somewhat as signal frequency increased to 4000 Hz. [Work supported by NIDCD.]

4aPP7. Modified masking period patterns as a function of frequency region and masker bandwidth. David A. Eddins (Psychoacoust. Lab., Dept. of Speech and Hear. Sci., Indiana Univ., Bloomington, IN 47405 deddins@indiana.edu)

The auditory system must encode dynamic aspects of acoustic stimuli at frequencies throughout the audible spectrum. This coding is limited by the temporal resolution of the ear. Recent measures of modulation detection and temporal gap detection thresholds with narrow-band noises of fixed bandwidth indicate that temporal resolution does not vary with noise carrier frequency from 500 to 4400 Hz. The present study investigated temporal processing using a modified masking period pattern technique. Masked thresholds were measured for pure tones of 500, 2000, and 4000 Hz that were 400 ms in duration. Maskers consisted of narrow-band noises centered on the signal frequency that were either unmodulated or sinusoi-

dally amplitude modulated (modulation depth=1.0) at frequencies from 4 to 256 Hz. The noise bandwidth ranged from 200 to 1600 Hz at each center frequency. By comparing masked thresholds for the unmodulated and amplitude modulated noises, one may assess both within-channel (narrow-band noises) and across-channel (wideband noises) temporal processing. Preliminary results indicate that thresholds for a given masker bandwidth depend strongly on modulation frequency but vary little with signal frequency over the range tested. The data will be discussed relative to measures of temporal resolution and comodulation masking release.

4aPP8. Effect of number and frequency spacing of masker components on multitone masking. Eunmi Oh and Robert A. Lutfi (Dept. of Psych. and Dept. of Commun. Disord., Univ. of Wisconsin, Madison, WI 53706)

Neff and Green [*Percept. Psychophys.* **41**, 409–415 (1987)] reported that the masking of single tones by random-frequency multitone maskers is greatest when the masker is comprised of between 10 and 50 tonal components. In this paper we present data to suggest that such results are related to the frequency spacing of masker components. Two conditions were run. In the first, the number of tones comprising the masker was varied from 20 to 906, with frequencies chosen at random over a fixed range from 0.1 to 10 kHz. In the second, the number of tones was fixed at 20, and the frequency range over which components were chosen was varied to match the frequency spacing of components in the first condition. An adaptive two-interval, forced-choice procedure was used to measure masked thresholds for a 1.0-kHz signal. Both conditions yielded large elevations in signal threshold. This was true even after compensating for the effects of energetic masking produced by components nearest the signal. Compensated thresholds showed little dependency on component number, but decreased systematically with decreasing frequency spacing of components. The results are consistent with a model of multitone masking based on component-relative entropy [R. A. Lutfi, *J. Acoust. Soc. Am.* **94**, 748–758 (1993)].

4aPP9. Range of performance for two tasks with random-frequency context tones. Donna L. Neff, Christina J. Kessler, Toktam Sadralodabai, and Traci R. Gleason (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, neff@boystown.org)

Forty listeners were tested on two tasks known to show large individual differences in performance. In one task, termed sample discrimination, pairs of target tones were drawn from each of two overlapping frequency distributions, and listeners indicated which pair came from the higher distribution. Target tones were presented alone and in the presence of two flanking context tones that were either fixed at 500 and 4000 Hz or had Gaussian variation centered at these two frequencies. In the second task, masked thresholds were determined for a 1000-Hz tone in the presence of ten-component, random-frequency, simultaneous maskers. The maskers were drawn on each presentation from a pool of either 200 or 10 waveforms. For sample discrimination, the range of performance was large for both the no-context and fixed-frequency context conditions. For the Gaussian-context condition, few listeners exceeded chance performance. Significant effects of testing order were found, with exposure to Gaussian-context conditions degrading later performance. For masking, performance did not differ for large versus small waveform pools, and the 35-dB range of performance corroborated previous work. For both tasks, random-frequency context degraded performance for the majority of listeners. [Work supported by NIDCD.]

4aPP10. Analysis of the performance of a model-based optimal auditory processor on a simultaneous masking task. Lisa C. Gresham and Leslie M. Collins (Dept. of Elec. Eng., Box 90291, Duke Univ., Durham, NC 27708-0291, lcollins@ee.duke.edu)

Psychoacoustic experiments indicate that the human auditory system is less sensitive than predicted by classical signal detection theory on a simultaneous masking task [C. M. Reed *et al.*, *J. Acoust. Soc. Am.* **53**, 1039–1044 (1973)]. Traditionally, the difference between the matched filter and the experimental performance has been compensated for by assuming an additional internal additive noise. However, this noise has not been completely explained in physiological terms. The inconsistency may also partially be the result of the suboptimal application of signal detection theory. To investigate this discrepancy, signal detection theory is integrated with Patterson's model of the human auditory system [R. D. Patterson *et al.*, *J. Acoust. Soc. Am.* **98**, 1890–1895 (1995)]. The performance of the optimal detector for a simultaneous masking task is compared to experimental data, allowing the theoretical performance bounds to be determined and the model to be verified. The results of this work suggest that with a more appropriate application of signal detection theory it is possible to partially explain the difference in performance between the matched filter and experimental data as the result of physical processes inherent in the auditory system. Utilizing *a priori* information regarding these processes results in an approach which more accurately predicts psychophysical behavior.

4aPP11. Bandwidth of spectral resolution for the “C-O-G” effect in vowel-like complex sounds. Qiang Xu, Lawrence L. Feth (Dept. of Speech and Hear. Sci., Ohio State Univ., Columbus, OH 43210), Jayanth N. Anantharaman, and Ashok K. Krishnamurthy (Ohio State Univ., Columbus, OH 43210)

Several investigators [e.g., Chistovitch *et al.* (1979); Carson *et al.* (1975); Beddor and Hawkins (1990)] report that listeners match an adjustable single-formant to the spectral center of gravity of a two-formant signal, as long as the two formants are separated by 3.5 Bark or less. This suggests that listeners use some form of spectral averaging in processing vowel sounds. Assmann's 1991 results do not confirm the C-O-G predictions. The preliminary work [Feth *et al.* (1996)] used the double-staircase procedure to confirm the C-O-G effect for two-formant complex sounds using both periodic pulse train and white noise excitation of the filter used to generate the signals. In the work to be reported here, the relationship between the C-O-G effect and an independent measure of spectral resolution is explored. In 1977, Feth and O'Malley used two-component complex tones described by Voelcker (1966) to estimate spectral resolving power. Resolution bandwidths paralleled the growth of the critical bandwidth with center frequency, but were 2.5–3 times wider. For the same listeners, the limit of spectral averaging for the C-O-G effect and the resolution bandwidth using Voelcker tone pairs have been measured. These results will be compared with predictions of an IWAIF model. [Work supported by a grant from AFOSR.]

4aPP12. Effects of masker uncertainty on masking by familiar sounds. Eunmi Oh and Robert A. Lutfi (Dept. of Psych. and Dept. of Commun. Disord., Univ. of Wisconsin, Madison, WI 53706)

Detection thresholds for a tone in an unfamiliar tonal pattern are greatly elevated under conditions of masker uncertainty [D. L. Neff and D. M. Green, *Percept. Psychophys.* **41**, 409–415 (1987)]. The present experiment was undertaken to determine whether a similar result is obtained when the masker is a familiar sound. Fifty common environmental sounds served as familiar maskers (e.g., dog bark, door slam, phone ring, etc.). Fifty Gaussian noise samples served as unfamiliar maskers. The magnitude and phase spectra of each masker were analyzed into 906 spectral components. Masker uncertainty was introduced by using fewer than 906 components to synthesize each sound, and by selecting the frequencies of these components at random on each presentation. Masked thresholds for a 1.0-kHz signal were obtained using an adaptive, two-interval, forced-choice

procedure. Results with noise replicate earlier results showing largest elevations in threshold for 10–20 spectral components. Results with familiar sounds showed a similar pattern but with even larger elevations in threshold. Trial-by-trial analyses showed a small detection advantage for sounds rated as highly familiar by listeners. Overall the results fail to provide strong evidence that listeners use stimulus familiarity to overcome the effects of masker uncertainty. [Research supported by NIDCD.]

4aPP13. Factors in the detection of environmental sounds. Brian Gygi (Psych. Dept., Indiana Univ., Bloomington, IN 47405), Charles S. Watson, and Gary R. Kidd (Indiana Univ., Bloomington, IN 47405)

In separate experiments, detection and identification thresholds were obtained for a set of 25 common environmental sounds (e.g., dog barking, car starting). In the detection task, threshold values of the event-to-noise ratio (Ev/N) were established using an adaptive tracking procedure. The identification experiment was part of a larger individual differences study reported earlier [Watson *et al.*, *J. Acoust. Soc. Am.* **99**, 2516(A) (1996)]. A 3AFC identification task was used, with sounds presented at eight levels of Ev/N. Thresholds were estimated from the psychometric functions derived from the data. The correlation between thresholds in the two tasks indicates that the identifiability of these sounds is only weakly related to their detectability. Several models were evaluated to identify the properties that determine the detectability of environmental sounds. Leaky integrator and cross-correlation models were used to predict detection data; identification data were evaluated using a multidimensional scaling algorithm.

4aPP14. Auditory attention beyond the ear: Precues and postcues. Denis McKeown (Dept. of Psych., Univ. of Leeds, Leeds LS2 9JT, UK)

In a “precueing” experiment listeners may come to pay attention to a narrow cued frequency band and fail to monitor other frequencies. In the experiments a cueing tone was presented at some short delay before or after a target signal. In experiment 1, one of two cues at different frequencies was presented shortly before the single interval of a yes/no signal trial. Signal and cue were at the same frequency on 50% of the trials, so the cue was not informative as to signal frequency. Performance was analyzed separately for trials in which the signal occurred at the same frequency as the cue (valid trials) and when it was at a different frequency (invalid trials); the cue to signal delay varied. At short delays performance was enhanced on valid relative-to-invalid trials, but not at longer delays. In experiment 2, the cue was presented following the signal. In this post-cueing experiment subjects responded following the cue; the delay from signal to cue was varied. Performance was below chance performance on invalid trials but above chance on valid trials. These findings suggest that attended listening may modulate the processing of detected signals rather than modulate detection itself.

4aPP15. Change in envelope beats as a possible cue in comodulation masking release (CMR). Emily Buss, Joseph Hall III, and John Grose (Dept. of Surgery, Div. of Otolaryngol., Univ. of North Carolina, Chapel Hill, NC 27599-7070)

The detection advantage associated with masker envelope coherence across frequency has typically been described in terms of comparisons across auditory channels. More recently it has been suggested that analysis of the output of a wider initial filter, similar to that suggested for the TMTF, can account for the data [Berg, *J. Acoust. Soc. Am.* **100**, 1013–1023 (1996)]. In particular, a change in envelope beats was proposed as the cue to the addition of a pure-tone signal. Data are presented for the detection of a tone added to multiple maskers with coherent envelopes. In one condition a change in envelope beats is an accurate potential cue, and in others it is a much less reliable or unreliable indicator of the presence of the signal. All conditions employing maskers with coherent envelopes produce very similar thresholds, and all showing improved sensitivity over the case of detecting a signal added to a single masker centered on the

signal frequency. Results are interpreted as evidence that a change in envelope beats does not form the basis of detection in CMR. One version of the cued-listening hypothesis, an across-channel model, is reconsidered in light of these results. [Work supported by NIH-NIDCD.]

4aPP16. Further examination between the relationship of auditory filter bandwidth and temporal resolution. Elizabeth A. Strickland (AUS Dept., Purdue Univ., West Lafayette, IN 47907-1353)

In a previous paper, the relationship between auditory filter bandwidth and temporal resolution was examined in the context of an envelope detector model [E. A. Strickland, *J. Acoust. Soc. Am.* **97**, 3330(A) (1995)]. This model predicts that temporal resolution will be limited by peripheral filtering at low frequencies, and by a central limitation at high frequencies. In the previous paper, time constants as measured by temporal modulation transfer functions (TMTFs) were compared to time constants predicted from the 3-dB bandwidths of auditory filters, for TMTFs and filters measured as a function of frequency region and level. Time constants for the TMTFs covaried with those estimated from the filters for low frequencies and levels, but were constant at high frequencies, as predicted. In the present paper, the auditory filters were used in the first stage of the envelope detector model, and TMTFs predicted by the model were compared to measured TMTFs. Time constants for the TMTFs increased as frequency decreased, but not as much as was predicted by the model. The model predicts time constants well for high frequencies, where the limitation is central. [Work partially supported by a post-doctoral fellowship from NIH.]

4aPP17. Temporal gap detection thresholds measured for conditions that minimize off-frequency listening. C. Formby, L. P. Sherlock (Div. of Otolaryngol.-HNS, Dept. of Surgery, Univ. of Maryland School of Medicine, 16 S. Eutaw St., Ste. 500, Baltimore, MD 21201), and T. G. Forrest (Univ. of North Carolina, Asheville, NC 28804)

Temporal gap detection (TGD) thresholds measured for silent gaps as a function of the frequency separation between a fixed-frequency pre-gap sinusoidal marker (F_1) and a variable-frequency post-gap sinusoidal marker (F_2) may be confounded by off-frequency listening (Formby *et al.*, *Aud. Neurosci.* **3**, 1–20). Evidence of this confounding role comes from simulation experiments with a single-channel envelope detector model. TGD stimulation results suggest that listeners can improve the signal-to-noise ratio and, in turn, TGD performance by adjusting their optimal auditory filter so that it is centered between F_1 and F_2 (Forrest and Formby, *Aud. Neurosci.* **3**, 21–33). Off-frequency listening between F_1 and F_2 may result in underestimates of TGD thresholds that appear as asymmetries in detection performance for F_2 markers presented below and above the F_1 frequency. To obtain more precise estimates of TGD, unaffected by off-frequency listening, performance has been measured and will be reported for three listeners with markers spaced symmetrically and asymmetrically in linear frequency below and above the center frequencies

$F_c = 500$ and 4000 Hz. These new results will be compared with an otherwise comparable set of results that were likely confounded by off-frequency listening (Formby *et al.*, *Aud. Neurosci.* **3**, 1–20). [Research supported by NIH.]

4aPP18. Evidence from a simple two-channel model for asymptotic gap detection. C. Formby, M. J. Gerber, L. P. Sherlock (Div. of Otolaryngol.-HNS, Dept. of Surgery, Univ. of Maryland School of Medicine, 16 S. Eutaw St., Ste. 500, Baltimore, MD 21201), and L. S. Magder (Univ. of Maryland School of Medicine, Baltimore, MD 21201)

Detection of silent temporal gaps is characterized by two prominent features when measured as a function of frequency separation between two sinusoids that mark the silent gap. First, over a range of about an octave separation, gap detection thresholds (GDTs) routinely increase as the frequency difference is increased between the two sinusoids. Second, GDTs become asymptotic for greater sinusoidal frequency separations. This characteristic GDT pattern probably reflects two different processes. The first process almost certainly reflects gap detection measured within a single auditory filter or channel. The nature of the second process is less certain, but may reflect across-channel processing of the silent gap stimulus in two or more independent frequency channels. To evaluate the idea that asymptotic GDTs can be explained with a simple two-channel model, GDTs were measured as a function of frequency separation between a pre-gap sinusoid presented to the left ear (channel 1) and a post-gap sinusoid presented to the right ear (channel 2). The resulting GDTs for standard pre-gap sinusoids from 250 to 2000 Hz correspond closely with asymptotic GDTs measured for three listeners for the same set of stimulus conditions presented monaurally. This correspondence of the data supports a two-channel hypothesis. [Research supported by NIH.]

4aPP19. Optimal interstimulus interval for auditory discrimination of intensity, duration, and frequency for normal adults. Amitava Biswas, Charles Watson, Steven Barlow, Brian Gygi, and Ward Drennan (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405)

Ten normal-hearing college students performed two-interval, two-alternative, forced choice discrimination tasks designed to determine the optimal range of interstimulus (ISI) intervals for auditory discrimination of intensity, duration, and frequency. Stimuli were randomized (roved) along frequency, intensity, and duration dimensions to reduce monotony and minimize the listener's reliance on a remembered standard. The center frequency was 500 Hz, the center duration was 30 ms, and the center intensity was 75 dB SPL. Differences to be discriminated were from 0.75 to 3.0 dB for intensity, 1.5 to 6 ms for duration, and 7 to 20 Hz for frequency. Interstimulus intervals were studied from 70 ms to 15 s. Maximum discrimination scores were observed in the range of ISIs from 1.0 to 3.0 s, for all three stimulus dimensions. These preliminary findings suggest the operation of a common, dimension-independent, sensory processing mechanism within the central nervous system. [Work supported by NIH and Neuro Logic, Inc.]

Session 4aSAa

Structural Acoustics and Vibration: Active Classification of Structures

Guillermo C. Gaunaud, Chair

*Naval Surface Warfare Center, Carderock Division, Code 684, West Bethesda, Maryland 20817-5700***Invited Papers**

8:00

4aSAa1. Mundane issues in active classification for naval purposes. Tommy G. Goldsberry (Appl. Res. Labs., Univ. of Texas, Austin, TX and Office of Naval Res., 800 N. Quincy St., Arlington, VA 22217-5660, goldsbt@onr.navy.mil)

The recent revolution in computational capability, combined with finite-element analysis techniques, has greatly expanded the ability to analyze vibrational modes of simple and complex structures and acoustic radiation from these structures. These developments have encouraged belief that one may be able to categorize or classify complex structures of interest to the U.S. Navy (e.g., submarines, mines, or swimmers) by analysis of the acoustic energy reflected and reradiated from such structures (echo structure). A detailed discussion of this topic is beyond the scope of this meeting. However, academic researchers wishing to obtain support for their research may benefit from some discussion of the importance of practical limitations such as finite transducer bandwidth, multipath propagation, and population densities on the relevance of classification methods for naval purposes. Some of these issues will be discussed in this presentation.

8:25

4aSAa2. An architecture for wave interaction with complex environments. Leopold B. Felsen (Dept. of Aerosp. and Mech. Eng. and Dept. of Elec. and Comput. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

Active classification of structures is an example of wave interrogation of complex environments. An algorithmic architecture is suggested that decomposes global complexity into problem-dependent clusters which interact across interfaces. A cluster is treated either analytically (preferably based on good physics) or numerically, deterministically and/or stochastically, etc., and the interconnects between clusters map representations on one side into those on the other side, subject to the continuity requirements of the relevant field equations. Projecting the field problem onto the cluster boundaries gives rise to a network formulation that impacts the analytic as well as numerical strategies. This field-based approach is coupled to, and modified by, considerations of signal processing. The challenge in the implementation is to effect an "optimal" decomposition into tractable clusters, and to treat each cluster problem in a fashion that anticipates its role as part of a larger conglomerate, taking into account the limitations imposed by the intended pre- and post-processing. Example scenarios are presented to illustrate these concepts.

8:50

4aSAa3. Active classification of submerged structures by means of wave packet decomposition. Nai-chyuan Yen and Louis R. Dragonette (Physical Acoust. Branch, Naval Res. Lab., Washington, DC 20375-5320, yen@noctilio.nrl.navy.mil)

This paper addresses the exploitation of the information contained in a signal by means of wave packet decomposition. This general technique has a wide range of signal processing applications: The application considered here is sonar systems with a particular emphasis on providing a broader view of time-frequency waveform analysis for target classification. The acoustic response of a submerged object depends on its physical shape and structure and on its elastic properties. A methodology of nonorthogonal decomposition is used to reduce the scattered echoes into wave packets associated with the important scattering mechanisms. This decomposition method is based on the energy distribution of the individual components and can be related to the structural scattering physics. In the case of a sonar target, mechanisms such as specular reflections, creeping waves, Bragg waves, and Bloch waves from the different parts of target or scattering centers can be incorporated into the target characterization. By applying pattern recognition logic, the present study can serve as a useful background for new sonar system development with advanced processing techniques and state-of-the-art computer hardware. Examples from numerical simulations and laboratory measurements are used to discuss the effectiveness of this target classification scheme. [Work supported in part by ONR.]

9:15

4aSAa4. Techniques for active classification of underwater structures. G. C. Gaunaud (Naval Surface Warfare Ctr., Carderock Div., Code 684, W. Bethesda, MD 20817-5700), H. C. Strifors (Natl. Defense Res. Est., S-17290 Stockholm, Sweden), P. W. B. Moore (NRaD, San Diego, CA 92152-6267), and H. Huang (Bowie, MD 20715)

Various techniques for the detailed classification of submerged shells insonified by short pulses from either an active sonar or a small explosion are discussed [Ultrasonics **33**, 147–153 (1995)]. The returned shell echoes in several signal domains are examined. These included the frequency, the time, and particularly the *joint* time-frequency domain. The use of *Wigner-type* distributions was

most informative in the latter case. Selected features in these echo-displays which provided information about a certain specific target characteristic were identified. The achieved “*in situ*” classification is rapid, unambiguous, and accurate. Examples dealing with short pulses simultaneously scattered by one, two, or more elastic shells will be shown. Theoretical predictions and measurements show good agreement. This novel analysis in the above domains determines the size, shape, wall thickness, material composition of the shell(s), and their possible filler substance(s), as well as the number of shells involved, and their ranges from the projector. The basis of this technique has been proposed as a possible explanation of how echolocating dolphins successfully identify submerged structures [J. Acoust. Soc. Am. **100**, 2820 (1996)].

Contributed Paper

9:40

4aSAa5. Broadband mine detection and classification—Target responses from a set of 2- to 20-kHz shallow-water experiments.

Peter J. Kaczowski, Christian J. Eggen, and James C. Luby (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105-6698, peter@apl.washington.edu)

The problem of discriminating between underwater mines and similarly sized false targets at ranges of several hundred meters remains a challenge. The Applied Physics Laboratory, University of Washington, in conjunction with Arete Engineering and Technologies Corporation, San Diego, have been developing detection and classification signal processing

schemes using data collected in several shallow-water experiments with a broadband (2–20 kHz) sonar. The use of relatively low frequencies for mine hunting permits greater penetration into bottom sediments for improved detection of buried or partially buried mines. Wide bandwidth signals mitigate the lack of spatial resolution expected from a narrow-band analysis at such frequencies, and provide a wealth of information for classification. A brief description of the sonar and of the experiments conducted in Puget Sound using mines and minelike false targets will be given. Target responses as a function of time, frequency, and aspect angle will be presented, as will results of using several feature extraction methods to reduce the dimensionality of the data.

THURSDAY MORNING, 19 JUNE 1997

ROOM Q, 10:10 TO 11:40 A.M.

Session 4aSAb

Structural Acoustics and Vibration: Scattering from Elastic Structures

Philip L. Marston, Chair

Physics Department, Washington State University, Pullman, Washington 99164

Contributed Papers

10:10

4aSAb1. Novel formalism for resonance scattering of acoustic and elastic waves. Huinam Rhee^{a)} and Youngjin Park (Dept. of Mech. Eng., KAIST Yusung-Ku, Taejon 305-701 Korea, email:hnrhee@ns.kopec.co.kr)

A novel formalism is proposed for the exact isolation of resonances from scattered waves for acoustic wave scattering from elastic or liquid bodies. The resonance scattering function (RSF) consisting of purely resonance information is proposed. Both magnitudes and phases of isolated resonances can be correctly obtained by using the proposed formalism while previous works based on the conventional resonance scattering theory (RST) can produce only magnitude correctly. The reason why previous works using RST could obtain correct magnitude information of resonances for the case of acoustic wave scattering is explained. Acoustic wave scattering from a variety of submerged bodies is analyzed by utilizing the RSF and the isolated resonances were compared with previously published studies. The exact π -phase shift through resonances and at the anti-resonances caused by the interaction between adjacent resonances, which has never been obtained by using RST, shows that the proposed formalism properly extracts the resonances from scattered waves. Due to the differences in phases, the novel formalism and the conventional RST generate different resonance spectrum. For elastic wave scattering, even magnitudes of resonances isolated by the proposed formalism will be different from those obtained by the conventional RST. ^{a)}Also at Dept. of Mech. Eng., KOPEC, 150 Duckjin-Dong, Yusung-ku, Taejon, 305-353 Korea.

10:25

4aSAb2. Scattering calculations by a hybrid GTD-boundary integral equation method. Paul E. Barbone and Ofer M. Michael (Dept. of Aersp. and Mech. Eng., Boston Univ., Boston, MA 02215)

We consider the application of a hybrid asymptotic/boundary integral equation (BIE) method to the problem of scattering from prismatically shaped objects. The hybrid method is based on patching a short wavelength asymptotic expansion of the scattered field to a BIE evaluation of the near field. In patching, the diffracted field shape functions with unknown amplitude are forced to agree smoothly with the solution in the near field along a curve at a prescribed distance from the diffraction points. This allows us to replace the original boundary value problem with an asymptotically equivalent boundary value problem, the domain of which is small and efficiently discretized. Since the domain of the numerical problem is small and may be chosen at will, we completely circumvent non-uniqueness problems associated with “forbidden frequencies.” Thus very high-frequency calculations can be performed using single layer potential equations with no problems of ill conditioning. The hybrid scattering solution shall be compared to a complete analytic field representation found using matched asymptotic expansions. [Work supported by ONR.]

10:40

4aSAb3. Backscattering of sound from ribbed plates and cylinders with oscillators attached to the ribs. Allan D. Pierce (Dept. of Aersp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02115)

A sequence of simple models is discussed to identify and quantify key theoretical issues pertaining to how such oscillators can affect sound scattering. The first such model consists of a sound wave incident obliquely on

a plate with a single rib which has a continuous smear of oscillators along it. The presence of the rib alters the apparent natural frequency of the oscillators and causes the backscattered sound to be anomalously high when the incident sound has the same frequency as the apparent natural frequency. The width of the spectral line in the plot of backscattered cross section versus frequency for the fixed angle of incidence is controlled both by the radiation of flexural waves away from the rib and by the overall loss of energy through radiated sound. Other models considered are an infinitely long cylinder with a single rib, smeared oscillators along the rib, an infinitely long cylinder with a single oscillator attached to a rib, a plate with a periodic array of ribs with attached oscillators, and an infinitely long cylinder with a similar hierarchy of configurations.

10:55

4aSAb4. Meridional leaky ray amplitudes for tilted cylinders and high-frequency backscattering enhancements for finite cylinders. Philip L. Marston (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814)

Consider the meridional plane which contains the axis of a circular cylinder and the incident acoustic wave vector. The meridional leaky rays which lie in this plane may be excited for certain tilt angles. Radiation resulting from the reflection of such rays off truncations on a cylinder gives strong high-frequency backscattering enhancements [Kaduchak *et al.*, *J. Acoust. Soc. Am.* **100**, 67–71 (1996)]. The present work gives ray approximations for such backscattering contributions and for the related contributions to scattering by tilted infinite cylinders. These are evaluated using a general convolution integral formulation [P. L. Marston, *J. Acoust. Soc. Am.* **100**, 2820(A) (1996)]. The infinite cylinder results for radiation by Rayleigh waves on a solid cylinder agree with the partial-wave series solution. The truncation enhancement for a finite cylinder causes a backward-directed three-dimensional wavefront possessing a vanishing Gaussian curvature. The amplitude from the far-field propagation integral associated with the caustic due to this wavefront can significantly exceed the amplitude for reflection off a rigid sphere having the same radius as the cylinder. The analysis does not rely on thin shell assumptions. [Work supported by the Office of Naval Research.]

11:10

4aSAb5. Analytical study of the acoustic scattering from a submerged stiffened cylindrical shell with two endplates. Azriel Harari and Jeffrey E. Boisvert (Naval Undersea Warfare Ctr. Div. Newport, Newport, RI 02841-5047, harari@caspr5.npt.navy.mil)

A mathematical model is described for the sound scattering from a finite cylindrical shell with two endplates submerged in a fluid, subjected to an incident wave impinging on the shell from a source in an arbitrary direction. The effect of structural damping and rib-stiffeners on the monostatic scattered response of the structure is considered for a full range of incident wave angles over the nondimensional frequency range $ka = 0.1–30$.

11:25

4aSAb6. Scattering and attenuation of an elastic wave by a viscoelastic cylinder. II. SV-wave incidence. Yin-Bin Liu and Chong-Fu Ying (Inst. of Acoust., Chinese Acad. of Sci., Beijing, 100080, PROC)

The scattering and attenuation of a shear wave by a circular, elastic cylinder with and without intrinsic attenuation are investigated. The analytical solutions for scattered and internal fields caused by a normally incident plane SV wave are derived. Resonance scattering, radiation pattern, scattering cross section, and synthetic seismogram are calculated. The calculated synthetic seismograms show that the creeping (diffraction) waves include the shear creeping wave $S_1\hat{S}_1S_1$ and exciting (or converting) compressional creeping wave $S_1\hat{P}_1P_1$. The creeping waves propagate on the surface outside the media and depend mainly on the outside media, and are little affected by the cylinder—absorbing property. The total fields are the superposition of the geometrically transmitted waves ($S_1P_2P_1$, $S_1\hat{S}_2P_1$, and $S_1\hat{S}_2S_1$) which go through the cylinder and creeping waves ($S_1\hat{S}_1S_1$ and $S_1\hat{P}_1P_1$) which propagate on the elastic side of the elastic–cylinder interface. The first arrivals within the shadow zone are a diffraction wave $S_1\hat{P}_1P_1$ for a low-velocity inclusion and a transmitted wave $S_1P_2P_1$ for a high-velocity inclusion. [Work supported by NSF of China.]

THURSDAY MORNING, 19 JUNE 1997

DEANS HALL II, 9:00 TO 11:00 A.M.

Session 4aSC

Speech Communication: Perception and Processing of Spoken Words (Poster Session)

Paul A. Luce, Chair

Department of Psychology, State University of New York at Buffalo, Park Hall, Buffalo, New York 14260

Contributed Papers

All posters will be on display from 9:00 a.m. to 11:00 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:00 a.m.

4aSC1. Temporal processing and phonotactic probability in spoken word recognition. Michael S. Vitevitch, Paul A. Luce (Dept. of Psych., Univ. at Buffalo, Buffalo, NY 14260), and Jan Charles-Luce (Univ. at Buffalo, Buffalo, NY 14260)

This study examined the influence of phonotactic information on spoken language recognition in adults. Subjects performed a lexical decision task on 120 phonotactically legal bisyllabic CVCCVC nonsense words that varied in their phonotactic probability and primary stress placement. Subjects responded faster to nonsense words containing low probability

initial syllables and slower to nonsense words containing high-probability initial syllables. On the other hand, subjects responded faster to nonsense words containing high probability final syllables and slower to nonsense words containing low-probability final syllables. Our results suggest that two phonotactically based processes may be used during word recognition. The first process is a winnowing of candidate words based on their phonotactic probability. The second process is a confirmatory process in which the lexical status of the sole candidate is confirmed based on the phonotactic probability of the item. [Work supported by NIH-NIDCD.]

4aSC2. The effect of word-final phonemes on spoken word recognition. Caroline S. Miner (Dept. of Psych., Univ. of Connecticut, 406 Babbidge Rd., U-20, Storrs, CT 06269-1020, csm93003@uconnvm.uconn.edu), Carol A. Fowler (Haskins Labs., New Haven, CT 06511-6695), and Jay G. Rueckl (Univ. of Connecticut, Storrs, CT 06269-1020)

Three experiments tested subjects' ability to recognize spoken words based on word-final phonological information. Experiment 1 replicated the finding [A. Salasoo and D. B. Pisoni, *J. Memory Lang.* **24**, 210–231 (1985)] that when presented with phonological information incrementally beginning from word-offset (backward gating), subjects correctly identified spoken words in the absence of word-initial phonemes. Additionally, experiment 1 demonstrated that acoustic neighborhood size is a significant predictor of recognition probability for both forward and backward gated words. In experiment 2, word-initial and word-final acoustic word fragments were used as primes in a crossmodal identity priming task with a naming response. Both word-initial and word-final primes significantly facilitated subsequent naming reaction times. Experiment 3 replicated experiment 2 using an associative priming paradigm. The results are interpreted from a connectionist perspective. [Work supported by NICHD.]

4aSC3. Effects of phonotactic probability on segmentation of words in continuous speech. Daniel E. Gaygen and Paul A. Luce (Lang. Percept. Lab., Dept. of Psych. and Ctr. for Cognit. Sci., SUNY, Buffalo, NY 14260-4110)

At the last meeting of the Society, evidence was presented regarding the role of phonotactic probability in the segmentation of spoken words in continuous speech. Results of further studies examining this issue will be presented. Participants made speeded word detection responses to sequences of spoken stimuli composed of target words preceded and followed by nonwords (i.e., NONWORD–TARGET WORD–NONWORD). Speed and accuracy of detection were examined as a function of the nonword contexts of the target words. In particular, probabilities of segmental transitions from the nonword contexts to the target words were manipulated: Pairs of segments composed of the last segment of the preceding nonword and the first segment of the target word, as well as pairs composed of the last segment of the target word and the first segment of the following nonword, were varied in terms of intraword transition probability (HIGH and NONE) and position specific segment probability (HIGH and LOW). Only CC transitions were used. Both spliced and coarticulated stimuli were used. The implications of our results for the use of phonotactic probabilities in the identification of words in fluent speech will be discussed.

4aSC4. The adaptive value of connotation in speech perception. Lee H. Wurm (Dept. of Psych., SUNY, Binghamton, NY 13902-6000, lwurm@binghamton.edu) and Douglas A. Vakoch (Vanderbilt Univ., Nashville, TN 37240)

In previous work it was demonstrated that reaction times (RTs) in a lexical decision paradigm were related to words' ratings on the dimensions of evaluation, potency, and activity (independent of word frequency). This was found to be true of words in general [D. A. Vakoch and L. H. Wurm, *Cognition and Emotion* (in press)], and also of words from the affective lexicon [L. H. Wurm and D. A. Vakoch, *Cognition and Emotion* **10**, 409–423 (1996)], but there were differences in the way these dimensions appeared to be used in each case. For emotion words, RTs were quickest for words that were rated as bad, strong, and fast (the evaluation \times potency \times activity interaction was significant), a finding interpreted in terms of the adaptiveness of danger avoidance. For words from the general lexicon, RTs were quickest to words that were rated as either *good* and strong, or *good* and fast, as though these represented resources that had to be seized quickly or lost. The current study clarifies the earlier findings, and expands the existing theory to include concepts such as the danger and usefulness of referents. [Work supported by AFOSR and NIMH.]

4aSC5. Word recognition skills of children with phonological disorders. Jan Edwards, Benjamin Munson, and Robert A. Fox (Dept. of Speech and Hearing Sci., Ohio State Univ., 110 Pressey, 1070 Carmack Rd., Columbus, OH 43210)

There has been little research on word recognition skills of phonologically disordered children, although recent work has shown they tend to perform more poorly than age peers on phoneme-identification tasks using synthetic speech stimuli. Six phonologically disordered preschool-aged children and six typically developing controls were asked to identify CVC words in which either the final consonant or the medial vowel had been removed in steps. The stimuli were digitized natural productions of familiar words. For the silent-center task, the phonologically disordered children were significantly less accurate than peers for the two conditions with the least acoustic information. For the gating task, the phonologically disordered children were significantly less accurate than peers for the two conditions with the most acoustic information. They were less accurate than peers at identifying digitized words with no acoustic information deleted, although they were not less accurate with live voice. This suggests that perceptual representation of final consonants for the phonologically disordered children is so fragile that even such a small degradation as a reduction in bandwidth affects their performance. Divergent patterns of group differences for the two tasks are probably related to differences in the nature of acoustic cues for medial vowels and final consonants.

4aSC6. Phonological variation in spoken word recognition. Thomas Deelman, Brian Lang, and Cynthia M. Connine (Dept. of Psych., SUNY, Binghamton, NY 13902-6000)

A common form of phonological variation in American English is flapping. For example, the word "pretty" may be pronounced as [prIDi] or [prIti]. A phonological priming experiment was conducted where a lexical decision was made on a flap or canonical target. The target was preceded by itself (repetition condition), by its variant (form condition), or by a phonologically unrelated word (control). The results showed that priming for a canonical variant was comparable in the repetition and form condition. In contrast, priming for flaps was reduced in the form condition compared to the repetition condition. These findings suggest that the underlying voiceless stop is recovered when processing a flapped variant. The results are discussed in terms of surface versus underlying representations as the currency of lexical activation.

4aSC7. The effects of aging on specificity of memory representations for spoken words. Mara B. Goodman, Paul A. Luce (Lang. Percept. Lab., Dept. of Psych. and Ctr. for Cognit. Sci., SUNY, Buffalo, NY 14260-4110, mara@deuro.fss.buffalo.edu), Jan Charles-Luce (SUNY, Buffalo, NY 14260), and Emily A. Lyons (SUNY, Buffalo, NY 14260)

Recent evidence suggests that lexical items are stored in memory as detailed exemplars that encode specific information about characteristics of speaker's voice. Lyons and Luce (1996) found that an incongruity in talker's voice (male/female) in an explicit recognition task led to slowed reaction times for college-aged participants. While this finding suggests that speaker information is preserved in stored exemplars, it can be asked whether lexical representations of aged individuals are equally detailed. Namely, it has been suggested that with age, phonological representations become more abstract or holistic in nature. If this is true, then a mismatch in talker voice should not affect the performance of aged participants. In order to explore this hypothesis, aged and college participants were tested in an explicit memory task. Preliminary results suggest that older adults are, in fact, more rather than less sensitive to variations in the surface characteristics of spoken words.

4aSC8. The interaction of lexical competition and semantic context in spoken word recognition by younger and older adults. Mitchell S. Sommers and Stephanie M. Danielson (Dept. of Psych., Washington Univ., Box 1125, St. Louis, MO 63130)

The present study examined the interaction of two operations, discrimination of lexical candidates and use of semantic context, that are central to the early stages of spoken word recognition. In addition, the investigation was designed to evaluate age differences in the interaction of these two processes. Identification scores were obtained for lexically hard (items phonetically similar to many high-frequency words) and easy (items phonetically similar to only a few low-frequency words) stimuli in three contexts: single words, low-predictability (LP) sentences, and high-predictability (HP) sentences. Although significant differences between easy and hard words were observed in all three contexts, the effects of lexical difficulty were attenuated in the HP, relative to the SW and LP conditions. The principal finding with respect to age was that older adults exhibited greater benefits of contextual information for hard, but not for easy words. The findings are discussed in terms of their implications for the interactive nature of spoken word recognition and for understanding age-related declines in speech processing. [This research was supported by the Brookdale Foundation.]

4aSC9. Positional neighborhood effects on spoken word recognition. Shigeaki Amano (NTT Basic Res. Labs. and Dept. of Psych., Indiana Univ., Bloomington, IN 47405, amano@harmonic.psych.indiana.edu), Gina M. Torretta (Indiana Univ., Bloomington, IN 47405), and Paul A. Luce (Univ. at Buffalo, Buffalo, NY 14260)

Previous work has shown that the composition of similarity neighborhoods has demonstrable effects on spoken word recognition. In a reanalysis of previously collected data of word identification and lexical decision, the effects of similarity neighborhoods on spoken word recognition were further explored by asking two specific questions. (1) Do neighbors at each phoneme position within a spoken word have equivalent effects on word

recognition? (2) Is average neighborhood frequency or the highest neighborhood frequency the superior predictor of recognition performance? Our results show that the number of neighbors defined on the basis of initial phoneme substitution affects word recognition more than the neighbors at other phoneme positions. And, it is found that the word with the maximum frequency in a neighborhood is as good a predictor of recognition performance as the average neighborhood frequency. The implications of these findings for theories of spoken word recognition will be discussed.

4aSC10. Hearing-impaired perceivers' encoding and retrieval speeds for auditory, visual, and audiovisual spoken words. Philip F. Seitz (Army Audiol. & Speech Ctr., Walter Reed Army Med. Ctr., Washington, DC 20307-5001, seitz@wrair-emhl.army.mil)

Perceptual encoding and memory retrieval processing speeds were assessed for spoken words in 26 subjects, mean age 66, with mild to moderate acquired sensorineural hearing loss. Subjects were trained to achieve error-free recognition of a set of ten spoken words in auditory, visual (speech reading), and audiovisual conditions. They then performed the Sternberg item recognition task in each of the modality conditions using the same set of ten words. The task involved presenting memory sets of one to four words, followed by a probe word to which subjects made a speeded "YES" or "NO" button response to indicate whether the probe matched any of the memory set items. Least-squares linear models provided good fits to subjects' memory-set size by reaction time functions (mean $r^2 > 0.90$ for all three conditions). Using the models' intercepts and slopes to represent encoding and retrieval times, respectively, Wilcoxon tests showed significant differences among the conditions with respect to both encoding and retrieval speed, with audiovisual fastest and visual slowest. These results are interpreted as evidence for: (1) audiovisual "benefit" to processing speed in hearing-impaired speech perception; (2) relative inefficiency of encoding visual speech; and (3) representation differences associated with the modalities. [Work supported by NIH-NIDCD.]

THURSDAY MORNING, 19 JUNE 1997

ROOM H, 8:00 TO 11:45 A.M.

Session 4aSP

Signal Processing in Acoustics: Model Based Signal Processing for Acoustic Propagation

David I. Havelock, Chair

National Research Council, Institute for Microstructural Sciences, M-36, Montreal Road, Ottawa, Ontario K1A 0R6, Canada

Chair's Introduction—8:00

Invited Papers

8:05

4aSP1. Model-based signal processing in acoustics: An overview. James V. Candy (Lawrence Livermore Natl. Lab., Univ. of California, P.O. Box 808, L-437, Livermore, CA 94551)

When the signal information to be extracted from a particular measurement is complex and severely contaminated with noise, then more information about the underlying physics, measurement dynamics, and noise must be incorporated into the processor. This concept defines the basic model-based approach to signal processing. Here, the development of model-based signal processing techniques applied to a diverse set of acoustic problems is discussed: (1) the extraction of sounds from a prosthetic heart valve for failure detection; (2) the enhancement of laser ultrasonic sounds in materials for nondestructive evaluation; and (3) the extraction of parameters from ocean acoustic measurements for target localization. Recall that model-based signal processing is a well-defined methodology enabling the inclusion of process, measurement, and noise models into sophisticated processing algorithms. The basic principle relies on the notion that the more information incorporated about the physics, measurements, and noise into the processor, the better (smaller error variance) the enhancement. When these parameters exactly "match" the process, then minimum variance can be achieved, yielding the best (in a mean-squared sense) possible results. Thus model-based processing finds its roots in minimum variance design where a cost function is investigated until it statistically yields the minimum error between predicted and measured values.

8:30

4aSP2. Model-based towed array processing. Edmund J. Sullivan (Code 82101, Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, sullivan@tech.npt.nuwc.navy.mil)

Model-based processing is a method of enhancing processing performance (smaller error variance) by the inclusion of physical models of the signal/medium, noise, and measurement systems into the processor. It is based on a state-space approach, which leads to Kalman type estimators. A major advantage of this approach is that it allows for the stochastic aspects of the problem to be included in a natural and self-consistent manner, thereby allowing for modeling errors to be accommodated by the processor. In this work we describe the application of model-based processing to the problem of processing towed array data. Here the focus is on four problems of interest in the field of towed array processing: (1) the bearing and source frequency estimation problem for plane waves; (2) the bearing, source frequency, and range estimation problem for the circular wavefront signal model; (3) problems (1) and (2) generalized to the case of the Rayleigh fading channel; and (4) the case of the stochastic broadband signal model. Results based on simulated data will be shown which will demonstrate the improvement in performance over conventional array processing techniques. In particular, it will be shown how this approach provides a passive synthetic aperture effect in a natural way.

8:55

4aSP3. Performance limits on underwater acoustic matched-field processing. Joseph Tabrikian and Jeffrey L. Krolik (Dept. of Elec. and Comput. Eng., Duke Univ., Box 90291, Durham, NC 27708)

Sensor array processing methods which use full-wave models of complex multipath propagation to facilitate estimation of underwater source locations and channel parameters are known to be prone to ambiguities and environmental variability. Particularly at low signal-to-noise ratios, the performance of matched-field methods may therefore deviate radically from the Cramer–Rao bound which assumes estimation errors are small. To study matched-field processing in the presence of anomalies, this paper considers the Barankin bound. The Barankin bound is particularly useful for determining the threshold signal-to-noise ratio below which any unbiased estimate of the source location or environmental parameters will be prone to large errors. In this paper, the Barankin bound is compared with a maximum likelihood (ML) estimator of source location in an uncertain shallow-water scenario. The results indicate that the Barankin bound predicts the threshold signal-to-diffuse-noise ratio of the ML method to within 3 dB. Further, in the presence of surface shipping, the impact of both vertical and horizontal spatial decorrelation of the interference on matched-field performance is studied for different array geometries. [Work supported by ONR.]

9:20

4aSP4. Linear and nonlinear sound wave propagation in turbulent and inhomogeneous media. Vladimir A. Krasilnikov (Dept. of Acoust., Moscow State Univ., Moscow 119899, Russia)

The author had the pleasure of being involved in the pioneering experiments on sound wave propagation in the turbulent atmosphere (1939–1950). The obtained results were interpreted from the point of view of the, then recently developed, Kolmogorov–Obukhov (KO) statistical theory of locally isotropic turbulence. These results were one of the first confirmations of this theory. It may be said that the KO theory and the experiments mentioned above had stimulated the broad developments in the problem of “waves and turbulence.” The resulting classical theory now provides a basis for the analysis of signals and the computer simulation of sound propagation through turbulence. This paper provides an historical overview of the development of the classical theory for sound propagation through turbulent media. Models of turbulence and experimental verification of these models will be discussed for sound propagation in the atmosphere, as well as in seismic and underwater media. Some examples of the problem of “wave turbulence and nonlinearity” also will be discussed.

9:45–10:00 Break

10:00

4aSP5. Some intrinsic differences between atmospheric and ocean acoustic parabolic equation models. Kenneth E. Gilbert and Xiao Di (Appl. Res. Lab. and the Graduate Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

It is often assumed that propagation models developed for ocean acoustics can be applied with minimum modification to atmospheric sound propagation. Experience shows that one often finds subtle but significant problems that can lead to serious numerical (prediction) errors, even in cases where the atmospheric environmental inputs seem similar to those in ocean acoustics. In this paper three sources of error and methods for reducing the errors to acceptable levels are discussed. The first error was recently discovered by Erik Salomons [E. M. Salomons, “Improved Green’s function parabolic equation (GFPE) method for atmospheric sound propagation,” *J. Acoust. Soc. Am.* (submitted)] and concerns the failure of standard FFT algorithms when applied to atmospheric split-step parabolic equation calculations for propagation over hard ground surfaces. The second problem concerns accurate numerical treatment of the surface wave that is excited by a point source over a locally reacting ground surface. The last issue discussed is the need for improved approaches to implementing an outgoing-wave boundary condition in calculations of long-range outdoor sound propagation. Numerical examples are presented showing the effect of the errors on the predicted signal. [Work supported by the Army Research Laboratory and the Pennsylvania State University Applied Research Laboratory.]

10:25

4aSP6. Atmospheric propagation modelling. John M. Noble (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783)

With the developments in modeling the propagation of sound through the atmosphere, it is now possible to merge these models with sensor models to determine the impact of the atmosphere on acoustic sensor performance. Early numerical models such as the Fast-Field Program (FFP) allowed users to predict the effects of refraction, geometric spreading, diffraction, and complex ground

impedance on sound propagation in the atmosphere. Coupling these numerical models with models for acoustic arrays, gave researchers the ability to predict the effects of the mean atmosphere on array performance. The newer numerical models like the Parabolic Equation (PE) and Green's Function PE (GFPE) have allowed researchers to incorporate homogeneous and isotropic turbulence and complex terrain into propagation models. These new capabilities will lead to determining the effects of turbulence and terrain on acoustic arrays. With the development of high-speed/low-power/low-cost microcomputers, it is now possible to even integrate simple propagation models into an acoustic array processor. This will allow the development of smart acoustic arrays which can determine the impact of the atmosphere on their performance and possibly modify how the array processes the data.

10:50

4aSP7. Modeling the propagation of sound through a turbulent atmosphere. Ph. Blanc-Benon and D. Juvé (Ecole Centrale de Lyon, LMFA UMR CNRS 5509, BP 163, 69131 Ecully Cedex, France, acous@ec-lyon.fr)

Recent advances in the numerical techniques and physical models used in sound propagation codes for the atmosphere have greatly improved the correspondence between calculated and measured fields. In particular, incorporating models of atmospheric turbulence have provided major advances. In this paper, an approach to turbulence modeling is described which the authors have developed in the last few years. The turbulence is represented as a superposition of a limited number of random Fourier modes and the acoustic waves are propagated through each individual realization of this field using a wide angle parabolic equation. Statistical results are obtained through ensemble averaging. Illustrations will be given for both line of sight propagation and for propagation into a refractive shadow zone. The influence of the physical model (vectorial versus scalar and Gaussian versus inertial) on the predicted sound field will be discussed. Extensions to time-domain analyses will also be considered.

Contributed Papers

11:15

4aSP8. Underwater acoustic communication over doubly spread channels. Trym H. Eggen (Dept. of Elec. Eng. and Comput. Sci., MIT, 77 Massachusetts Ave., Cambridge, MA 02139) and James C. Preisig (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543-1053)

Acoustic communication in the ocean with carrier frequency in the range 5–50 kHz has been performed in both deep and shallow water, and the communication channels can be sorted according to their time and frequency dispersion. Some of the underwater communication channels are significantly spread in both frequency and time; thus they are doubly spread channels. Scattering function estimates computed from real data of different doubly spread channels are presented, and physical scenarios that would produce the different channels are identified by simulations using ray trace and time variant FIR filters. The frequency dispersion depends in particular on relative speed between the transmitter and channel scatterers, or between transmitter and receiver for direct paths, and the eigenrays of some of the physical scenarios can have significantly different Doppler spreads. Current receivers are unable to establish reliable communication over such channels. A new receiver is proposed which can successfully demodulate signals which have propagated through some of the doubly spread channels with multiple Doppler spreads. The receiver makes use of filter taps spaced in both the delay and the Doppler dimensions. Its capabilities and limitations are demonstrated on both real and simulated data.

11:30

4aSP9. Environmentally adaptive signal processing. Leon H. Sibul and Teresa L. Dixon (Appl. Res. Lab., Penn State Univ., State College, PA 16804-0030)

To achieve required system performance under difficult propagation and reverberation conditions, signal processing systems must be environmentally adaptive. This paper uses results from statistical signal processing and wavelet transform theory to develop an estimator-correlator (EC) approach to environmental adaptivity. The EC is an optimum (in the maximum likelihood sense) detector and estimator that incorporates propagation and interference (reverberation and ambient noise) models as an essential part of its formulation. It is a doubly adaptive processor because it uses propagation models and on-line adaptation to determine conditional mean estimates of the full-field replicas that it correlates with optimally/adaptively filtered received data. The EC concept is applicable to both active and passive sonar. This paper concentrates on wavelet transform domain implementation of active EC that achieves recombination of multipath and multihighlight signals. This paper reviews the underlying wavelet transform concepts, basic EC theory and medium characterization by wideband spreading and scattering functions. Some computer simulation results are presented. [This work was supported by ONR under Grant No. N000149510965, Code ONR 313.]

4a THU. AM

Session 4aUW**Underwater Acoustics: Synthetic Aperture Sonar**

Antares Parvelescu, Cochair
Naval Research Laboratory, Code 7130, Washington, DC 20375

Frank Henyey, Cochair
Applied Physics Laboratory, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105

Invited Papers**8:00**

4aUW1. The prehistory of synthetic aperture sonar. Antares Parvelescu (Naval Res. Lab., Code 7130, Washington, DC 20375)

It is necessary to go back to 1958 (A.D.) and the summer of 1959. The concept of using recorded data obtained from a single moving element to provide a "memorized array" instead of the multiple-element physical array, was offered to ONR (Aubrey Pryce), whose reaction will be described. The years 1960–68 saw slow (mini-funded) progress at Hudson Labs of Columbia University. Two algorithms were conceived and (mini-)tested; a 6-m towed "fish" was built, then progress slowed further at the Hawaii Inst. of Geophysics. Graduate students wrote theses and bootlegged computer time; names will be named and credit will be given. The unquenched urge to "see" acoustically the seabed in two or three dimensions is at least almost here, as the next few papers show. Some formerly classified reports may have been released by the time of this meeting.

8:20

4aUW2. Efficient SAS image-reconstruction algorithms. David Hawkins and Peter Gough (Dept. of Elec. and Electron. Eng., Univ. of Canterbury, Private Bag 4800, Christchurch, New Zealand)

Now that several prototype strip-map synthetic aperture sonar (SAS) systems are in operation and producing images (albeit on an irregular basis), much of the focus of current research is on the efficiency of the various strip-map image reconstruction algorithms. Although sometimes still used by several sonar groups, the old "delay-and-sum" beamforming, or exact algorithm, is clearly inadequate for high throughput imaging. Newer algorithms such as the range/Doppler, wave number, and chirp-scaling algorithms are considered. Using a simulated target field of point reflectors, the results from all of these algorithms are compared and the relative computational loads given. As well, the more recently developed (em accelerated) chirp-scaling algorithm is explained.

8:40

4aUW3. SAMI: A low-frequency wideband prototype for synthetic aperture mapping and imaging. Manell E. Zakharia and Jacques Chatillon (CPE Lyon, LISA/LASSO, BP 2077, Bat 308, 43 Blvd. du 11 Novembre 1918, 69616, Villeurbanne Cedex, France)

A low-frequency wideband (6–12 kHz) prototype has been developed in the framework of the SAMI (synthetic mapping and imaging) project. The prototype uses a subsurface towed fish. One transmitting array and two parallel receiving arrays 2 m long were installed on the tow-fish. Each array is associated to a synthetic aperture sonar channel. Thanks to the use of suitable transmitted signals (linear period modulated chirps), array synthesis could simply be achieved in the time domain; pulse compression and azimuthal matched filtering could be separated in order to reduce the processing complexity. Synthetic image resolution was measured at sea using reflectors on a frame; the results obtained were comparable to the predicted performance: 0.5×0.15 m. The cross correlation between both synthetic aperture channels output was used for measuring the elevation of the sea bottom reverberation cells. Synthetic aperture maps were then produced. Real-time processing was achieved on board the vessel for a swath of 500 m. Data were stored and post-processing was achieved off-line. Several experimental data obtained at sea in both shallow and deep (2400 m) water are presented. [Work supported by the EC-MAST (Marine Science and Technology) program.]

9:00

4aUW4. Test results for a synthetic aperture sonar (SAS). Mark L. Neudorfer (Hughes Aircraft Co., Naval and Maritime Systems, 6500 Harbour Heights Pkwy., Mukilteo, WA 98275-4844, mark_neudorfer@atk.com), Dennis Garrood, Tony Luk (Hughes Aircraft Co., Mukilteo, WA 98275), and Matt Nelson (Dynamics Technology, Inc., Torrance, CA)

A special purpose synthetic aperture sonar (SAS) system has been developed and built by Hughes Aircraft Co. and Dynamics Technology, Inc. The main features of the system are a 3.2-m-long, 32-element linear array, and a data acquisition system capable of recording the elemental data in real time while preserving the phase and amplitude fidelity required by SAS processing. High-resolution, large (100^2 m²) and small (3^2 m²) SAS images of various objects will be presented that demonstrate resolutions of 20 cm at 1000 m. The innovation of the work is that resolution achievable only with short-range, high-frequency sonar has been demonstrated at ranges obtainable only with a low-frequency sonar. [Work supported by DARPA.]

9:20

4aUW5. Medium effects on high-frequency propagation: A recent experiment. Kevin Williams, Frank Henyey, Terry Ewart, Jim Grochocinski, Steve Reynolds, and Daniel Rouseff (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

In August 1996 an experiment was performed southeast of Long Island, NY (40 30.16 N, 70 30.58 W) to examine water column effects on synthetic aperture sonar (SAS). Both the experiment and the pre-experimental modeling that guided the experiment will be presented. Salient features of the experimental equipment and procedure include the use of bottom mounted towers for stability, transmission frequencies of 6, 20, 75, and 129 kHz, reciprocal tower-to-tower transmissions, and receivers spaced over a 10-m horizontal and 6-m vertical aperture. First look results will be presented for a 24-h period that included a solibore event. Large variations in propagation conditions over time scales of minutes are evident in the data. The SAS relevant quantities to be extracted from this data set will be discussed. [Work supported by ONR.]

9:40

4aUW6. An approximation theoretic approach to synthetic aperture processing. Raymond Fitzgerald (Vail Res. and Technol. Corp., 5285 Shawnee Rd., St. 210, Alexandria, VA 22312)

An algorithm for synthetic aperture processing derived using approximation theory is presented. Improvements offered by this approximation theoretic algorithm and the analytic relationships between the algorithm and other algorithms are described. The approximation theoretic approach is shown to provide a unified formalism for comparing and evaluating existing and proposed algorithms as well as a method for generalizing and improving them.

10:00

4aUW7. Mid-frequency synthetic aperture sonar imaging. Kenneth D. Rolt, Geoffrey S. Edelson, Bradford W. Gillespie (Sanders, A Lockheed Martin Co., P.O. Box 868, Nashua, NH 03061-0868, kdrolt@sanders.com), Stephen A. Celuzza, Philip A. Abbot, and Charles J. Gedney (Ocean Acoust. Services and Instrumentation Systems, Inc., Lexington, MA 02173)

A detailed review of many temporal stability and spatial coherence experiments is used to develop a reasonable model for the temporal and spatial limitations on mid-frequency synthetic aperture sonar (SAS) in shallow water. A comparison of deep- versus shallow-water coherence experiments is made based on an extensive review of the literature. This comparison reveals that the spatial limits for arrays often dominate over temporal limits. The upper bounds on real or synthesized array lengths in shallow water that result from this analysis are presented. The feasibility of mid-frequency SAS imaging in shallow water is shown, and the predicted resolution in range, azimuth, and depth is compared to the theoretical values for a perfectly stable ocean and precise motion compensation. As a first step to validate the spatial and temporal limitation analysis, a mid-frequency surface sonar is modeled to create artificial data for SAS imaging in shallow water. The modeled environment includes a time varying ocean surface and sound-speed profiles, bottom roughness, penetration, and scattering. Useful SAS imaging is shown. [Work supported by NUWC.]

10:20–10:35 Break

Contributed Papers

10:35

4aUW8. Design procedures for synthetic aperture sonar. Theo Kooij (Defense Adv. Res. Projects Agency (DARPA), 3701 N. Fairfax Dr., Arlington, VA 22203-1714, tkooij@darpa.mil)

An engineering approach is presented for the top down design of synthetic aperture sonar (SAS) systems. The principal system parameters and the equations governing their relations will be identified. The difference between the variables and the equations determines the number of free variables, which can be selected either arbitrarily or according to the system requirements. Some simple rules will be stated, and by example it will be shown how to apply the design steps.

10:50

4aUW9. Multistatic scattering from bottom targets in the presence of anisotropically rough interfaces. Jaiyong Lee and Henrik Schmidt (Dept. of Ocean Eng., MIT, Cambridge, MA 02139, jai@arctic.mit.edu)

The performance of active, multistatic sonar systems can be enhanced significantly, if the 3-dimensional environmental physics is incorporated into the processing. Thus fundamental understanding of the difference between target scattering and bottom reverberation could provide the basis for advanced, multistatic clutter reduction. Strong bottom interaction is the most distinct acoustic feature in shallow-water environments. Consequently, bottom roughness becomes the major reverberation mechanism limiting sonar system performance. To investigate the three-dimensional

spatial characteristics of multistatic target scattering and bottom reverberation, a unified numerical model is being developed. It combines existing theoretical models for scattering from shells of finite length, anisotropic interface scattering and seismo-acoustic wave propagation in stratified media. The model generates Monte Carlo statistics for multistatic target scattering in the presence of a strongly reverberant, rough elastic bottom. The effects of anisotropic statistics of the bottom roughness on the multistatic field are investigated, and the significance of the geometric and acoustic properties of target is addressed. [Work supported by ONR.]

11:05

4aUW10. Passive synthetic aperture sonars, revisited. Edmund J. Sullivan (Naval Undersea Warfare Ctr., Newport, RI 02841, sullivan@tech.npt.navy.mil) and William M. Carey (MIT, Cambridge, MA 02139)

Experimental and analytical studies have clearly shown the feasibility and applicability of passive and active synthetic aperture sonars (SAS). These investigations have spanned the infrasonic to ultrasonic frequency range. However, passive SAS has not gained widespread use due to three arguments. The first is based on a single hydrophone system, and knowledge of the source frequency is required for bearing estimation. However, with multiple hydrophones, both the bearing and frequency can be estimated simultaneously. The second argument claimed that there was no "new gain," since the number of spatial and temporal degrees of freedom are constant. However, this analysis failed to account explicitly for the

4a THU. AM

movement of the array, thereby ignoring the Doppler information. The third argument claimed that temporal coherence time must be at least equal to the time of synthesization. This argument ignores the applicability of sequential coherence corrections. This paper will clarify and place these arguments in perspective. Experimental results and analyses showing the performance improvements will be presented.

11:20

4aUW11. Modal mapping in a complex shallow water environment.

George V. Frisk, Laurence N. Connor, Kyle M. Becker (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), and Barry J. Doust (Penn State Univ., State College, PA 16804)

An experimental method is described for mapping the wave-number spectrum of the normal-mode field as a function of position in a complex, shallow-water waveguide environment whose acoustic properties vary in three spatial dimensions. The experimental configuration consists of a fixed source radiating one or more pure tones to a field of freely drifting buoys, each containing a hydrophone, GPS and acoustic navigation, and radio telemetry. The precisely navigated, drifting hydrophones form a two-dimensional, synthetic aperture planar array that can be used to determine the evolution of the normal modes in range and azimuth. By describing the spatially varying spectral content of the modal field, the method provides a direct measure of the propagation characteristics of the waveguide. The resulting modal maps can also be used as input data to inversion techniques for obtaining the acoustic properties of the waveguide. [Work supported by ONR and ARL Penn State Univ.]

11:35

4aUW12. Three-dimensional modal mapping in shallow water using simulated acoustic field data.

Kyle M. Becker, George V. Frisk, Laurence N. Connor (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), and Barry J. Doust (Penn State Univ., State College, PA 16804)

It is well known that the acoustic field propagated through a shallow-water waveguide is sensitive to the properties of the waveguide and its boundaries. Through adiabatic mode theory, it is understood that the modal eigenfunctions and eigenvalues adapt to the local environment and changing boundary conditions. It has been proposed that modal tracking may be employed in a laterally varying waveguide to infer properties of the normal-mode field and the environment. In support of a three-dimensional modal mapping experiment, synthetic three-dimensional acoustic fields are generated to simulate propagation in a shallow-water region in the vicinity of the East Coast *strataform* site. Synthetic fields are computed using normal-mode and parabolic equation codes and transformed to the wave-number domain to obtain estimates of the spatially varying spectral content of the modal field. The resulting modal maps are discussed for several experimental regions. [Work supported by ONR and ARL Penn State Univ.]

11:50

4aUW13. Reduced wavenumber synthetic aperture. Ira Ekhaus (Synap Corp., 100 Wildwood Ave., Arlington, MA 02174)

Underwater acoustic imaging applications such as mine counter measures and sidescan seafloor surveys would benefit from an operational synthetic aperture sonar system. While the potential of high-resolution synthetic aperture (SA) sonar designs to provide unprecedented detection and discrimination performance is clear, the real-world issues of residual platform motion and temporal/spatial variations in sound speed (SSP) have prevented operational deployment of existing designs. High azimuthal resolution requires a synthesized aperture of lengths approaching ten thousand wavelengths. This imposes a tight constraint on residual motion error and the ranges of environmental scenarios where present designs can operate. Synap Corporation has developed the reduced wavenumber synthetic aperture (REWSA) algorithm/design based on an interferometric preprocessing stage operating on multiple element baseband receptions, and a unique inversion stage. The REWSA algorithm is shown to have a reduced wavenumber characteristic that is inherently robust to the effects of the sonar environment. Discussion of resolution, spatial sampling, and aperture requirements will be supported by simulation and experimental data results. [This work was sponsored by NOAA under Contract 50-DKNA-5-00206.]

12:05

4aUW14. Space-time signal processing for complicatedly moving antennas. Igor I. Gorban (Inst. of Mathematical Machines and Systems, Natl. Acad. of Sci., 42, Ave. Acad. Glushkov, Kiev 252187, Ukraine)

The results of the author's last work in the field of space-time signal processing (STSP) were presented at the 132nd Meeting of the Acoustical Society of America [I. I. Gorban, "Space-time signal processing algorithms for moving antennae," J. Acoust. Soc. Am. **100**, 2638(A) (1996)]. It touched the two problems of STSP in complicated dynamic conditions, where there was antenna motion with variable velocity and angle rotations, and modifying of the form of the antenna. The first problem is STSP optimization in conjunction with the complicated antenna motion, the noises, and the medium together. The second is the problem of super-multichannel STSP with low calculating efforts. New approaches were proposed for solving these problems. It was found that complicated antenna motion, often regarded as a destabilizing factor, may play a greatly positive role for increasing space-time selectivity and noise immunity of the systems. Rather than struggle with this destabilized factor it was proposed that one should use it. Ways to essentially reduce calculations for complicated STSP in dynamic conditions were also found. At the 133rd Meeting, the new results of the research developing the proposed approaches, for towed arrays in particular, will be presented.

Meeting of Accredited Standards Committee S12 on Noise

to be held jointly with the

U. S. Technical Advisory Group for ISO/TC 43/SC1 Noise and ISO/TC 94/SC12 Hearing Protection

P. D. Schomer, Chair S12, and Chair U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC1, Noise
U. S. CERL, P.O. Box 4005, Champaign, Illinois 61820

B. M. Brooks, Vice Chair, S12
Brooks Acoustics Group, P.O. Box 3322, Vernon, Connecticut 06066

H. E. von Gierke, Vice Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC1, Noise
1325 Meadow Lane, Yellow Springs, Ohio 45387

E. H. Berger, Chairman, U.S. Technical Advisory Group (TAG) for ISO/TC 94/SC12, Hearing Protection
E-A-R/Aearo Company, 7911 Zionsville Road, Indianapolis, Indiana 46268-1657

Accredited Standards Committee S12 on Noise. Working group chairs will report on their progress for the production of noise standards. The interaction with ISO/TC 43/SC1 and ISO/TC 94/SC12 activities will also be discussed, with reference to the international standards under preparation. A report will be given on the activities of ISO/TC 43/SC1. The Chairs of the respective U.S. Technical Advisory Groups (P. D. Schomer and E. H. Berger) will report on current activities of these international Technical Subcommittees under ISO.

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, 19 JUNE 1997

ROOM E, 3:30 TO 5:00 P.M.

Session 4pAA

Architectural Acoustics: Distinguished Lecture

William J. Cavanaugh, Chair

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

Chair's Introduction—3:30

Distinguished Lecture

3:45

4pAA1. Dynamic multiple-use concert hall-theater design and multi-form theater design for the twenty-first century or From dampened steel and reinforced concrete to mosquito netting and back again to dampened steel and reinforced concrete. George C. Izenour (George C. Izenour Assoc., Inc., 16 Flying Point Rd., Stony Creek, CT 06405)

Except for the largest and wealthiest urban centers, it is an undisputed fact that the *raison d'être* for design of multiple-use facilities for the performing arts in municipalities of smaller size and relative wealth is an economic necessity for capital as well as for operational funding. Solutions to this many-faceted, much-disputed and much-maligned problem *vis-a-vis* owners, architects, and engineering consultants have been and continues to be many and varied. Overall in the USA, two schools of thought have emerged. One is continuation of the traditional static architectural approach inherited from the 18th-century high baroque. The other originated and developed by this consultant and favoring a dynamic engineering approach is, instead, firmly rooted in modern times. This paper, accompanied by concise graphic exposition of certain seminal historical and related contemporary realized recent designs and two proposed unrealized designs is an extrapolation into the next century of the author's technologically based, as opposed to the conventional architecturally based, approach to the problem.

THIS SESSION WILL BE FOLLOWED BY A WINE AND CHEESE RECEPTION
IN SENATE ROOM I. SPONSORED BY THE NATIONAL COUNCIL
OF ACOUSTICAL CONSULTANTS

THURSDAY AFTERNOON, 19 JUNE 1997

ROOM P, 2:00 TO 4:40 P.M.

Session 4pAB

Animal Bioacoustics: Low Frequency Bioacoustics

Adam S. Frankel, Chair

Bioacoustics, Cornell Laboratory of Ornithology, 159 Sapsucker Woods Road, Ithaca, New York 14850

Invited Papers

2:00

4pAB1. A survey of research on low-frequency acoustic communication in elephants. Katharine Payne (Bioacoust. Res. Program, Cornell Lab. of Ornithology, Cornell Univ., Ithaca, NY 14850)

Research projects by K. Payne, W. Langbauer, E. Thomas, and J. Poole have shown that Asian and African elephants make powerful infrasonic calls, some of which are used in long-distance communication. Although full classification has not been achieved, some of these calls are social, others reproductive; behavioral responses show individual recognition. Playback experiments yielded a measure of the distances over which conspecifics respond during daylight hours (Langbauer *et al.*) and of the social information imbedded in certain clearly definable calls (K. McCoun). Long-distance communication appears to be implicated in the coordinated movements between separated elephant families; a field study using radio collars with implanted voice-activated microphones yielded suggestive results (Langbauer *et al.*). A meteorological study by meteorologists M. Garstang and D. Larom led them to predict

large-scale diurnal fluctuations in low-frequency sound propagation in dry season savannas. They and Payne plan to collaborate in research on the relation between these phenomena and calling behaviors of savanna animals. [Support for the presenter's projects came from World Wildlife, National Geographic, NSF, and a Guggenheim fellowship.]

2:25

4pAB2. Low-frequency signaling behavior in mysticete whales. Christopher W. Clark (Cornell Univ., Lab. of Ornithology, Ithaca, NY 14850) and William T. Ellison (Marine Acoust., Inc.)

Mysticete whales produce a variety of low-frequency sounds. These range from the highly labile song of the humpback whale (*Megaptera novaeangliae*) to the infrasonic sounds of the blue (*Balaenoptera musculus*) and fin (*Balaenoptera physalus*) whales. Biological functions of these sounds remain largely untested, the most common being that sounds are a male reproductive display for social communication. Recent access to fixed arrays of bottom-mounted hydrophones and towed arrays has provided a novel mechanism for documenting low-frequency whale sounds throughout large ocean areas in relation to season, bathymetric features, and ocean conditions. Ocean sound propagation conditions impose severe constraints and offer dramatic opportunities for acoustic transmission for whales. The bioacoustic data in combination with sound propagation models provide novel insights into the possible biological functions for low-frequency whale sounds, and offer testable predictions. Improved understanding of low-frequency whale bioacoustics is critical for knowing the potential impact of human-made low-frequency sounds on whales, and offers insights into improved signal design. Biological functions will be presented in conjunction with empirical evidence to support, for example, that blue and fin whales navigate using reverberation and bistatics, and humpback males should prefer deep rather than shallow water singing habitat.

Contributed Papers

2:50

4pAB3. Comparison of low-frequency communication by footdrumming in three species of solitary, desert rodent, kangaroo rats (*Dipodomys*). Jan A. Randall (Dept. of Biol., San Francisco State Univ., San Francisco, CA 94132, jrandall@sfsu.edu)

The ears of desert rodents are well adapted for receiving low-frequency vibrations created during footdrumming. Kangaroo rats drum species-specific patterns, ranging from single thumps to individual footdrumming signatures. The desert kangaroo rat, *D. deserti*, drums single thumps spaced 0.25 to 0.30 s apart. The giant kangaroo rat, *D. ingens*, drums long footrolls that can average over 100 drums at 18 drum/s. The banner-tailed kangaroo rat, *D. spectabilis*, drums footdrumming signatures consisting of 3–38 footdrums in a footroll combined into sequences of 2–13 footrolls. In playback tests, all three species stood alert and entered the burrow in response to footdrumming of their own and other species. The rats also responded in species-specific ways. *D. spectabilis* drummed to its own species' footdrumming, but not to playbacks of single drums of *D. deserti*. *D. deserti* did not footdrum in response to the playbacks, but approached the speaker more frequently than either of the other two species. *D. ingens* footdrummed equally to all footdrumming playbacks. The species' differences reflect differences in social tolerance and spacing. *D. deserti* chases visitors from the burrow, *D. spectabilis* engages in frequent footdrumming exchanges, and *D. ingens* tolerates close neighbors and footdrums periodically.

3:05

4pAB4. Comparison of the low-frequency response of the peripheral auditory organs in the goldfish and oscar. Corrie Derenburger, James J. Finneran, and Mardi C. Hastings (Dept. of Mech. Eng., Ohio State Univ., 206 W. 18th Ave., Columbus, OH 43210)

A noninvasive ultrasonic measurement system was used to measure the amplitude and phase response of the peripheral auditory organs of the goldfish, considered auditory specialists, and the oscar, having an auditory threshold noticeably higher than that of the goldfish. The fish were excited with an underwater acoustic source at frequencies ranging from 12.5–3000 Hz. For frequencies between 12.5–400 Hz, tests were performed in a 14-m acoustic waveguide. For tests above 400 Hz, a 2.43-m-diam pool with a depth of 0.4 m was used. The response of the goldfish swimbladders, Weberian ossicles, and otoliths were measured. In addition, the swimbladders were deflated and measurements were made of the Weberian ossicles and the otoliths. Similar measurements were obtained for the auditory organs of the oscars, which vary from the goldfish in that they contain only one swimbladder and lack the Weberian apparatus. Comparisons within

and between species are presented based on these four groups of data. [Work supported by ONR Grant No. N00014-94-1-0337.]

3:20–3:40 Break

3:40

4pAB5. Dynamic mechanical measurements of marine mammal tissues. Edwin R. Fitzgerald, John D. Strandberg (Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218), and Brent R. Whitaker (Natl. Aquarium, Baltimore, MD 21202)

An automated measurement system for complex shear compliance, $J^* = J' - iJ''$, shear modulus, $G^* = G' + iG''$, and shear wave velocity and attenuation from 2 to 10 000 Hz at temperatures from -25 to 150 °C [E. R. Fitzgerald, Proc. Am. Chem. Soc. **60**, 573–578 (1989); U. S. Patent 5,081,870 (1992)] has been used to determine these viscoelastic parameters for blubber and skin of several marine mammals. Excised samples from two stranded harbor seals, for which rescue efforts failed, were measured at 20 °C from 2 to 1000 Hz, at short intervals, from 1.9 to 12 h after death. Values of the elastic (J') and viscous (J'') compliance components decreased with time after death, and showed sharp, "life-to-death" slope transitions at 4 to 6 h after death, as found in prior animal tissue measurements [Fitzgerald *et al.*, J. Acoust. Soc. Am. **29**, 61–64 (1957); E. R. Fitzgerald, Biorheology **12**, 397–408 (1975)]. These measurements are compared with those previously reported for pilot whale and dolphin blubber [E. R. Fitzgerald and J. W. Fitzgerald, Mater. Sci. Eng. C **2**, 209–214 (1995); Proc. Third Int. Conf. Intel. Mater. SPIE **2779**, 83–88 (1996)]. [This work was supported, in part, by the Advanced Research Projects Agency and the Office of Naval Research.]

3:55

4pAB6. Canine acoustics. II. Frequencies, transmission, and annoyance. Peter M. Scheifele (Natl. Undersea Res. Ctr., Univ. of Connecticut—Avery Point, Groton, CT 06340-6097, scheifel@uconnvm.uconn.edu), David G. Browning (Univ. of Rhode Island, Kingston, RI 02881), and Lesa M. Scheifele (The Lost Ark, Inc., Norwich, CT 06360)

In a previous paper [P. M. Scheifele and D. G. Browning, J. Acoust. Soc. Am. **100**, 2710(A) (1996)] it was found that the source level for a howling Eastern Coyote is a relatively high 93 dB. At a primary frequency of 500 Hz this howl should have a strong signal throughout a typical territory (4×4 km) under normal atmospheric transmission conditions. Based on reported [H. E. Heffner, Behav. Neurosci. **97**(2), 310–318

(1983)] canine hearing response, however, howls appear louder to humans than to canines. For low-frequency noise, such as from road traffic, the opposite is true. This brings up some interesting questions of mutual annoyance which will be discussed.

4:10

4pAB7. Potential impacts of low-frequency anthropogenic noise on the hearing of subarctic beluga whales in the Saint Lawrence estuary. Peter M. Scheifele (Marine Sci. and Technol. Ctr., Univ. of Connecticut, 1084 Shennecossett Rd., Groton, CT 06340-6097, scheifele@uconnvm.uconn.edu)

A small population of about 500 beluga whales, *Delphinapterus leucas*, resides in the Saint Lawrence estuary, isolated from other populations of its species. This population has been extensively studied over the last 12 years in terms of its seasonal distribution, size, age, group structure, pathology, and group behavior. It is now considered to be threatened by human activities including elevated noise levels due to a wide range of anthropogenic sources including merchant shipping and whale watching activities. A whale recovery plan is in development that will include hearing conservation. Ambient noise measurements were made at the three major sites of habitation of these whales to gain an understanding of the daily fluctuations in noise levels and to establish a baseline characterization of the acoustic environment at each site. The noise levels were related statistically by site, time of day, and human activity and compared to beluga hearing sensitivity curves [W. Au, *The Sonar of Dolphins* (1993)] and indicate the probability that hearing damage will occur for animals

occupying two of the three sites. Specifically, 200 Hz, 500 Hz, and 1 kHz and 40 kHz were scrutinized. [Work supported by World Wildlife Fund Canada, Humane Society Canada, and Parks Canada.]

4:25

4pAB8. Investigation of a smart trigger for an acoustic deterrent device. Dehua Huang, Steve G. Christensen, Stephen G. Boucher, and Roger H. Tancress (Airmar Technol. Corp., Meadowbrook Dr., Milford, NH 03055)

Attacks of pinnipeds on fish pens cause considerable destruction to the fish farming industry. An acoustic deterrent device (ADD) operated at 10–16 kHz with a source level around 200 dB *re*: 1 μ Pa at 1 m is an effective way of predation deterrence. The present technique is to operate the ADD system on a continuous basis. This constant insonification could lead to predator habituation and could create serious man-made acoustic noise. In this paper, techniques for creating a device to trigger the existing ADD system are investigated. The trigger will turn on the ADD system only when the fish pens are threatened. Field tests were conducted by applying both passive hydrophone and active Doppler methods to detect the presence of marauding pinnipeds. During the test, acoustic devices were located inside the fish pen or outside the fish cage, and the observation objects were fish in the pen or predators in the open sea. Design concept for the trigger devices, related DSP technique and pattern recognition, along with data collected in the field will be presented and discussed. [The principal investigator, Dr. Huang, acknowledges the funding from USDA/NRAC (Project No. 5555506).]

THURSDAY AFTERNOON, 19 JUNE 1997

ROOM F, 2:00 TO 4:45 P.M.

Session 4pEA

Engineering Acoustics: Modeling and Predictive Capabilities for Transducers II

James M. Powers, Chair

Naval Undersea Warfare Center, Code 2131, New London Detachment, Newport, Rhode Island 02841

Chair's Introduction—2:00

Contributed Papers

2:05

4pEA1. Finite-element analysis of both flexural and Helmholtz resonance modes in a flextensional transducer. Dennis F. Jones (Defence Res. Establishment Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada)

During the last decade, several types of compact piezoceramic-driven barrel-stave flextensional transducers were researched and developed at the Defence Research Establishment Atlantic (DREA) for low-frequency high-power sonar applications. These transducers were designed to operate at the fundamental flexural resonance where they radiate sound efficiently and omnidirectionally. However, an important limitation common to almost all underwater flextensional transducers is that they require some form of depth compensation for operation at water depths exceeding a few hundred meters. One technique that may eliminate the need for depth compensation (and the associated design complications) for the barrel-stave flextensional transducer is to free flood the cavity inside the cylindrical ring-stack piezoceramic driver with the surrounding acoustic fluid medium and exploit the resulting Helmholtz resonances. To examine the feasibility of such a design approach, the flexural and Helmholtz resonances of a theoretical barrel-stave flextensional transducer with a free-flooded driver have been studied using finite-element techniques, and the

results are presented in this paper. In addition, it is shown that the Helmholtz resonance in an experimental transducer does exist and may be practical.

2:20

4pEA2. A 3-kHz class VII flextensional transducer. Stephen C. Butler and Jan F. Lindberg (Naval Undersea Warfare Ctr., Newport, RI 02841)

A 3-kHz class VII flextensional transducer was developed at NUWC. The class VII flextensional is a modified class IV shape having concave radiating surfaces rather than convex. This transducer is commonly referred to as a "dogbone" flextensional because of its similar shape. This design, originally described by Merchant in U.S. Patent 3,258,738 (28 June 1966), has the advantages that all the surfaces of the shell radiate in-phase and a positive compressive stress is maintained on the ceramic driver as the operation depth increases. The shell can support thinner walls, since hydrostatic pressures do not have to be overcome in order to keep the piezoelectric ceramic driver under compression, as in a conventional class IV transducer. The transducer was designed using two different modeling approaches: finite element modeling (using ATILA) and an equivalent circuit model based on plane-wave and lumped parameters. Based on the modeled results, a transducer was fabricated and then tested in August 1996. The transducer volume is 5 in. (12.7 cm) high, 5 in. (12.7 cm) long,

and 2.5 in. (6.35 cm) wide, the shell material is stainless steel, and the ceramic stack is composed of 30 0.1-in. (2.54-mm)-thick PZT-8 plates, two of which are unpoled and used for insulators. The in-air weight is 10 lbs. The transducer performed exceptionally, matching the modeled data while producing an SL of 202 dB at 10 kV/in., a resonance frequency of 3050 Hz, a mechanical Q of 5, an in-water effective coupling coefficient of 30% (typical of class IV transducers), and an electro-acoustic efficiency of 80%. [Work supported by ONR.]

2:35

4pEA3. Nonlinear time-domain analysis of electrostrictive materials by the finite element method. Jocelyne F. Coutte-Dubois, Jean-Claude Debus, Bertrand Dubus, and Regis Bossut (IEMN, Dept. ISEN, UMR CNRS 9929, 90 rue de Solferino, 59046 Lille Cedex, France)

Lead magnesium niobate ceramics (PMN) are promising materials for applications in the field of underwater acoustic projectors. A finite element procedure has been developed in the ATILA code to model the static deformation of these materials [J. C. Debus *et al.*, J. Acoust. Soc. Am. **100**, 2584(A) (1996)]. Two different elements are available according to whether or not the saturation of polarization is included. An extension of this model to nonlinear transient analysis is presented in this paper. The procedure is derived for time stepping by finite difference, using a central difference scheme. This new capability is demonstrated by analyzing the transient response of an electrostrictive bar submitted to different electrical excitations (voltage step, charge step, continuous sine). The validation of the model is carried out by comparing the results obtained with the analytical results for a lumped constants model. The validity of commonly used transducer characteristics is discussed to describe electrostrictive transducers. [Work supported by ONR contract and DRET/DCN grant.]

2:50

4pEA4. Finite-element study and experimental verification on irregular composite transducer design. Wenkang Qi and Wenwu Cao (Mater. Res. Lab., Penn State Univ., University Park, PA 16802, wkq@sun01.mrl.psu.edu)

In this paper 2-2 composite transducers with both regular and irregular kerfs have been studied by using FEM and experimental methods. Undesirable lateral modes are shown in the electrical impedance curves of periodic 2-2 composites. These lateral modes are well suppressed in a 2-2 composite with irregular kerfs when the ceramic volume percentage is low. However, the beam pattern shows some degree of distortion. In order to restore the beam pattern, mirror symmetry is introduced into a 2-2 composite with irregular kerfs, and the beam pattern of these 2-2 composites has been experimentally measured and calculated using FEM. Excellent agreement has been achieved. As expected, when the system is more disordered, the lateral modes are well suppressed, but the beam pattern is poorly focused. On the other hand, when the system is more ordered, the lateral modes are poorly suppressed, but the beam pattern is more focused. A trade-off is a sequence of 3; in this case the lateral modes can be effectively suppressed, and the beam pattern is also well focused.

3:05

4pEA5. Finite-element and experimental study of ultrasonic beam pattern characterization. Mark Draheim and Wenwu Cao (Penn State Univ., University Park, PA 16802, mrd145@psu.edu)

In this paper a study on the characterization of the acoustic beam pattern for two PZT-5H, 3.5-MHz transducers using both experimental and FEA methods is reported. To obtain actual pressure-field measurements, a three-axis positioning system was constructed. The transducers were submerged in an acoustic test tank and the positioning system was used to maneuver a 200-micron PVDF hydrophone within the pressure field. At each location the output voltage of the hydrophone was measured along with the corresponding location. Using SPYGLASS[®], these data were plotted to yield a two-dimensional "slice" of the actual pressure field. A finite-element model of the water-loaded ultrasonic transducer was created using ANSYS[®]. Material properties that were needed to model the piezo-

ceramic and polymers properly in the transducer were measured via pulse-echo technique. To validate the FEM results the electrical impedance of the transducers was calculated and compared to the actual measured values. The frequency-dependence of the impedance curves match closely. Using harmonic analysis, the pressure field in the water generated by the transducer was calculated and compared to the measured pressure-field data. Excellent agreement was obtained.

3:20-3:30 Break

3:30

4pEA6. Use of automatic modeler and small array receiver for acoustic source location. Xiaorong Lu, Wolfgang Sachse (Theoret. and Appl. Mech., Cornell Univ., Ithaca, NY 14853, xiaorong@msc.cornell.edu), and Igor Grabec (Univ. of Ljubljana, Slovenia)

A statistical, nonparametric regression analysis utilizing a conditional average and a smoothed empirical probability distribution forms the basis of an automatic modeler that resembles the operation of a neural-like network. Coupled with a small receiver array, the modeler can determine both the distance and direction to a source of acoustic emission. The use of such a small receiver array to locate sources of emission on a thick plate is demonstrated. The characteristic property of such a system is its capability for predicting the locations of various sources in a structure from the emitted waveforms. Using the modeler involves two steps. First, the modeler is trained with the computed Green's functions for the structure. In the examples shown, these correspond to a normal force and the corresponding displacement signal for a broad range of source/receiver separations. The database of these responses forms the basis of the modeler memory. After training, the modeler can be used to locate sources of emission on the plate. The operation of the system is demonstrated with both synthetic as well as measured waveforms corresponding to impact and step-loading forces on a thick plate. [Work supported by AFOSR.]

3:45

4pEA7. Universal equivalent circuit for transducers driven by a plate-shaped actuator. Jean C. Piquette (Naval Undersea Warfare Ctr., Code 21601, Bldg. 1171, Third Fl., 1176 Howell St., Newport, RI 02841-1708)

In recent years interest in the use of electrostrictive materials in transducers has increased greatly, in view of the development of materials such as lead magnesium niobate (PMN) and lead magnesium niobate/lead titanate (PMN/PT). These materials produce approximately an order of magnitude greater strain than that produced by piezoelectric materials when subjected to a comparable electric field level. Unfortunately, the behavior of these materials cannot be linearized by the usual procedure, since they will not retain a significant remanent pole for more than a short time. Hence, it is necessary to develop methods for predicting transducer performance based directly on the electrostrictive properties of these materials. This paper presents an equivalent circuit useful for predicting the first-order behavior of a transducer incorporating an electrostrictive material in the form of a plate-shaped actuator. The methods used are sufficiently general that the resulting circuit is also applicable to piezoelectric and electrostatic transducers. Thus, in this sense, the equivalent circuit presented is "universal," since it is equally applicable to electrostrictive, piezoelectric, and electrostatic transducers which utilize a plate-shaped actuator.

4:00

4pEA8. Sensitivity of a spherically focused transducer to focal length and aperture. Martin Manley (261 Congressional Ln., Rockville, MD 20852)

A method is presented to calculate the impulse response of a spherically focused transducer which generates a signal and then receives the reflection of the signal from a rigid surface. Results of calculations are shown in order to illustrate the effects of varying the focal length and the aperture.

These results suggest a method for empirical determination of effective values of the governing parameters of the transducer impulse response.

4:15

4pEA9. Comb transducer design for optimal guided wave performance. Semyen P. Pelts, Joseph L. Rose, and Michael J. Quarry (Dept. of Eng. Sci. and Mech., Penn State Univ., 114 Hallowell Bldg., University Park, PA 16802, j1r@psuvm.psu.edu)

Benefits of a comb transducer for producing guided waves in a structure include an ability to produce specific guided waves of choice in a low modulus material as well as good flexibility in mode choice and generation. As a result, excellent sensitivity and penetration power can be obtained by optimal mode choice. The comb transducer design for effective excitation of guided waves in an isotropic elastic plate is therefore considered in this study. The design involves optimizing geometrical and mechanical parameters of the transducer such as number of elements, element width, comb sizing, and frequency bandwidth. The mathematical model is necessary to provide framework for this design. The impulse excitation in an elastic plate is studied by applying a double Fourier transform in the wave-number and frequency domains. The method of residues

and numerical integration are used to evaluate the solution of the boundary value problem. The solution demonstrates how various parameters affect the transmission of energy in the elastic plate. Experimental results are obtained on the basis of this model. [Work supported by the Office of Naval Research.]

4:30

4pEA10. Composite transducer with spring's structure. B. L. Jiao and J. D. Zhang (Dept. of Electron., Peking Univ., 100871 Peking, PROC, xywang@radioms.radio.pku.edu.cn)

The objective of this paper is to introduce spring's structure to a piezolaminated beam for transducer application. The stacking sequence of the plies is along the thickness and the elastoelectric conversion is through the bending modes, in which some plies are compressed while the others are stretched. The piezoelectrical equations are given under the approximation for small deflection. As an application to the receivers, the leaf and spiral springs' structures are adapted to the numerical analysis. The acoustic impedance defined by the values in the mean is found much lower than that of the conventional transducers, while the sensitivities are significantly heightened.

THURSDAY AFTERNOON, 19 JUNE 1997

ROOM G, 1:25 TO 3:25 P.M.

Session 4pMU

Musical Acoustics: Music Cognition; General

Judith C. Brown, Chair

Department of Physics, Wellesley College, Wellesley, Massachusetts 02181

Contributed Papers

1:25

4pMU1. Sensitivity to key movement in a Beethoven excerpt: Theory and evidence. Lola L. Cuddy (Dept. of Psych., Queen's Univ., Kingston, Canada) and Nicholas A. Smith (Univ. of Toronto, Toronto, Canada)

After a brief introduction to historical and contemporary descriptions of musical key, key relationships, and key movement, the results of two experiments testing convergence of music theory with listener response will be reported. Perceptual sensitivity to key and key movement in the opening measures of the second movement of Beethoven's piano sonata Opus 53 (Waldstein) was evaluated by the probe-tone technique. Listeners were either familiar (experiment 1, $n=14$) or unfamiliar (experiment 2, $n=14$) with the excerpt. The excerpt was performed by an experienced pianist and recorded as MIDI files. On each trial, the excerpt was presented from the beginning to one of nine successive time points. Time points were the first beat, the first bar, and each successive bar. For each time point there were 12 presentations, each followed by a probe tone, and 1 of the 12 chromatic divisions of the octave randomly selected without replacement. Listeners judged the goodness-of-fit of each probe tone to the music presented; the set of 12 ratings yielded a probe-tone profile. A significant proportion of profile variance was accounted for by quantified music-theoretic and psychoacoustic predictors, the proportion increasing with familiarity. [Work supported by NSERC.]

1:40

4pMU2. Ability to detect changes in musical pieces is a function of musical training and musical context. Edward J. Crawley, Barbara E. Acker, and Richard E. Pastore (Ctr. for Cognit. and Psycholinguistic Sci., SUNY, Binghamton, NY 13902, br00437@binghamton.edu)

The ability of musicians and nonmusicians to detect a one-note change from a fixed three-voiced musical piece (the standard) was evaluated. Following previous studies, which had tested only musicians, several potentially important variables were manipulated. Musical context (polyphonic or homophonic) was manipulated between subjects. Melody location (high, middle, or low voice), and thus the standard, was varied for each subject across sessions. Within a session, changes varied in harmonic relatedness (related or unrelated), and voice location (high, middle, or low). Subjects performed a same-different task and indicated their confidence as to whether or not a change had been present. Furthermore, a signal detection analysis of the subjects' pattern of responses was performed to directly measure sensitivity to the presence of changes. As expected, musicians were more confident in their responses and revealed a higher sensitivity to changes than nonmusicians. Interestingly, the effects of musical training on sensitivity to changes did not interact with any variable except musical context. The results will be discussed in terms of general and music-specific aspects of perception and training. [Work supported by AFOSR.]

4pMU3. Effects of timbre manipulations on melody perception.

Barbara E. Acker and Richard E. Pastore (Ctr. for Cognit. and Psycholinguistic Sci., SUNY, Binghamton, NY 13902, bb31074@binghamton.edu)

The effects of timbre on melody perception were explored in three-voice homophonic (*H*) and polyphonic (*P*) pieces. Musically trained subjects learned a 12-note melody in isolation, then listened to *H* and *P* pieces that contained the melody in either the high, middle, or low spectral positions. Three timbre conditions were created: (1) homogeneous, where all three voices were in the same timbre; (2) enhanced, where the melody was presented in a timbre distinct from the other two voices; and (3) inconsistent, where each of the three timbres formed distinct patterns which were independent of the melody (*H* pieces only). In a discrimination task, subjects responded to errors only in the melody, although errors could occur in the other two voices. Several patterns of results occurred which were dependent on the type of timbre manipulation and the musical context employed. Thus the results demonstrate important aspects of how principles from music cognition (melody memory, timbre manipulations) and music theory (*H* and *P* pieces) can interact. [Work supported by AFOSR.]

2:10

4pMU4. The role of timbre, pitch, and loudness changes in determining perceived metrical structure.

Punita Singh (Sound Sense Consultancy Services, 20-A Aurangzeb Rd., New Delhi 110011, India)

The role of differences in pitch, loudness, and timbre as determinants of metrical structures was investigated. A repeated sequence of 12 pure tones was systematically partitioned into 4 groups of 3 tones or 3 groups of 4 tones by introducing changes in *F*₀, intensity, and spectral complexity between sounds at serial positions 1, 4, 7, 10 or 1, 5, 9, respectively. Listeners used a rating scale to indicate if a triple, quadruple, or ambiguous meter was perceived. For changes in only one parameter at a time, perception of rhythmic structure followed the physical markers. When changes in more than one parameter were made concurrently, multiple cues for triple or quadruple meter were available. Coincident changes led to reinforcement of the rhythm demarcated by the points of change. Conflicting changes led to different outcomes: Timbre and pitch changes dominated over a loudness-based accent structure. Pitch versus timbre stimuli were rated as having ambiguous meter. A combination of any two parameters versus the third dominated in determining the metrical structure. The results raise questions about the role of timbre in rhythm perception in general and as a tool for marking metrical units in particular.

2:25

4pMU5. Perceptual effects of simplifying musical instrument sound time-frequency representations.

James W. Beauchamp (2136 Music Bldg., Univ. of Illinois, 1114 W. Nevada, Urbana, IL 61801), Stephen McAdams, and Suzanna Meneguzzi (Univ. Rene Descartes, F-75006 Paris, France and Inst. de Recherche et de Coordination Acoustique/Musique (IRCAM), F-75004 Paris, France)

The perceptual salience of several outstanding features of musical instrument quasi-harmonic time-variant spectra were investigated. These are: amplitude spectrum shape variation, harmonic amplitude (or frequency) micro-variations, spectral envelope irregularity, harmonic frequency incoherence, and frequency inharmonicity. Seven musical instrument sounds (clarinet, flute, oboe, trumpet, violin, harpsichord, and marimba) were spectrum-analyzed to produce time-varying harmonic amplitude and frequency data. Tones resynthesized from these data were equalized in pitch, loudness, and duration. Six basic data simplifications were applied together with five combinations of them: amplitude versus time smoothing, spectrum shape fixing, spectral envelope smoothing, harmonic frequency coherence, frequency versus time smoothing, and harmonic frequency fixing. Twenty subjects participated in a discrimination study to determine the ease of distinguishing sounds synthesized from simplified data from

sounds resynthesized from the full data. Averaged over the seven instruments, the mean discrimination scores were: spectral envelope smoothing, 96%; spectrum shape fixing, 91%; harmonic frequency fixing, 71%; frequency smoothing, 70%; frequency coherence, 69%; and amplitude smoothing, 66%. In conclusion, spectral envelope smoothing had a profound effect (discrimination >95%) for all instruments except the trumpet. Spectrum shape fixing, which eliminates any spectral centroid variation, had a very large effect except for the clarinet and the oboe.

2:40

4pMU6. Cluster-based probability model for musical instrument identification.

Judith C. Brown (Phys. Dept., Wellesley College, Wellesley, MA 02181 and Media Lab., MIT, Cambridge, MA 02139, brown@media.mit.edu)

Mel based cepstral coefficients have been calculated for 30 short (1–2 s) segments of oboe sounds and 30 short saxophone sounds. These were used as features in a pattern analysis to determine for each of these sounds comprising the test set whether it belongs to the oboe or to the sax class. The training set consisted of longer sound segments of 1 min or more for each of the instruments. A *k* means algorithm was used to calculate clusters for the training data, and Gaussian probability density functions were formed from the mean and variance of each of the clusters. Each member of the test set was then analyzed to determine the probability that it belonged to each of the two classes, and a Bayes decision rule was invoked to assign it to one of the classes. Initial results classifying the oboe sounds have been very good, but are less impressive for the saxophone set. Results of a human perception experiment identifying these same sound segments will be reported.

2:55

4pMU7. Use of note partials' energy to improve the identification of polyphonic piano signals.

Lucile Rossi and Gerard Girolami (Univ. of Corsica, Quartier Grosseto, 20250 Corte, France)

The identification of the fundamental frequency of musical sounds is a difficult task which has been studied for more than 20 years. Methods using spectral approaches are used to identify polyphonic sounds. A novel approach for identifying monophonic and polyphonic piano sound signals has been developed by the authors [L. Rossi and G. Girolami, *J. Acoust. Soc. Am.* **100**, 2842(A) (1996)]. The identification is carried out by comparing the incoming spectrum with a database (automatically built previously) containing the frequential distribution of the partials of each piano note. The rate of correct identification is in the 90% range for polyphonic sounds. One kind of error is due to the artificial construction of a note by the combination of partials belonging to other notes; another one results from the difficulty of separating the case of one note played a long time with the case of this same note played two successive times with an overlap. The aim of this paper is to show how the evolution of the partials' energy associated with the search for information about the time of the note attacks can be used to reduce the false identification.

3:10

4pMU8. Vocal accuracy in performance of well-known melodies.

Clare L. Henderson (Dept. of Arts and Lang., School of Education, Private Bag 3105, Univ. of Waikato, Hamilton, New Zealand, clhend@waikato.ac.nz) and Lloyd Smith (Univ. of Waikato, Hamilton, New Zealand)

Vocalization is an important aspect of music making, regardless of the culture. Psychologists have carried out considerable research on human melody recognition and perception, which includes analysis of musical cognitive processes. However, questions that relate directly to the ways in which people perform what they recall have been left largely unanswered. In this experiment, a number of people were asked to sing, from memory,

ten well-known songs. Their performances were recorded on audio tape and analyzed and compared with notated versions of the songs. The results showed that pitch maintenance was more accurate when songs moved by steps rather than leaps and when intervals moved up rather than down. Pitch problems were more prevalent in songs using downward leaps or

changes of tonality. Wide leaps, such as fifths or sixths, were often compressed, with subjects singing the top note of an ascending interval flat, or the bottom of a descending interval sharp. The results of this experiment have useful applications for musicologists, music educators, composers, and those involved with data retrieval processes.

THURSDAY AFTERNOON, 19 JUNE 1997

ROOM N, 2:00 TO 3:30 P.M.

Session 4pPA

Physical Acoustics: Ultrasonics

L. C. Krysac, Chair

Department of Physics, Pennsylvania State University, 104 Davey Laboratory, University Park, Pennsylvania 16802

Contributed Papers

2:00

4pPA1. Experimental studies of diffuse decay curvature. John Burkhardt (Dept. of Eng., Purdue Univ., Fort Wayne, IN 46805-1499)

Reverberant, ultrasonic decay experiments were performed on an irregularly shaped aluminum block on which varying sized water drops were placed. The resulting nonproportionally damped systems were found to possess nonexponential decays in agreement with theoretically derived results. These findings support the author's proposition that the features of nonexponential decays are a possible quantitative nondestructive technique for the characterization of localized fatigue damage in metals.

2:15

4pPA2. Measuring the frequency-dependent attenuation in highly lossy material using spread spectrum ultrasound. Grant A. Gordon (Appl. Res. Lab., Penn State Univ., State College, PA 16804)

Broadband signals are commonly used in ultrasonic spectroscopy to measure the frequency-dependent attenuation characteristics of lossy solid media. As compared with narrow-band signals, broadband signals are preferred since they do not require tedious frequency scanning and extensive data reduction efforts. By using spread spectrum techniques to design signals with given bandwidth and temporal characteristics, the overall accuracy and resolution of ultrasonic spectroscopy can be improved. An expression for ultrasonic attenuation is developed and compared with the traditional analysis approach to illustrate the improved accuracy of the new technique. Experimental results obtained using a direct sequence modulation system will also be presented for the attenuation spectra of various lossy solids.

2:30

4pPA3. Temperature-dependent ultrasonic properties of aluminosilicates glass ceramics. J. H. So (Dept. of Phys. and Astronomy, CMSS Prog., Ohio Univ., Athens, OH 45701, jhs@helios.phy.ohiou.edu), D. H. Green, and S. S. Yun (Ohio Univ., Athens, OH 45701)

The longitudinal wave velocity and attenuation and shear wave velocity for machinable aluminosilicate glass-ceramic (MACOR) samples were measured as functions of frequency and temperature with the pulse-echo method. Frequencies of 5–30 MHz were used in the temperature range of 100–300 K. The velocity differences between 100 and 300 K were about 1% for longitudinal waves and 3% for shear waves. The monotonic decrease in the longitudinal attenuation coefficient between 100 and 300 K was approximately 50%. The longitudinal wave attenuation coefficient increases linearly with frequency and nonlinearly with decreasing temperature. This contrasts with the longitudinal and shear wave velocities, both of

which decrease with increasing temperature. These data were used to calculate the elastic moduli, Poisson's ratio, and Lamé parameters for the material. These results are interpreted in terms of thermal stress cracking and microstructure-related absorption.

2:45

4pPA4. Acoustic spectroscopy of deep levels in Cr-doped GaAs using light beam generated SAW. Peter Bury, Ivan Baják, and Karol Grondžák (Dept. of Phys., Žilina Univ., 010 26 Žilina, Slovakia, bury@fel.utc.sk)

A light-beam-generated space-charge inhomogeneity was used for the first time as a probing tool to study the behavior of deep levels in high-resistivity, Cr-doped GaAs using acoustic transient spectroscopy. The space-charge inhomogeneity produced by the illumination of photoresistive piezoelectric semiconductors in a high-frequency electric field can generate, in proper conditions, both the surface and transversal or longitudinal acoustic wave. The temperature dependence of the time development of such a generated surface acoustic wave (SAW) after illumination by pulsed monochromatic light was used to investigate deep levels in high-resistivity GaAs using the technique of acoustic deep-level transient spectroscopy (A-DLTS) [P. Bury and I. Jamnický, Proc. World Congress on Ultrasonics, Berlin, 1995, p. 535]. Deep-level parameters such as activation energy and capture cross-section were determined. [Work supported by a Grant of the Slovak Ministry of Education, No 1/1310/96.]

3:00

4pPA5. Ultrasonic study of vortex viscosity and phase transitions in $\text{YBa}_2\text{Cu}_3\text{O}_7$ single crystals. D. Dasgupta, C. Hucho,^{a)} J. R. Feller, B. K. Sarma, and M. Levy (Dept. of Phys., Univ. of Wisconsin, Milwaukee, WI 53211)

A novel ultrasonic technique (sampled cw method) is used to probe into the viscosity of vortex systems in untwinned and twinned YBCO single crystals. Motion of the flux lines in the superconducting sample is induced via coupling with the moving ionic lattice in the bulk of the sample. Hence, the observed dissipation has its origin in the (induced) motion of the vortices with respect to the ionic background of conduction charge carriers. In the experiments which were performed, pronounced field-dependent attenuation changes were observed which are indicative of transitions from a soft vortex system at low fields to a rigid vortex system at high fields. [Research supported by Office of Naval Research and C. Hucho was supported by Deutsche Forschungsgemeinschaft.] ^{a)}Currently at: FB Physik, Augsburg, Germany.

4pPA6. A novel method for characterizing contact-mode shear wave transducers in immersion. David K. Hsu, Brent A. Fischer (Center for NDE, Iowa State Univ., Ames, IA 50011, dhsu@cnde.iastate.edu), and Vinay Dayal (Iowa State Univ., Ames, IA 50011)

The dominating shear motion of a contact-mode shear wave transducer is often accompanied by a weak parasitic longitudinal mode due to its finite size and constrained boundary conditions. The amplitude and phase of this out-of-plane vibration may be exploited for determining the shear polarization direction and for detecting any asymmetry or distortion of the shear transducer's vibration pattern. In practice, the contact-mode shear

probe is immersed in water and energized by an rf pulser to serve as the transmitter. A C-scan is then made of the face of the shear transducer using a focused longitudinal wave transducer as the receiver. The resulting image shows the out-of-plane longitudinal vibration of the shear transducer. For commercial contact-mode shear probes, this pattern usually consists of two crescent-shaped regions of higher vibration amplitude but of opposite phase. A line bisecting the two crescents and passing through their centers defines the shear polarization direction. Finite-element modeling of the shear displacement of a circular disk with constrained boundary confirmed the experimentally observed behavior. This approach provides a simple method for characterizing commercial shear wave transducers. [Work supported by Center for NDE, Iowa State University.]

THURSDAY AFTERNOON, 19 JUNE 1997

DEANS HALL I, 12:45 TO 3:30 P.M.

Session 4pPPa

Psychological and Physiological Acoustics: Pitch and Loudness

Mary Florentine, Chair

Department of Speech-Language Pathology and Audiology, Northeastern University (133FR), 360 Huntington Avenue, Boston, Massachusetts 02115

Contributed Papers

12:45

4pPPa1. The effect of temporal fringes on the fundamental-frequency discrimination of resolved and unresolved harmonic complexes. Christophe Micheyl and Robert P. Carlyon (MRC-APU, 15 Chaucer Rd., Cambridge CB2 2EF, UK, christophe.micheyl@mrc-apu.cam.ac.uk)

Three experiments investigated the finding that F_0 discrimination for a harmonic complex can be impaired by temporal fringes [R. P. Carlyon, J. Acoust. Soc. Am. **99**, 525–533 (1996)]. The 100-ms target and the 200-ms forward and backward fringes were filtered (1375–1875 Hz) and presented in a pink noise background. Two F_0 's were used: 88 Hz, where the harmonics were unresolved by the peripheral auditory system, and 250 Hz, where the harmonics were resolved. The fringe produced a larger deterioration when it had a similar F_0 to the target than when the F_0 's differed. Experiment 2 showed that adding off-frequency components to the fringe did not affect this pattern of results, thereby ruling out interpretations based on spectral confusion between the target and the fringe. Experiment 3 further examined the dependence of the deterioration on the similarity between target and fringe F_0 's. It showed that FDLs for an 88-Hz target were not reduced when the components of an 88-Hz fringe were set to alternating phase, thereby doubling their pitch without affecting resolvability of the components. The implications for single- and dual-mechanism models of pitch will be discussed.

1:00

4pPPa2. Auditory pitch influenced dramatically by dynamic intensity change. Michael K. McBeath (Dept. of Psych., Kent State Univ., Kent, OH 44242-0001) and John G. Neuhoff (Lafayette College, Easton, PA 18042)

Historically, auditory pitch is considered to be principally a function of acoustic frequency with only a small effect due to absolute intensity. Yet, when tones are Doppler shifted, the pitch dramatically rises and falls with dynamic intensity. This study uses a matching procedure to document the magnitude of pitch drop of dynamic Doppler stimuli. Listeners heard Doppler-shifted tones with a mean frequency of either 1046 or 175 Hz, a total fall of 2 semitones, and an intensity change of 58 to 86 dB and back to 58 dB. They compared this drop to a pair of 75-dB, 0.25-s discrete tones

that dropped in frequency by intervals ranging from 0 to 24 semitones. The average match between experienced sizes of Doppler and discrete pitch change occurred at a discrete drop of 8 semitones, four times larger than the actual Doppler frequency change. This effect opposes and is an order of magnitude larger than the well known effect due to discrete intensity change. It is proposed that the interaction between dynamic pitch and loudness reflects a natural correlation between changes in frequency and intensity that is neurally encoded to facilitate processing of meaningful acoustic patterns.

1:15

4pPPa3. The interaction of pitch and loudness in dynamic stimuli: Beyond the Doppler illusion. John G. Neuhoff (Dept. of Psych., Lafayette College, Easton, PA 18042) and Michael K. McBeath (Kent State Univ., Kent, OH 44242)

Listeners tend to experience rising pitch even though frequency falls as a sound source approaches. The phenomenon, called the Doppler illusion [J. G. Neuhoff and M. K. McBeath, J. Exp. Psychol. Hum. Percept. Perform. **22**, 970–985 (1996)], shows that dynamic intensity change influences perceived pitch in a way that is qualitatively different from discrete static intensity change. Two new studies show a dynamic influence of intensity change on perceived pitch and a dynamic influence of frequency change on perceived loudness. Listeners were presented with square wave tones of either rising, falling, or constant intensity that either rose, fell, or remained constant in frequency for 6 s. Listeners responded in real time to either changes in pitch or loudness by moving a response wheel. It was found that dynamic intensity sweeps contribute to the perception of dynamic pitch change in the direction of the intensity sweep. In addition, dynamic frequency sweeps contribute to the perception of perceived loudness change in the direction of the frequency sweep. The results imply that pitch and loudness perception interact under dynamic conditions in a way that cannot be predicted by perceptual models derived from the presentation of discrete static tones.

4pPPa4. Discrimination thresholds for glides in frequency as a function of transition span, duration, and direction. John P. Madden (Dept. of Commun. Disord., Univ. of North Dakota, Grand Forks, ND 58202, madden@badlands.nodak.edu)

The study measured discrimination thresholds for glides in frequency as a function of magnitude of transition span, duration, and direction. The purpose was to determine whether the Weber fraction is constant across a range of transition spans, as suggested by an earlier study [J. P. Madden and K. M. Fire, *J. Acoust. Soc. Am.* **100**, 3754–3760 (1996)]. Subjects were asked to distinguish between a comparison glide with a fixed transition span and a target glide with a greater transition span. Comparison signal frequency transition spans were 0.5, 1, 2, 4, and 8 times the ERB of the auditory filter at the signal center frequency. The nominal center frequency of the glides was 1.8 kHz, but the actual signals were roved over a range of center frequencies. Both up- and down-glides were used; signal durations were 50 ms and 400 ms. The results indicate that the Weber fraction is essentially constant for transition spans of 1–8 ERBs, but increases below 1 ERB. The effects of duration and direction were small. [Work supported by Research Grant No. 1 R15 DC 02662-01 from the National Institute on Deafness and Other Communication Disorders, National Institutes of Health.]

4pPPa5. Temporal integration of loudness for tones under partial masking. Mary Florentine (Commun. Res. Lab., Dept. of Speech-Lang. Pathol. and Audiol., 133 FR, Northeastern Univ., Boston, MA 02115), Søren Buus, and Monica Robinson (Northeastern Univ., Boston, MA 02115)

Temporal integration of loudness for 1-kHz tones presented in quiet and under white-noise masking was measured for 5- and 200-ms tones using an adaptive 2I,2AFC procedure. Levels ranged from 5 to 90 dB SL. Results for six listeners with normal hearing show that the amount of temporal integration, defined as the level difference between equally loud 5- and 200-ms stimuli, varies nonmonotonically with level and is greatest at moderate levels. The average amount of temporal integration is about 15 dB near threshold, increases to a peak of 27 dB when the 5-ms tone is about 53 dB SPL, and decreases to about 15 dB near 100 dB SPL. For masker levels of 40, 60, and 80 dB SPL, the amount of temporal integration near masked threshold remains about 15 dB. The maximum amount of temporal integration decreases as masker level increases and occurs at progressively higher levels. At high levels, the amount of temporal integration is nearly the same as in quiet for all masker levels. These results are consistent with the loudness function being steeper near masked threshold than at the same SPLs in quiet. [Work supported by NIH-NIDCD R01DC02241.]

4pPPa6. Loudness summation at low levels and the form of the loudness function near threshold. Søren Buus (Commun. and Digital Signal Processing Ctr., Dept. of Elec. and Comput. Eng., 409 DA, Northeastern Univ., Boston, MA 02115), Mary Florentine, and Hannes Müsch (Northeastern Univ., Boston, MA 02115)

This study uses measurements of loudness summation to examine the form of the loudness function for tones near threshold. An adaptive, two-interval, two-alternative, forced-choice, roving-level procedure was used to obtain loudness-balance measurements between a 1600-Hz tone and tone complexes centered at 1600 Hz. The complexes consisted of equal-SL components with levels between about -3 and 20 dB SL. Frequency separations were one, two, four, or six critical bands for four-tone complexes and one or two critical bands for ten-tone complexes. Results for six young listeners with normal hearing show that the SL difference between the pure tone and the components of the equally loud tone complex near threshold averages about 4 dB for the four-tone complexes and about 7 dB for the ten-tone complexes. When the component level of the four-tone complexes with wide frequency separations is around 20 dB SL, the

equally loud pure tone is near 47 dB SL. These results indicate that loudness may grow approximately as $I^{1.5}$ (where I is tone intensity) near threshold and as $I^{0.2}$ at moderate levels. [Work supported by NIH-NIDCD R01DC02241.]

4pPPa7. Loudness recalibration as a function of recalibration and comparison tone level. Dan Mapes-Riordan and William A. Yost (Parmly Hearing Inst., Loyola Univ. of Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626, dmapes@luc.edu)

Presenting loud tones at one frequency and quiet tones at a different frequency makes the quiet tones appear relatively louder. This phenomenon, dubbed loudness recalibration [Marks, *J. Exp. Psychol.* **20**, 382–396 (1994)], was studied using an adaptive, two-track loudness comparison procedure. In this study, a baseline loudness comparison was initially established between two tones. Immediately following this baseline sequence, a sequence of trials were given in which the two comparison tones were preceded by a recalibration tone. The amount of steady-state loudness recalibration was measured as a function of the recalibration tone level and the baseline comparison tone level. The results showed that loudness recalibration is present when the recalibration tone level is much larger than the comparison tone level and that no loudness recalibration is generated when the recalibration tone level is less than or equal to the comparison tone level. In addition, it was found that a recalibration tone did not affect the threshold level of detection. These results support those of Marks [J. Acoust. Soc. Am. **100**, 473–480 (1996)] indicating that loudness recalibration is a centrally based, fatiguelike phenomenon. [Work supported by a Program Project Grant from NIDCD.]

4pPPa8. Magnitude estimation scaling of the loudness component of pure tones, narrow band noise, broadband noise, rock music, and babble speech. Donald Fucci (School of Hearing and Speech Sci., Ohio Univ., Lindley Hall 219, Athens, OH 45701, dfucci1@ohiou.edu), Linda Petrosino (Bowling Green State Univ., Bowling Green, OH 43403), Denise Wyatt, and Corry Wilcox (Ohio Univ., Athens, OH 45701)

This study was designed to extend the range of auditory stimuli used to study magnitude estimation scaling of loudness. The five stimuli chosen were: a 1000-Hz pure tone; narrow band noise (700–1300 Hz bandwidth); broadband noise (100–10 000 Hz bandwidth); rock music [Led Zeppelin, Atlantic Recording Corp., CD Recording No. 19127-2 (1969)]; and babble speech [D. Kalikow, K. Stevens, and L. Elliott, *J. Acoust. Soc. Am.* **61**, 1337–1351 (1977)]. Subjects were 30 normal young adult women. During the auditory magnitude estimation task for each of the five stimuli, a subject was instructed to assign numbers to that stimulus which was presented in a randomly ordered series of nine sensation levels (10, 20, 30, 40, 50, 60, 70, 80, and 90 dB SL). An analysis of variance repeated measured design showed no difference in the numerical responses of the subjects for the five different stimuli. Results suggest the presence of an underlying stabilizing factor (internal scaling mechanism) that allows adult subjects to consistently scale loudness irrespective of the type of auditory stimulus presented [J. Zwislöcki and D. Goodman, *Percept. Psychophys.* **28**, 28–38 (1980)].

4pPPa9. Effect of age on the comfortable loudness range in women. Vishakha W. Rawool (Commun. Disord. and Special Education, Bloomsburg Univ., Bloomsburg, PA 17815) and Joseph P. Pillion (Kennedy Krieger Inst., Baltimore, MD 21205)

This study is designed to evaluate the effect of age on the range of sound pressure levels perceived as “comfortable.” Ten young (22–26 years) and eight older (42–50 years) female subjects participated in the study. All the subjects had auditory sensitivity within 20 dB HL across 0.5 to 4 kHz. Subjects were asked to judge the loudness of warble tones in the following categories: not audible, very soft, soft, comfortable, loud and very loud. The loudness judgments were made at 0.5, 1, 2, and 4 kHz. For any loudness judgement to be considered valid, the same loudness judgement had to be assigned by the subject at least twice to the same sound

pressure level. The comfortable loudness range was calculated for each subject by subtracting the highest sound pressure levels judged as "soft" from the lowest sound pressure levels judged as "loud." The mean "range of comfortable loudness" values for the older female subjects is lower than that obtained from the younger subjects. Additional data are being collected. Following completion of data collection, the data will be submitted to the mixed multivariate analyses of variance. The results and possible implications of the findings will be discussed. [Work supported by Research and Disciplinary grant from Bloomsburg University.]

3:00

4pPPa10. Puzzles of loudness constancy, loudness adaptation, and speech frequencies. Ernest M. Weiler, David E. Sandman, and Hongwei Dou (ML #379, Hearing Lab., CSD, Univ. of Cincinnati, Cincinnati, OH 45221)

Loudness constancy is suggested as a term for the failure to observe simple adaptation above 30 dB, noting however that Miskiewicz *et al.* [J. Acoust. Soc. Am. **94**, 1281–1286 (1993)] indicated that simple adaptation is found at higher intensities above 4 kHz. Conversely, using the ipsilateral comparison paradigm (ICP), Dange *et al.* [J. Gen. Psychol. **120**, 217–243 (1993)] found adaptation at 1000 Hz, from 45 to 75 dB. They saw this as revealing underlying neural adaptation, normally concealed by processes of loudness constancy. As a further test, Hellman [personal communication (1996)] urged that ICP adaptation be attempted without the usual designated modulus. In the present study ($N=20$), strong ICP loudness adaptation was indeed found with and without the designated modulus. Recently, Janson *et al.* [Br. J. Audiol. **29**, 288–297 (1996)] found that ICP

adaptation at 1 kHz was much less consistent for subjects with higher frequency loss than for nonimpaired listeners. Discussion of the possible value of loudness constancy in the speech frequencies will be discussed in light of current data and previous studies.

3:15

4pPPa11. The theory of mental testing and the correlation between physical noise level and annoyance. Karl Th. Kalveram (Dept. of Cybernetical Psych., Univ. of Duesseldorf, 40225 Duesseldorf, Germany)

Applying the "theory of psychological testing," physical noise measurement procedures are regarded as "tests" variables and the related annoyance as the criterion variable. The concepts of "reliability," "validity" and "equivalence" then can be used to assess the precision of different noise measurements. Referring to data of an investigation on aircraft noise in the vicinity of Munich Airport in 1969, in which different physical as well as annoyance data were sampled from the same subjects, it turns out, that the measurements called L_{eq1} , L_s , L_{eq3} , L_{eq4} , NNI and FB1 meet the equivalence criterions of test theory, that is, they cannot statistically be distinguished, and their coefficients of reliability are close to one. Other measurements like D10, H81, $\log N$ and L_{eq10} , equivalent to the former six. However, even the former six are only of moderate validity (about 0.5). Regarding psychological testing, therefore, physical noise is only a poor measure when used to predict individual annoyance, but it suffices when used to predict group averages. Moreover, it can be concluded from the theory that the attempt to enhance validity by modification of the highly reliable physical measuring procedures cannot be successful. It would be more effective to enlarge the reliability of the psychological measurement procedures of annoyance.

THURSDAY AFTERNOON, 19 JUNE 1997

DEANS HALL I, 3:35 TO 5:25 P.M.

Session 4pPPb

Psychological and Physiological Acoustics, Musical Acoustics and Noise: W. Dixon Ward Memorial Session

Christopher W. Turner, Chair

University of Iowa, Wendell Johnson Speech and Hearing Center, Iowa City, Iowa 52242-1012

Chair's Introduction—3:35

Invited Papers

3:40

4pPPb1. W. Dixon Ward—Professor, mentor, university citizen. Arlene Earley Carney (Dept. of Commun. Disord., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, carne005@gold.tc.umn.edu)

W. Dixon Ward had a colorful and active presence in the scientific community and in the Acoustical Society of America as an officer, editor, presenter, and author. Many of his ideas and much of his written work sparked debate and discussion. On the homefront at the University of Minnesota, he had an equally colorful presence as a mentor to graduate students and as a frequent and controversial letter writer to the Minnesota Daily, the university newspaper. Selections of Dix's letters on a variety of topics from word usage to university and world affairs will be presented. In this introduction, Dix's professional life outside the Acoustical Society will be reviewed and discussed through the eyes of his students and colleagues.

3:55

4pPPb2. W. Dixon Ward—A personal retrospective. Ira J. Hirsh (Central Inst. for the Deaf, 818 S. Euclid Ave., St. Louis, MO 63110, ira@cidmv1.wustl.edu)

"Dix" Ward's professional career began about 50 years ago, when he was an undergraduate majoring in physics and practicing a life-long interest in music, both as a student and as pianist and singer. Encouraged by one of his college teachers, W. A. Rosenblith, he pursued graduate work at Harvard in the Psychoacoustic Laboratory, under S. S. Stevens. A dissertation on subjective musical pitch and experimental study of the recovery of hearing after acoustic stimulation set him on a double course which continued his musical interests (first with D. W. Martin) and the study of hearing damage following noise exposure (with H. Davis and A. Glorig). Those

interests were to continue as he reached his final and long-lasting appointments at the University of Minnesota. His scientific contributions have been empirical, not theoretical. He focused on finding out what is demonstrably true. His contributions as a scientific citizen were many—in the Acoustical Society, in national and international standards, in the International Society of Audiology, in CHABA, and in related government activities related to noise and noise damage.

4:10

4pPPb3. The idiotone and spontaneous otoacoustic emissions. Robert C. Bilger (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana—Champaign, 901 S. Sixth St., Champaign, IL 61820, r-bilger@uiuc.edu)

Although the preponderance of Dix Ward's work was in the area of the effects of noise on hearing, his paper on tonal monaural displacuisis [J. Acoust. Soc. Am. **27**, 365–372 (1955)] has always been my favorite from among all of his papers. In this paper, published several years after he had completed his work at Harvard and was being introduced to noise-induced hearing loss through the Central Institute for the Deaf, he performed a consummate study of his own tonal monaural displacuisis and that of a woman who was studying music at Harvard. Not only did he specify their displacuisis exhaustively, his conclusions from that study foretold of many future developments in psychoacoustics, hearing loss, tinnitus, and signal analysis. The import of this monumental work will be iterated.

4:25

4pPPb4. W. D. Ward as musical acoustician, journal editor, and reviewer. Daniel W. Martin (7349 Clough Pike, Cincinnati, OH 45244)

W. Dixon Ward's musical talent as an expert pianist and a singer, combined with experience and training in both psychology and physics, provided the background for his significant doctoral research on pitch at Harvard University; for subsequent research in musical acoustics at the Baldwin Piano and Organ Company; and for continuing research in musical sound throughout his career. Twice Dix served the *Journal* as an associate editor for psychological acoustics. He peer-reviewed many *Journal* articles in psychoacoustics, musical acoustics, and noise. He reviewed numerous books, and served on acoustical standards committees including acoustical terminology.

4:40

4pPPb5. Possible correction factors for intermittent noise exposure. D. L. Johnson (EG&G MSI, P.O. Box 9100, Albuquerque, NM 87119-9100, dljohnsn@rt66.com)

A standard on Occupational Hearing Loss, ANSI S3.44 (1996), has been recently published. One of its features, unlike ISO R-1999 upon which it is based, allows a trading rule other than one based solely on equal energy. To account for the fact that intermittent noise, as compared to steady noise of the same energy, may not be as harmful to hearing, equal energy with a correction factor is being investigated. ANSI Working Group S3-58 is evaluating at least seven different approaches. In order to evaluate the best approach for correcting for intermittency, W. Dixon Ward's temporary threshold shift (TTS) data, from a variety of intermittent noise exposures [W. D. Ward, *Effects of Noise on Hearing* (Raven, New York, 1976), pp. 407–420], will serve as one evaluation criterion. Actual hearing loss data from exposures to various intermittent noises will serve as the other evaluation criterion.

4:55

4pPPb6. Perception of musical pitch. Edward M. Burns (Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, pwa@u.washington.edu)

Although most of Dix Ward's publications were concerned with the effects of noise on hearing, his first love was music perception. The publication derived from his doctoral dissertation, "Subjective musical pitch" [J. Acoust. Soc. Am. **26**, 369–380 (1954)], is one of the seminal works in the area, as is his treatise on absolute pitch [Sound **2**(3), 14–21 (1963); **2**(4), 33–41 (1963)]. In these publications, he proved that the performance of certain musicians on pitch tasks, such as ratio production and absolute identification, was an order of magnitude more precise than predicted by the then-current psychophysical "laws," derived from research on "normal-hearing" subjects. Dix Ward's research in this area will be discussed, as will recent research which extends and complements his findings. Taken together, this body of work gives empirical evidence for something of which Dix himself was long aware: It is music perception which most closely utilizes the full capabilities of the auditory system; and it is music, rather than speech, which is in fact "special."

5:10

4pPPb7. Risk assessment for occupational noise induced hearing loss: W. Dixon Ward and the "Anti-Chicken Little Movement." William W. Clark (Central Inst. for the Deaf, 818 S. Euclid Ave., St. Louis, MO 63110)

One of the hallmarks of W. Dixon Ward's illustrious career was his proclivity to criticize effectively the efforts of his colleagues. Depending upon the topic and the victim, this ability led him to be characterized as an outstanding editor, a thoughtful reviewer, an overzealous critic, or an impediment to consensus. Despite the variety of responses, Dix's critiques shared one characteristic: They were almost always correct. One of his favorite targets was the international standard, "Acoustics—Determination of occupational noise exposure and estimation of noise-induced hearing impairment" (ISO 1999, 1990). Throughout the two decades of its development Dix took to task the framers of the standard, challenging the assumptions which underly it. Dix was particularly vexed by ISO's use of the 3-dB exchange rate, the presumption of 0% risk of noise induced hearing loss for those nonoccupationally exposed, and the estimation that normal 18-year-olds have perfect hearing. In this tribute, Dix's criticisms of the ISO standard will be articulated and supported with recent data, and his role as the leader of the "Anti-Chicken Little Movement" justified.

Session 4pSA

Structural Acoustics and Vibration: Acoustic Radiation from Structures

Joel M. Garrelick, Chair

Cambridge Acoustical Associates, Inc., 200 Boston Avenue, Medford, Massachusetts 02155

Contributed Papers

2:00

4pSA1. The forward and backward projection of harmonic pressure fields from complex three-dimensional bodies. Peter R. Stepanishen (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882-1197)

The forward and backward projection of pressure fields from complex three-dimensional harmonically vibrating bodies is an important problem in noise control. A new generalized internal source density (GISD) method is presented to address the projection problem. The GISD approach is based on decomposing the pressure field on a closed surface of revolution in the fluid into a summation of circumferential orders where the pressure field for each order is associated with an internal linear distribution of ring sources on the axis of revolution. The axial variation of each source distribution is formulated as the solution of a mean square error problem and the resulting distributions can then be simply used to determine the entire external and/or surface pressure and velocity fields of the vibrating body. Analytical and numerical results are presented for several examples to illustrate the basic approach for bodies with different aspect ratios. Exact expressions for the source strength distributions for a sphere are developed and shown to have a vanishingly small region of support about the center of the sphere. For cylindrically shaped bodies with large aspect ratios, the spatial characteristics of the source strength distributions more closely match the normal velocity distribution.

2:15

4pSA2. An analytical solution for a rectangular piston independently driven at two edges. John MacGillivray (Graduate Program in Acoust., Penn State Univ., State College, PA 16804)

Far-field radiation of a baffled rectangular piston is described using a frequency-domain Rayleigh integral solution. The piston model is driven at a single frequency at two opposing edges, where these individual edge sources are independent of one another in both amplitude and phase. Directivity plots of the analytical solution are used to graphically observe directional effects caused by changing the phase and amplitude of these edge sources. Interesting and unexpected beamforming characteristics are shown to be produced by certain amplitude and phase combinations. [Work supported by IBM through the Shared University Research program and the Applied Research Laboratory at Penn State Univ. University.]

2:30

4pSA3. A resistance probe based on the lumped parameter model for the acoustic power output of a vibrating structure. John B. Fahnline (Appl. Res. Lab., Penn State Univ., University Park, PA 16802), Gary H. Koopmann, and Weicheng Chen (Penn State Univ., University Park, PA 16802)

Previous papers by the authors addressed the analytical derivation and the numerical implementation of a lumped parameter model for the acoustic power output of a vibrating structure. Here the experimental measurement of the resistance is addressed. The acoustic resistance, which is related to the Green's function of the second kind, can be determined experimentally by: (1) constructing an acoustically hard physical model of

the boundary surface; (2) representing the discrete simple source with a small loudspeaker; and (3) measuring the acoustic field at a receiver location with a small microphone. The design of an acoustic resistance probe is described, along with an analysis of both the calibration procedure and the subsequent calculation of the resistance matrix. The analysis is validated through several example problems, with the accuracy of the measured acoustic resistance assessed by direct comparison to analytical or numerical solutions. For most of the cases considered, the experimental measurements and numerical predictions agree very well, but for some of the example problems discrepancies occur due to the limited dynamic range of the probe and acoustic excitation of structural vibrations.

2:45

4pSA4. Accurate determination of total sound power radiated from a rectangular panel. W. L. Li and H. J. Gibelung (United Technologies Carrier Corp., Carrier Pkwy., Syracuse, NY 13211)

Radiation of sound from planar vibrating panels has been extensively studied for years. In most of these investigations, the total sound power is approximately obtained from the modal radiation efficiencies without considering the contributions from modal cross couplings. The primary reason for doing this is perhaps that the inclusion of cross-coupling terms may result in a tremendous increase in computational load, particularly when a large number of structural modes is involved. In a recent paper [R. D. Snyder and N. Tanaka, *J. Acoust. Soc. Am.* **97**, 1702–1709 (1995)], an approximate estimate of the cross-coupling terms from the modal radiation efficiencies was discussed. It is, however, valid only in a low-frequency range and deteriorates rapidly as frequency increases. In this paper, the effect of modal cross couplings on the total sound power radiated from a simply supported rectangular panel is discussed. A formula is derived for an accurate and efficient calculation of the cross-coupling terms which allows for effortlessly including the effects of the modal cross couplings in the total sound power calculation. In addition, the present formulation is valid in the whole frequency range. Results are presented.

3:00

4pSA5. Acoustic radiation in light fluid of a vibrating beam with uncertain material properties. Jean-Sébastien S. Genot, François Charron, and Nouredine Atalla (Dept. of Mech. Eng., Univ. of Sherbrooke, 2500 boulevard de l'Université, Sherbrooke PQ J1K 2R1, Canada)

Engineers are aware of the need to account for uncertainties in their predictions in knowledge of the material and cross-section properties (e.g., Young's modulus, damping factor, section area). Furthermore, acousticians are often interested in the prediction of the maximum sound power that could be radiated by a given structure. But they face a numerical challenge due to the amount of input parameter combinations they must solve to compute the maximum response. In this context, a simple case of acoustic radiation in two dimensions—a simply supported beam in light fluid—is a good way to investigate the sensitivity of the acoustic power to the uncertainties of the input parameters. To study this case, an analytical method with a perturbation scheme up to the second order has been intensively used, and compared with a Monte Carlo method. In this presentation, it will be shown that the acoustic power is very sensitive to the

variations of the structural parameters in the neighborhood of the beam eigenfrequencies, and that second-order perturbation techniques are not sufficient. Consequently, a hybrid method is proposed to improve the results locally.

3:15–3:30 Break

3:30

4pSA6. Influence of partial coatings on the acoustic radiation from a fluid-loaded structure. Joseph M. Cuschieri (Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431, joe@jmc.oe.fau.edu) and David Feit (David Taylor Res. Ctr., Bethesda, MD 20084)

In general, an acoustic coating is applied uniformly on the surface of a fluid-loaded structure to minimize acoustic radiation and scattering. There are, however, some inherent advantages to optimize the distribution of the coating around areas from which the acoustic radiation appears to emanate. This would be analogous to the application of damping treatment in areas of a vibrating structure which have high vibration levels. In the case of the acoustic radiation the problem is more complex because of the coupling between the acoustic fluid and the structure. In this paper, the acoustic radiation from a partially coated fluid-loaded structure, of infinite extent and excited by a point or line force, is examined by developing an analytic/numeric solution. The solution follows along some of the procedures that the authors developed in previous work related to the scattering from fluid-loaded plates and shells. The coating is assumed to provide mainly a decoupling layer between the acoustic medium and the structure; that is, it does not add mass or stiffness to the base structure. The influence of added mass or stiffness of the coating can be included as an added inhomogeneity and treated separately in the solution. [Work sponsored by NSWCD.]

3:45

4pSA7. A novel acceleration method for the variational boundary element approach based on multipole expansions. Michel Tournour and Noureddine Atalla (Dept. de Genie Mech., Univ. de Sherbrooke, Sherbrooke, PQ J1H 2R1, Canada)

The acoustic radiation of general structures with Neumann's boundary condition using the variational boundary element method (VBEM) is considered. The classical numerical implementation of the VBEM suffers from the computation cost associated with double surface integrals. To circumvent this limitation a novel acceleration method is proposed. It is based on the expansion of the cross influence matrices in terms of multipoles using the expansion of the Green's function in terms of spherical Bessel functions. Since the resulting multipoles are not dependent on the elements locations, the technique results in large computation time savings for homogeneous meshes. The theory behind the approach, its convergence, and its numerical implementation in a VBEM code will be presented. It will be shown that by accounting for the monopole, dipole, and quadrupole terms in the multipoles expansion, the classical convergence criteria usually used for quadratic element hold. Numerical applications to acoustic radiation from plates, open cylinders, and closed boxes will be presented to demonstrate the accuracy and efficiency of the method.

4:00

4pSA8. A normal decomposition technique for fully coupled problems in structural acoustics. Debashis Basu and Suresh Kumar S. (State Univ. of New York at Buffalo, 212 Ketter Hall, Amherst, NY 14260)

Problems in structural acoustics are inherently coupled. Whenever a structure vibrates, there is a noise associated to it. Similarly, any noise would induce vibrations in the structure, however small, depending on how thin it is. The indirect boundary element technique has been used for quite some time now to model the acoustic domain. Again, plate/shell formulations for thin structures using finite elements are well known. Usually, for a coupled structural acoustic problem, the associated eigenvalue problem of the structure is solved first, and then the acoustic system is coupled using a predetermined number of modal participation. For weak coupling, this technique gives accurate results considering a few modes, while a greater number of modes are necessary for problems with strong coupling. An alternate technique is presented in this paper, where the structural degrees of freedom are condensed to couple directly with the acoustic system. For the frequency under consideration, there are absolutely no assumptions involved and the coupling is exact. Several examples illustrate the validity of the method. For problems where the level of coupling (weak or strong) and the number of modes to be included are difficult to ascertain, the current technique provides an excellent alternative. With noise control playing a major role in developed and developing countries, a better simulation of the field problem clearly identifies areas for noise reduction measures.

4:15

4pSA9. Energy balance for radiated waves and propagating vibration in a fluid-loaded beam under point excitation. Svetlana I. Kovinskaya (Sci. Res. Ctr. for Ecological Safety, Russian Acad. of Sci., Korpunaya St. 18, St. Petersburg 197042, Russia, svetlana@srces.samson.spb.su)

Using a compensating solution method [S. Kovinskaya and I. Krasnov, *Acta Acust.* **82**, 237 (1996)] a field of flexural waves propagating along a submerged beam under a point excitation and a radiated acoustical field are represented analytically. The method is based on the Fourier integral transform of the vibration equation and is similar to the hybrid method [J. M. Cuschieri and D. Feit, *J. Acoust. Soc. Am.* **95**, 1998–2005 (1994)]. The main and compensating solutions are determined as a result of the inverse Fourier transform along a real wave number axis of nonuniform and uniform equations, correspondingly. A sum of the main and compensating solutions must supply the vibration radiation condition. Let the last equation determine an indefinite coefficient in the compensating solution. The energy balance equation based on the solution of the problem leads to an energetic equivalence between a decreasing vibration and radiated spherical wave and the same between a cylindrical acoustical wave and an energy difference in the excitation point and infinity for propagating flexural wave.

Session 4pSCa

Speech Communication: Production and Dynamic Modeling I (Poster Session)

P. F. Castellanos, Chair

*Division of Otolaryngology, Head and Neck Surgery, University of Maryland, 22 South Greene Street,
Baltimore, Maryland 21201*

All posters will be on display from 1:15 p.m. to 3:15 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:15 p.m. to 2:15 p.m. and contributors of even-numbered papers will be at their posters from 2:15 p.m. to 3:15 p.m.

Contributed Papers

4pSCa1. Normalization for articulatory recovery. Richard S. McGowan (Sensimetrics Corp., 26 Lansdowne St., Cambridge, MA 01239)

Articulatory recovery in an analysis-by-synthesis procedure must account for individual differences in vocal tracts. In the procedure described here, all individual vocal tracts are referenced to the vocal tract of an articulatory synthesizer, known as the standard vocal tract. In order to normalize for any given talker, characteristics of the standard vocal tract are adjusted until its sound output closely matches the sound output of the talker over a range of speech sounds. After these standard vocal tract characteristics are fixed, the analysis-by-synthesis procedure is enabled to recover the articulation from the talker. Experiments are conducted with articulatory and acoustic data from the Wisconsin x-ray microbeam database. The midsagittal sections of the standard vocal tract are made to match those of the x-ray microbeam over a set of phones. Characteristics of the standard vocal tract are adjusted so that its acoustic output matches that of the talker as closely as possible over this range of phones. The normalization procedure is checked by choosing a novel utterance for recovery, and comparing the resulting recovered midsagittals with and without prior normalization. [This work was supported by Grant NIDCD 01247.]

4pSCa2. Acoustic-to-articulatory mapping of isolated and intervocalic fricatives. Edward L. Riegelsberger and Ashok K. Krishnamurthy (Dept. of Elec. Eng., Ohio State Univ., 2015 Neil Ave., Columbus, OH 43210, riegelse@er4.eng.ohio-state.edu)

While acoustic-to-articulatory mapping has been successful for purely voiced speech, a complete acoustic-to-articulatory mapping solution for all classes of speech sounds remains to be found. This work investigates extending acoustic-to-articulatory mapping techniques to fricative sounds. A normalized spectral correlation measure has been developed for an acoustic-to-articulatory mapping routine that produces good acoustic fits and reasonable articulatory configurations for isolated fricatives. The mapping routine can, by using this distance measure along with amplitude information, effectively separate fricative classes by place of articulation. Based on results with isolated fricatives, the acoustic-to-articulatory mapping of fricatives has been extended to the intervocalic case, where information in the voiced segments preceding and following a fricative is used to initialize its optimization. Confidence in the acoustic-to-articulatory mapping of fricatives in isolation influences how they are used in this dynamic case. Heuristics for controlling frication amplitude and mixed excitation are discussed.

4pSCa3. Visible speech revisited: An acoustically driven model of lip and tongue motion. Jay T. Moody (Dept. of Cognit. Sci., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0515, jmoody@cogsci.ucsd.edu) and Maureen Stone (Univ. of Maryland Med. School, Baltimore, MD 21201)

A method is presented for converting acoustic speech data into a "speech readable" movie of a canonical talking face using a neural network. Target values for the network are created by finding the principal components of a set of video frames, with each frame consisting of side-by-side images of the face and tongue of a single speaker. Tongue contours are extracted from mid-sagittal ultrasound images. Input to the network consists of a set of cepstral parameters and their derivatives calculated over 22-ms windows (overlapping by 11 ms). Two such input frames are matched to each video frame (33 ms). Recurrent (backward) connections in the network encourage it to learn not only the acoustic-articulatory pairings, but also articulatory trajectories (expectations of the next articulatory state based on recent articulatory states). It is hypothesized that this supplemental trajectory information helps to alleviate the uncertainties inherent in the vocal tract inverse mapping. After training, the network is presented with new (untrained) tokens (audio only) of utterances from the training corpus and the network's video output is recorded. The network's output is compared to the actual recorded video. [Work supported by US Dept. of Education and NIH.]

4pSCa4. Muscle-based modeling of facial dynamics during speech. Jorge C. Lucero, Kevin G. Munhall (Dept. of Psych., Queen's Univ., Kingston, ON, Canada), Eric Vatikiotis-Bateson (ATR Human Information Processing Res. Labs., Kyoto, Japan), Vincent L. Gracco (Haskins Labs., New Haven, CT 06511-6695), and Demetri Terzopoulos (Univ. of Toronto, Toronto, Canada)

A dynamical, muscle-based model of the face is being developed to extend the facial model of Terzopoulos *et al.* [e.g., D. Terzopoulos and K. Waters, IEEE Trans. Pattern Anal. Machine Intell. **15**, 569-579 (1993)]. The purpose of this work is to characterize facial dynamics during speech with a physiologically realistic model. The model consists of a multilayered deformable mesh of lumped masses connected by springs and viscous elements which represents the layered structure of facial tissue. The spring and viscous constants approximate the stress/strain and viscous characteristics of facial skin. The mesh is deformed by forces generated by a set of modeled muscles of facial expression whose physical characteristics are being determined empirically. The shape of the facial model is individualized to subjects facial morphology with data from a laser ranger finder (Cyberware scanner). In this report, work on driving the model with intramuscular, perioral EMG signals is presented. Recordings of the sampled

three-dimensional position of a subject's facial surface using OPTOTRAK are compared to the patterns of deformation of the epidermal mesh in the model. Results will be discussed in terms of the strengths and weaknesses of this modeling approach. [Work supported by NIH-NIDCD Grant No. DC-00594 and NSERC.]

4pSCa5. The effect of task manipulation on the temporal ordering of articulatory events. Susan Shaiman (Dept. of Commun. Sci. and Disord., Univ. of Pittsburgh, 4033 Forbes Tower, Pittsburgh, PA 15260, shaiman@csd.upmc.edu)

This investigation examined the sequencing of lip and jaw movement onsets and peak velocities for bilabial closure as a function of speaking rate, phonetic context, and position of the closing gesture within the utterance. Articulatory movements were transduced using head-mounted strain gauge transducers. Five normal speakers repeated the target words /paep/, /paeps/, and /paepst/, within the carrier phrase, "Now say ____ again." Speaking rate (normal, fast, and slow) was manipulated using a magnitude production scale with no external time constraints. Most subjects demonstrated consistency in the sequencing of articulatory events at fast speaking rates, while normal and slow rates evidenced increasing variability. The position of the closing gesture within the utterance also affected the sequencing, with the closing gesture for the first /p/ usually exhibiting greater consistency than the closing gesture for the second /p/. The production of a single consonant versus consonant clusters appeared to have little impact on the sequencing of articulatory events for bilabial closure. Despite these general trends, individual subject differences were evident, with some speakers demonstrating greater consistency than others. These findings suggest that temporal ordering of articulatory events is flexibly organized based on the specific task demands. [Work supported in part by NSERC.]

4pSCa6. Kinematic characteristics of gestural onsets and the single-geminate consonant distinction in Italian and Finnish. Margaret Hall Dunn (Haskins Labs., 270 Crown St., New Haven, CT 06511, dunn@haskins.yale.edu)

Selspot traces of lip aperture change in the movement-to-closure portions (movement from preceding vowel into consonant) of single and geminate bilabial consonants in two unrelated languages, Italian and Finnish, were assessed to test the hypothesis that geminates are equivalent to sequences of identical overlapping single consonant gestures. If the closing portions of single and geminate consonants exhibited the same kinematic structure, this would be evidence that the geminate is composed of overlapping gestures. Duration of closing movement, displacement during closing, and peak velocity achieved during closing were compared for productions of single and geminate bilabial consonants for four speakers in each language. For Italian, differences in these kinematic variables during the closing portions of singles and geminates could be accounted for in terms of differences in the timing of consonants with respect to preceding vowels and the overlapping-gesture hypothesis was supported; in Finnish the kinematics of the two types of consonants was not equivalent, and it was concluded that Finnish includes a distinct inventory of gestures for the production of long, rather than geminate, consonants. [Work supported by NIH Grant No. DC-00121 to Haskins Laboratories.]

4pSCa7. Temporal relationships of the articulatory, acoustic, and perceptual effects of fricative-vowel coarticulation. Ying Xu (Dept. Speech and Hearing Sci., The Ohio State Univ., Columbus, OH 43210)

Anticipatory coarticulation has been extensively studied as part of the general phenomenon of coarticulation. However, the temporal relationships of the articulatory, acoustic, and perceptual effects of this particular type of coarticulation have not received much attention. The purpose of this study was to systematically examine lip protrusion gestures, acoustic and perceptual effects in fricative-vowel contexts, and explore the temporal relationships among these three effects. The onset of lip protrusion

and acoustic effects as obtained and compared by employing a minimal contrast technique. The results showed that the onset of lip protrusion was much more consistent in a relative timeframe than in an absolute timeframe, and it was much earlier than, but independent of, the onset of traceable changes in formant frequencies. A reliable effect of anticipatory coarticulation on perception was also observed and the onset of this perceptual effect occurs tens of milliseconds later than the onset of the acoustic effect.

4pSCa8. Modeling the acoustics of American English /r/. Carol Y. Espy-Wilson (Dept. of Elec. and Comput. Eng., Boston Univ., Boston, MA 02215), Narayanan Shrikanth, Abeer Alwan (Univ. of California, Los Angeles, CA, 90024), and Suzanne E. Boyce (Boston Univ., Boston, MA 02215)

It is often assumed that speakers use idiosyncratic combinations of constrictions at the lips, pharynx, and along the hard palate to achieve the low F_3 's—typically between 1300 and 1800 Hz—characteristic of American English /r/. This account was tested in a modeling study using attested vocal tract dimensions derived from three-dimensional magnetic resonance imaging data from four speakers [Alwan *et al.*, *J. Acoust. Soc. Am.* (to be published) (1997)]. These dimensions were used as input to the Maeda computer model. It was determined that (1) a constriction in the post-alveolar region of the palate and at least one other constriction are needed to bring F_3 into the 1300- to 1800-Hz range, (2) the maximum effect of the lip constriction was around 100 Hz, while that of the pharyngeal constriction was around 200 Hz, and (3) the effects of changing constrictions were not additive. Over all manipulations, the lowest F_3 achieved was 1495 Hz, which is high compared to many /r/ productions. Using "retroflex" dimensions, the lowest F_3 achieved was 2224 Hz. It may be that additional acoustic mechanisms are required to account for the full range of acoustic profiles seen for /r/. [Work supported by NSF and NIH.]

4pSCa9. Acoustic estimates of vowel centralization in persons who stutter. Michael Blomgren, Michael P. Robb, and Yang Chen (Dept. of Commun. Sci., Univ. of Connecticut, 850 Bolton Rd., U-85, Storrs, CT 06269-1085, mib95003@uconnvm.uconn.edu)

The magnitude of tongue movement for fluently produced utterances of persons who stutter and those of normally fluent controls was inferred through examination of the vocal tract vowel space. Fifteen adult males served as subjects comprising separate groups of untreated stutterers, stutterers who had completed a fluency-shaping treatment program, and non-stuttering controls. The steady-state portion of formant one (F_1) and formant two (F_2) was examined in the production of various CVC tokens embedded in a carrier phrase. Average F_1 and F_2 values for /i/, /a/, and /u/ were plotted in an F_1 - F_2 vowel space triangle for each subject. The corresponding vowel space was measured by determining the area (Hz^2) of the triangle. Results indicated the untreated group of persons who stutter had the smallest average vowel space, while the largest vowel space was observed in the control group. Discussion focuses on the vocal tract articulation characterizing fluent speech production.

4pSCa10. Using principal component analysis of tongue surface shapes to distinguish among vowels and speakers. Maureen Stone (Univ. of Maryland, School of Medicine, Div. of Otolaryngol.-HNS, Frenkil Bldg., Ste. 500, 16 S. Eutaw St., Baltimore, MD 21201), Y. Cheng (Univ. of Maryland, College Park, MD 20742), and A. Lundberg (Johns Hopkins Univ., Baltimore, MD 21218)

The present study uses principal component analysis (PCA) to examine sagittal tongue contours for five English vowels taken from ultrasound images. The vowels are repeated three to five times each in a /pVp/ carrier utterance. Of particular interest is the use of plots of the coefficients of the eigenvectors to distinguish both subjects and vowels, and the building of a

linear model to fit their family of data. Data will be transformed to normalize surface length across contours. Short contours will be stretched in the x direction to the length of the longest curve. The x and y dimensions will be stretched equally for each curve to preserve scale. Preliminary data indicate fairly good success distinguishing among three subjects and four vowels using a linear model based on the first few components. Methods of improving this result are being explored using factor analysis or optimal fitting.

4pSCa11. A computerized ultrasound transducer scanning system.

Edward Davis (Dept. of Mech. Eng., Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD 21218) and Maureen Stone (Univ. of Maryland, Baltimore, MD 21201-1595)

A computerized electromechanical ultrasound transducer holder is being developed to allow representation of three-dimensional tongue motion. Current three-dimensional ultrasound systems are comparatively slow (10 s/volume) and hence are not useful for the study of dynamic three-dimensional speech. The proposed mechanism will (automatically) both scan along and rotate about an axis, sensing transducer position and providing feedback. This motion will be motorized and controlled by a computer according to custom-developed software and parameters. One can program the computer to perform a synchronized series of approximately five sagittal or coronal slices. These slices can later be assembled and interpolated to form a three-dimensional surface model. Mechanically, this system is advantageous because it rotates the transducer about a virtual center, which is located along the axis of the center of the arc of the piezoelectric crystals within the transducer, and does not necessitate additional mechanical structures near the subject. The electrical and computer control system will allow automated imaging of a series of coronal or sagittal slices synchronized in time and located precisely in space. This system will be added to and used in conjunction with the HATS system described in Stone and Davis [J. Acoust. Soc. Am. **98**, 3107–3112 (1995)]. [Work supported by Northrup Grumman and NIH.]

4pSCa12. Acoustic, aerodynamic, and physiological characteristics of modal and vocal fry registers.

Yang Chen (Dept. of Commun. Sci., Univ. of Connecticut, Storrs, CT 06268, yac93001@uconnvm.uconn.edu), Michael Blomgren (Univ. of Connecticut, Storrs, CT 06268), Manwa L. Ng (Illinois State Univ., Normal, IL 61790), and Harvey R. Gilbert (Univ. of Connecticut, Storrs, CT 06268)

This study examined production of the modal and vocal fry registers using various acoustic and aerodynamic parameters. Airflow (AF), air pressure (AP), electroglottographic (EGG), and acoustic data were obtained simultaneously for four isolated vowels ($/i/$, $/æ/$, $/a/$, $/u/$) and the stop consonant produced in a string of $/pi/$ syllables for both modal and vocal fry registers. Twenty normal speakers (10 males, 10 females) who were able to produce vocal fry participated in this study. Results demonstrated considerable differences among AF, AP, EGG, and acoustic signals for the modal and vocal fry registers. The fundamental frequency in vocal fry register (male: 49.1 Hz; female: 48.1 Hz) was found to be significantly lower than that in modal register (male: 117.5 Hz; female: 211.0 Hz). Also, the average air pressure values decreased in vocal fry register (male: 5.51 H₂O; female: 5.25 H₂O) compared to values obtained in the modal register (male: 7.45 H₂O; female: 7.56 H₂O). In vowel production, the airflow rate in the modal register was almost three times as high as the airflow rate in vocal fry register. For the production of $/pi/$, airflow rate associated with the stop consonant in the modal register was approximately 1.5 times higher than that in the vocal fry register.

4pSCa13. Two types of vowel devoicing in Japanese: Evidence from articulatory data.

Ayako Tsuchida (Dept. of Linguist., Cornell Univ., Ithaca, NY 14853, at18@cornell.edu), Shigeru Kiritani, and Seiji Niimi (Univ. of Tokyo, Bunkyo-ku, Tokyo, 113 Japan)

Japanese high vowels $[i, u]$ become devoiced when they occur between voiceless segments: e.g., $[kita]$ “north.” Vowel devoicing (VD) occurs systematically, except when a high vowel appears between two voiceless fricatives, where VD is less consistent and nonobligatory. The present study examined the laryngeal gestures during the production of Japanese vowels in various phonological contexts using fiberoptic and electromyographic techniques. Stimuli consisted of $[CiCe]$ words, where the consonants were stops (S) or fricatives (F). The results revealed that the glottal gestures for typical devoiced vowels (those in $[SiSe]$, $[FiSe]$, or $[SiFe]$) and devoiced vowels in $[FiFe]$ were distinct, although the two types were indistinguishable in the spectrograms. Consistent with observations in a previous study [H. Hirose, *Phonetica* **23**, 156–170 (1971)], $[CiC]$ sequences with typical devoiced vowels were produced with a wide open glottis accompanied with high activities of the abductor muscle. In contrast, in $[FiF]$ sequences, the glottis was wide open in each of the fricatives, but was almost closed during the vowel, resulting in a bimodal pattern. When the glottis reached complete closure during the vowel, a voiced vowel was observed. The present study shows two distinct mechanisms for VD, suggesting that they are two independent processes.

4pSCa14. Do children with phonological disorders use more ballistic articulatory movements?

Marios Fourakis (Dept. of Speech and Hearing Sci., Ohio State Univ., 110 Pressey, 1070 Carmack Rd., Columbus, OH 43210), Mary E. Beckman, and Jan Edwards (Ohio State Univ., Columbus, OH 43210)

This examination of speech motor control in phonologically disordered children adapted a paradigm that has been used in the dysarthria literature [Weismer *et al.*, J. Acoust. Soc. Am. **91**, 1085–1098 (1992)]. Six preschool-aged children with phonological disorders, six age controls, and six adults repeated the phrases “baby dog” and “good baby” at normal and fast rates. $F2$ tracks of the fast rate tokens were processed using an in-house algorithm to calculate the transition slope in Hz/ms. For the alveolar and velar stops (but not for the labial stops) the $F2$ slope distributions for the typically developing children were quite similar to those of the adults, and both differed from the children with phonological disorders in showing smaller slope values. The phonologically disordered children had faster movements than either their age controls or adults, but only in the sequences where the tongue–jaw complex was harnessed for the consonant as well as for the contiguous vowel. This pattern suggests that the phonologically disordered children are using a faster, more ballistic gesture for lowering the tongue–jaw complex for all three stop places of articulation, while the typically developing children and adults use this strategy only after a labial closure.

4pSCa15. Disordered phonology: An acoustic description of the acquisition of the anterior distinction among coronal fricatives.

Adele W. Miccio (Dept. of Commun. Disord., Penn State Univ., 105 Moore Bldg., University Park, PA 16802, awm4@psu.edu) and Karen Forrest (Indiana Univ., Bloomington, IN 47405)

This paper reports on a longitudinal analysis of the acoustic properties of speech samples of a child, age 5 years and 7 months, with disordered phonology. The child, who produced $[s]$ for both alveolar and postalveolar fricatives, was enrolled in a remediation program that focused on the correct production of the voiceless postalveolar fricative $[ʃ]$ “esh.” Training involved imitation of $[ʃ]$ in isolation and spontaneous production of $[s]$ and $[ʃ]$ in minimally paired words. Acoustic measurements were taken prior to treatment, immediately following the 2-week treatment program, and again 2 months following termination of treatment. Temporal and spectral measurements were conducted. Results indicated no significant differences between $[s]$ produced for the target alveolar fricative $/s/$ vs $[s]$

produced for the target postalveolar fricative /ʃ/ prior to treatment. Following treatment, the child produced the anterior–nonanterior distinction among coronal fricatives. Acoustic measurements showed that she did not simply add a postalveolar fricative to her repertoire of speech sounds but also changed her productions of target-appropriate /s/. [Work supported in part by NIH-DC 00260.]

4pSCa16. Variability in /s/ and /ʃ/ productions within and across talkers. Rochelle S. Newman (Psych. Dept., Park Hall, SUNY, Buffalo, NY 14260)

A primary issue in speech perception research is the apparent ‘‘lack of invariance’’ between the acoustic information in a signal and the listener’s phonemic perception. The same intended phoneme can be produced with a wide range of acoustic values, and different intended phonemes may be produced with the same values. Although the existence of this variability is not in question, the degree to which listeners have to account for it is unclear. Some studies have attempted to measure the extent of this variability, but few have done so for fricatives, making it difficult to determine the degree to which individuals need to normalize fricative input for the individual talker. The current experiment attempts to rectify this hole in the literature by examining the fricative centroids for over 100 utterances beginning with /s/ and /ʃ/ from each of 20 different speakers. Although there was variability in the centroid values, the categories showed virtually no overlap within a talker. However, there was substantial overlap across talkers. This suggests that if listeners use centroids to cue the /s/–/ʃ/ distinction, they would first need to normalize the signal for the individual talker’s frequency range. [Work supported by NIDCD Grant No. R01 DC00219 to SUNY at Buffalo.]

4pSCa17. An articulatory and acoustic analysis of English flaps: Evidence for attentional reduction in production. Kenneth J. de Jong (Dept. of Linguist., Indiana Univ., Bloomington, IN 47405)

Traditional models of American English phonology include a rule that coronal stops appear as flaps before unstressed vowels. Recent studies suggest that this flapping is better explained as the result of blending the articulatory specifications for the stops with neighboring vowels. This blending produces compromise articulatory positions for the stops, which

cross over the perceptual boundary between stops and flaps. This paper tests this explanation against an x-ray microbeam corpus of American English stops. The occurrence of flapping was determined via transcriptions. Regression analyses indicate that acoustic parameters such as VOT and voicing occurrence better predict transcriptions than articulatory parameters such as tongue position and motion during closure. In addition, articulatory analyses indicate a variation from [t] to flap which is generally consistent with a blending account. However, the contextual influence from the consonants on neighboring vowels and various aspects of articulatory postures in the consonants themselves is more consistent with a reduction of the stop, rather than a blending with neighboring vowels. These results suggest that flapping results from a linguistic convention which allows hypoarticulation of unstressed items and a secondary result of some of these items straying across an acoustic boundary. [Supported by the NSF and NIDCD.]

4pSCa18. Acoustic evidence for featural and subfeatural errors in speech production. Stefan Frisch and Richard Wright (Speech Res. Lab., Indiana Univ., Bloomington, IN 47401)

Phonological speech errors are a traditional source of psycholinguistic evidence for the representations of phonology theory. In an electromyographic (EMG) study of experimentally induced phonological speech errors, Mowrey and MacKay [J. Acoust. Soc. Am. **88**, 1299–1312] found that speech errors frequently occur at a subfeatural, gestural level, with no apparent effect on the percept of the word. Mowrey and MacKay’s study considered the activity of a single muscle, and thus was unable to determine whether single gestures acted independently of gestural constellations. This study is a preliminary report from an ongoing acoustic analysis of speech errors. The data are tape recordings of an error-inducing experiment using nonsense tongue twisters. Recordings of six speakers producing four different tongue twisters targeting /s/ and /z/, e.g., sit zap zoo sip, were digitized and analyzed. Some errors involved multiple changes in acoustic properties, including simultaneous changes in periodicity, amplitude of frication, and duration, while others involved a subset of these properties. This evidence suggests that errors can occur at both the single gesture level, affecting noncontrastive acoustic properties, and at the level of the gestural complex or segment, creating a perceptible, linguistically contrastive change. [Work supported by NIH Grant No. DC00012.]

THURSDAY AFTERNOON, 19 JUNE 1997

ROOM D, 3:30 TO 5:00 P.M.

Session 4pSCb

Speech Communication: Production and Dynamic Modeling II

P. F. Castellanos, Chair

Division of Otolaryngology, Head and Neck Surgery, University of Maryland, 22 South Greene Street, Baltimore, Maryland 21201

Contributed Papers

3:30

4pSCb1. Evidence for a second laryngeal sound source. P. F. Castellanos (Div. of Otolaryngol.—Head and Neck Surgery, Univ. of Maryland School of Medicine, 22 S. Greene St., Baltimore, MD 21201) and S. A. Elder (U.S. Naval Acad., Annapolis, MD 21402)

Sondhi reflectionless tube and strobed video have been used to investigate single glottal pulse gestures in the vocal fry range. There appear to be two sources of sound in normal VF phonation, one monopole, the other

quadrupole. It is quadrupole sound, in fact, that seems to define the shape of observed pressure trace in the single glottic pulse, or SGP. This sound pulse, which lasts 10 ms or less, resembles a single cycle of negative sine wave beginning just before the closing phase, and may be recognized even in sound emissions outside the tube, where continuous tone samples can be identified as SGP wave trains, each link beginning with a characteristic downturn. Monopole sound, emitted in shorter pulses during the abrupt zipping and sometimes during unzipping phases of the SGP, shows up along the main wave trace in the Sondhi tube as a small superposed peak, followed by a string of head echoes. The source of quadrupole sound can

be traced to fluctuating Bernoulli pressures during closure which produce opposing forces on the vocal folds. The quadrupole or q -wave forms the acoustic signature of the SGP.

3:45

4pSCb2. Control of oral closure in alveolar and velar stop consonant production. Anders Lofqvist and Vincent L. Gracco (Haskins Labs., 270 Crown St., New Haven, CT 06511, lofquist@haskins.yale.edu)

Earlier work has shown that the lips are moving at a high velocity at the instant of oral closure for bilabial stop consonants, resulting in tissue compression. The lips may thus have virtual targets that would require them to move beyond each other. The present study examines events at the oral closure for stops produced with the tongue. Tongue movements were recorded using a magnetometer system. Four subjects participated and produced ten tokens each of VCV sequences with all possible combinations of the three vowels /i, a, u/ and the consonants /t, d, k, g/. Consistent with the results for bilabial stops, the tongue was moving at a high velocity at the instant of oral closure. The tongue movement trajectories were more complex with a larger horizontal component than the lips. For velar stops, the tongue body was usually moving forward at stop closure, with the velocity influenced by vowel context. For alveolar stops, the tongue tip horizontal movement direction depended on the preceding vowel, but also differed across subjects. Between the onset and release of the oral closure, the tongue tip and tongue body moved through a trajectory of usually less than 1 cm. [Work supported by NIH.]

4:00

4pSCb3. Congruence of articulatory and acoustic variability. Alice Faber (Haskins Labs., 270 Crown St., New Haven, CT 06511) and Julie M. Brown (Univ. of Connecticut and Haskins Labs., New Haven, CT)

Johnson *et al.* [J. Acoust. Soc. Am. **94**, 701–714 (1993)] suggest, on the basis of observed inter-speaker variability in discriminant analyses of articulatory measures, that speakers utilize acoustically defined targets in speech production. The present paper compares inter-speaker variability in simultaneously recorded acoustic and articulatory data (s_t words) from five New England speakers. The articulatory data were x and y coordinates of coils on the tongue, lips, and jaw, transduced by the Haskins Laboratories EMMA system and recorded at three locations in the target vowel; acoustic data were $F1$, $F2$, and $F3$ measures at the same temporal locations. Discriminant analyses of the articulatory and acoustic data sets reveal congruent patterns of inter-speaker variability in the two domains. The inter-speaker differences do not reflect superficial dialect or idiolect differences (e.g., extent to which /a/ and /ɔ/ contrast, or tendency to glottalize syllable-final /t/). Rather, they reflect differences in the way subjects vary jaw position, especially height, suggesting an anatomical basis for the observed differences. Thus, within the limits of their oral morphology, these speakers are using comparable articulatory targets for speech sounds.

4:15

4pSCb4. Dynamic shear modulus of vocal-fold tissues and phonosurgical biomaterials. Roger W. Chan and Ingo R. Titze (Nat. Ctr. for Voice and Speech, Dept. of Speech Pathol. and Audiol., Univ. of Iowa, Iowa City, IA 52242)

Knowledge of the mechanical properties of vocal fold tissues is necessary for the constitutive modeling of vocal-fold mechanics. Assuming transverse isotropy in a linear, elastic continuum model of vocal-fold tissues, five material constants are needed to solve the constitutive equation

[D. A. Berry and I. R. Titze, J. Acoust. Soc. Am. **100**, 3345–3354 (1996)]. Among these constants, Poisson's ratios can be estimated by assuming tissue incompressibility, while the shear moduli and Young's moduli are related to one another. A parallel-plate rotational rheometer was used to examine the dynamic shear behavior of human vocal fold tissues and three commonly used phonosurgical biomaterials (bovine collagen suspension, absorbable gelatin suspension, and human subcutaneous fat). In oscillation at 0.1–10 Hz and at 37 °C, the magnitude of dynamic shear modulus of vocal fold mucosa was on the order of 100 Pa, close to that of fat. The shear modulus magnitudes of collagen and gelatin were an order of magnitude higher. These results suggest that the use of fat for vocal-fold augmentation surgery is more conducive to phonation, because of its similarity to the vocal-fold mucosa in shear stiffness. [Work supported by NIH.]

4:30

4pSCb5. Methods for acoustic-to-articulatory mapping and voice mimic systems. S. Chennoukh, D. Sinder, G. Richard, and J. Flanagan (CAIP Ctr., Rutgers Univ., Piscataway, NJ 08855)

The articulatory codebook approach to voice mimic systems is complicated by large amounts of data and expensive acoustic comparisons when searching the codebook. If, during codebook construction, the acoustic parameters are quantized and mapped back to the articulatory space, the inverse is accomplished without acoustic comparisons. Since the inverse is nonunique, articulatory trajectories must also be estimated to resolve acoustic parameters which map to multiple model shapes. This can be managed with a tracking technique that uses dynamic properties (position and velocity) of each articulatory parameter to estimate the next model shape. A recurrent algorithm results which finds an optimal path through the model shape variations. These approaches are helpful, but they do not address the problem that codebooks may not cover the entire articulatory space. Furthermore, there is no performance measure which indicates the extent of the coverage. This limitation is overcome by an analytic inverse mapping, rather than a table look-up, which relates the first two formant frequencies with articulatory parameters. This relationship, which is well established for vowels, uses a distinctive region model (DRM) to approximate three area function parameters as a function of the first two formant frequencies. [Research supported by ARPA DAST 63-93-C-0064.]

4:45

4pSCb6. Fluid dynamic studies of speech production. D. Sinder, M. Krane, S. Chennoukh, G. Richard, J. Flanagan (CAIP Ctr., Rutgers Univ., Piscataway, NJ 08855), S. Levinson (Lucent Technologies, Bell Labs, Murray Hill, NJ 07974), S. Slimon, and D. Davis (Electric Boat Corp., Groton, CT 06340)

Results are presented describing an investigation of aerodynamic sources in the vocal tract. The expected contribution of this study is a new parametrization for fricative sources based on geometric and flow conditions. The flow in an idealized fricative geometry is computed numerically using a slightly compressible, Reynolds-averaged form of the Navier–Stokes equations. In this formulation, the aerodynamic sources of sound are simulated with minimal approximation. This approach has produced encouraging results for vowel sounds. Comparisons of the computational results with physical experiments using the same geometry have also been performed. The results shed light on the source mechanisms for fricative sounds and also show the need for the development of numerical boundary conditions which can simultaneously handle convective and propagative phenomena. [Research supported by NSF/ARPA IRI-9314946 and ARPA DAST 63-93-C-0064.]

4p THU. PM

Session 4pUW

Underwater Acoustics: Propagation: General Topics

Martin A. Mazur, Chair

Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804

Contributed Papers

1:30

4pUW1. The effect of an absorptive bottom on the propagation of the coherence function in a shallow-water channel. T. Barnard and M. J. Beran (Dept. of Elec. Eng., The Catholic Univ. of America, Cardinal Station, Washington, DC 20064)

When a transverse plane wave propagates through a shallow-water channel with random sound-speed fluctuations, the waveforms at different transverse separations no longer correlate perfectly. The associated coherence falls as the in-line propagation distance and the transverse separation increase. In the (lossless) rigid-bottom case, when the waveforms are represented as a summation of normal modes, the multimodal coherence vector obeys a first-order matrix differential equation with in-line propagation distance as the independent variable. The scattering matrix in this differential equation is a function of transverse separation. As the in-line propagation distance approaches infinity, the coherence vector approaches a constant vector times a scalar Dirac delta function centered at zero transverse separation. If the bottom is absorptive rather than rigid, an additional diffusion term appears in the matrix partial differential equation governing the coherence. Diffusion along the transverse separation axis then prevents the creation of a Dirac delta function as the in-line propagation distance increases without limit. This diffusion occurs whenever the imaginary part of the horizontal wave-number component for a particular mode is non-zero. Some graphical outputs depicting the coherence propagation for an absorptive bottom are presented in this paper. [Work supported by ONR Code 321.]

1:45

4pUW2. Singular value decomposition for normal-mode computations. Ronald T. Kessel (School of Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3055, Victoria, BC V8W 3P6, Canada) and Trevor W. Dawson (Univ. of Victoria, Victoria, BC V8W 3P6, Canada)

Mode models of sound propagation in layered media begin with a search for modes in the complex wave-number plane, followed by computation of the vertical mode functions and mode superposition. Sound propagation modelers would like to search for modes ever more efficiently and thoroughly. The search amounts to the mathematical problem of finding the roots (zeros) of a complex-valued function—a problem that defies definitive solution. Rather than reducing the numerical test for modes (roots) to a single characteristic equation, singular value decomposition (SVD) can be used to test when the global matrix describing the propagation of plane waves through the layered media is singular, which is an equivalent condition for a mode. The advantages of this approach are that it is numerically stable, it automatically gives the vertical mode functions (although un-normalized), it can detect duct and interface modes at any depth, and it can identify and resolve close-mode pairs (double roots) and their antisymmetric mode functions occurring in weakly coupled sound channels.

2:00

4pUW3. Mode excitation for a source on a rough, elastic, sloping bottom. Arthur B. Baggeroer, Henrik Schmidt, and Brian Sperry (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

In the ATOC program, the source was sited on a slope at a depth of the SOFAR axis so that the signal would excite the axial modes efficiently for long-range propagation. One of the important issues was the elevation of the source off the seafloor. Buoying it on a mooring into the water column presented significantly more engineering challenges than simply resting it on a platform on the seafloor. Here, the impact upon the mode excitation spectrum of the source depth on a rough, elastic, sloping bottom is examined using the new range-dependent OASES code. This code matches vertical wave numbers across range-independent sectors, which may include interface roughness, as it steps out in range. It has been very successful in modeling the related problem of the effect of epicenter depth for T-phase excitation on a sloping bottom [Sperry *et al.*, J. Acoust. Soc. Am. **100**, 2641(A) (1996)]. A Munk sound-speed profile is assumed with a minimum at 1000 m and a sloping bottom out to a range of 30 km after which the bottom depth is constant. The field is projected onto the depth-dependent modes to determine their excitation. Also, the group delays are examined to determine the impact on travel times. Initial results suggest that even a modest (100 m) elevation concentrates the power in the low-order modes. In addition to the source elevation, the slope gradient and roughness and the elastic properties of the bottom are important parameters. [Work supported by ONR.]

2:15

4pUW4. Normal-mode analysis for signal fluctuations in the Yellow Sea. Renhe Zhang, Zhenge Sun, Liewei Sha, Liangying Lei, Longsheng Hao, Nan Sun, Fenghua Li (Natl. Lab. of Acoust., Chinese Acad. of Sci., Beijing 100080, PROC), Ji-Xun Zhou, Peter H. Rogers, and Gary W. Caille (Georgia Inst. of Technol., Atlanta, GA 30332)

This paper presents a normal mode approach for analyzing signal fluctuation in shallow water with a thermocline which fluctuates due to tides and internal waves. In the Yellow Sea 1996 experiment, variations of thermocline structure and internal waves were observed by CTD and thermistor chains. The cw and FM acoustic signals were measured over a 12-h period with two 16-hydrophone vertical arrays deployed 3.11 and 10.32 km from the source. The cw pulses were used to observe fluctuation at four specified frequencies. Amplitude fluctuations of more than 20 dB were observed. The FM signals were used to isolate individual normal modes by pulse compression and mode filtering. The influence of internal waves on the normal-mode structure and signal fluctuation was examined. It was found that the variation of the thermocline was significant in determining acoustic signal fluctuation. The normal-mode structure and dispersion relations in the channel depend strongly on the thermocline. Variation of the thermocline thus causes phase and group velocity to become variable. This leads to signal fluctuation, since the amplitude of the received signal is determined by interference between arriving modes. The study also showed that the amplitude fluctuation is weaker at shorter range.

2:30

4pUW5. Sound-field sensitivity to geoacoustic parameters in range-dependent media using normal modes. Ronald T. Kessel (School of Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3055, Victoria, BC V8W 3P6, Canada)

Sensitivity in sound propagation modeling is a measure of the ability of environmental parameters to affect the sound field. The most sensitive parameters are often considered most important because errors or uncertainties in their values are among the most likely sources of error in the model predictions. There are many ways to define the measure of sensitivity. It generally depends on the sound field itself, on the energy flow through regions where the geoacoustic parameters under consideration lie, and on the way the field is monitored by receivers. But it is useful to define sensitivity without reference to a particular field or source-receiver configuration, as if it were a parameter of the environment alone. Then conclusions drawn from a sensitivity study apply as much as possible to all source-receiver configurations, and for both inverse and forward modeling. Such a measure has been derived using an adiabatic mode model for range-dependent media, exploiting the fact that the normal modes of vibration embody the essential properties of the field without reference to a particular source-receiver configuration. The new measure is demonstrated for range-dependent media using computer simulations.

2:45

4pUW6. Acoustic mode decoupling via environmentally adaptive coordinates. Peter C. Mignerey (Acoust. Div. 7120, Naval Res. Lab., Washington DC 20375-5350, mignerey@nrl.navy.mil)

A normal mode expansion is used to relate the coupling of energy between modes to horizontal derivatives of the eigenfunctions. A curvilinear coordinate system is constructed using a WKB approximation for a reference eigenfunction such that horizontal derivatives causing coupling vanish. The curvilinear coordinate system is defined such that constant surfaces of the vertical wave number integral coincide with surfaces of constant curvilinear depth. Thus any particular zero of an eigenfunction lies on a constant depth surface in the curvilinear system. The horizontal derivatives of all other modes depend on a ratio of vertical wave numbers with the reference mode. Nearly constant horizontal behavior of this wave number ratio supports the decoupling of all modes in the curvilinear system. Such coordinate systems naturally adapt to the environment even with realistic sound-speed profiles that include discontinuities as well as sloping bottom bathymetry. They provide an adiabatic normal-mode basis for constructing acoustic models in the fully three-dimensional environments of continental shelf regions. Environmentally adaptive coordinate systems also provide a rational basis for interpolating three-dimensional environmental data.

3:00

4pUW7. The generalized eigenvalue problem, warping matrices, and the transformation of an isovelocity environment to a variable velocity environment. M. F. Werby (Naval Res. Lab., Code 7181, Stennis Space Center, MS 39529) and N. A. Sidorovskaia (Univ. of New Orleans, New Orleans, LA 70148)

The calculation of eigenvalues for a layered isovelocity oceanic environment and associated vertical modes is very quick and easy to do. A method of transforming the isovelocity problem to a variable velocity problem, by means of a matrix transformation based on the theory of generalized eigenvalues, is described. Along with modern transformation techniques, the rapid reconstruction of variable velocity solutions is demonstrated. It is possible to determine the mode spectrum extremely rapidly by first starting out with those associated with a reference isovelocity eigenvalue set, and then by using an iteration procedure based on integral transform methods, the Householder transformation, polar decomposition theorems, shifting, and deflation methods in linear algebra to converge to

the set of eigenvalues rapidly. A first iteration may take a fraction of a second to calculate on a PC, while full iteration takes on the order of a second to calculate 100 eigenvalues. [Work sponsored by ONR, the Naval Res. Lab., and the Univ. of New Orleans.]

3:15

4pUW8. An adaptive coupled range-dependent normal-mode method. M. F. Werby (Naval Res. Lab., Code 7181, Stennis Space Center, MS 39529) and N. A. Sidorovskaia (Univ. of New Orleans, LA 70148)

Based on the warping method and the generalized eigenvalue problem the range-independent solution is extended to that of range dependence. In this method, coupling in range is analytical and may be either one way or two way. Since the range steps are not dependent on finite-difference schemes, the optimal range step at each point is determined. This leads to two advantages: (1) There is no need for input of range step; and (2) the optimal range step is predetermined by several conditions, thus avoiding overkill and reruns to determine stability. Comparisons with other range-dependent methods are presented. [Work sponsored by ONR, the Naval Res. Lab., and the Univ. of New Orleans.]

3:30-3:45 Break

3:45

4pUW9. Analytic implementation of a parabolic equation solution. Dalcio K. Dacol and Michael D. Collins (Naval Res. Lab., Washington, DC 20375, dacol@abyss.nrl.navy.mil)

There are two basic approaches for discretizing the differential equations that are solved in propagation models: (1) Approximate the differential operator with a finite-difference formula; and (2) approximate the medium in terms of layers that may be treated analytically and solve for a set of coefficients. The analytic approach is more efficient for problems involving a small number of layers. Both approaches have been implemented for normal mode and wave-number integration models. The finite-difference approach has been implemented for parabolic equation models. This paper describes an analytic implementation of a parabolic equation model that is efficient for geoacoustic inversion problems [R. J. Cederberg and M. D. Collins, "Application of an improved self-starter to geoacoustic inversion," IEEE J. Ocean. Eng. (in press)]. Using the finite-difference implementation, the self-starter can solve nonlinear inversion problems in minutes on a workstation (techniques that were developed previously require hours of run time). The analytic implementation reduces run times to seconds. [Work supported by ONR.]

4:00

4pUW10. Energy-conserving spectral solutions. Michael D. Collins (Naval Res. Lab., Washington, DC 20375, collins@ram.nrl.navy.mil), Henrik Schmidt (MIT, Cambridge, MA 02139), and William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180)

Energy conservation is an important issue in range-dependent propagation modeling. Energy-conserving coupled mode and parabolic equation solutions have been developed [J. Acoust. Soc. Am. **89**, 1058-1075 (1991)]. This presentation will describe an energy-conserving solution based on the wave-number spectrum. Range dependence is handled by dividing the medium into a sequence of range-independent regions and enforcing the complete energy-conservation condition [J. Acoust. Soc. Am. **94**, 975-982 (1993)] at the vertical interfaces between regions. This

4p THU. PM

condition involves the square root of the horizontal wave number in the forcing function of the spectral equation and in the spectral integral. This approach has been implemented and tested for both fluid and elastic media. [Work supported by ONR.]

4:15

4pUW11. Analysis of selected data from a recent Navy exercise. Stanley A. Chin-Bing, David B. King, Guy V. Norton (Naval Res. Lab., Stennis Space Center, MS 39529-5004), and Jorge C. Novarini (Planning Systems, Inc., Long Beach, MS 39560-9702)

Several ocean acoustic computer models were used to analyze shallow-water, higher-frequency (around 4-kHz) acoustic data taken during a recent Navy exercise. This effort enlisted models (EFEPE-CM, FEPE, ASTRAL, RAYMODE) and computer resources (Cray C90, SGI Challenge-8, 386 desktop PC) that spanned the RDT&E modeling spectrum, from basic research to Fleet operations. The research models were used to identify and prioritize dominant physical mechanisms. Once these were known, errors and weaknesses in the operational models were easily identified and corrected. The result was consistent with predictions from the entire suite of models that compared favorably with data. Advantages were gained in computational speed and accuracy. An acceptable shallow-water, high-frequency, acoustic prediction was obtained in 2 s using a Fleet operational model (ASTRAL) and a 386 desktop PC computer. Two important lessons were relearned: (1) Current ocean acoustic research models include the necessary physics to make accurate high-frequency predictions in shallow-water environments, even when realistic sea-surface conditions dominate; and (2) the more physics included in the models, the less knowledgeable need be the model operators. Details and examples will be presented. [Work supported by ONR/NRL, MMoDS, AUAMP, and DoD HPC.]

4:30

4pUW12. Parabolic equation models for anisotropic sediments. Andrew J. Fredricks, William L. Siegmann (Dept. of Math. Sci., Rensselaer Polytechnic Inst., Troy, NY 12180), and Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

Because of the current interest in shallow water, propagation model capabilities have been developed for elastic, poro-elastic, and poro-acoustic sediments. These models incorporate depth and range heterogeneities in the sediments, which are assumed to be spatially isotropic. However, many shallow environments have a layered structure which often is more appropriately characterized as transversely isotropic. This feature arises from deposition and layering processes influenced by gravity, for example. Transversely isotropic sediments have elastic properties with considerable variations perpendicular to their natural plane, in which relatively little variation occurs. Efficient and accurate two-dimensional PE models are available for a variety of spatially isotropic sediments, and generalizations are needed for transversely isotropic cases. Our initial two-dimensional development assumes that the plane of the sediment is approximately horizontal. The PE formulation is presented for an elastic sediment and then extended to poro-elastic cases. An initial high-order implementation will be described. [Work supported by ONR.]

4:45

4pUW13. Boundary conditions for finite-difference PE solvers. David Yevick (Dept. of Elec. Eng., Queen's Univ., Kingston, ON K7L 3N6, Canada) and David J. Thomson (Defence Res. Establishment Atlantic, Victoria, BC V9A 7N2, Canada)

In theoretical/numerical models of underwater sound propagation, a downgoing radiation condition is usually imposed on the acoustic field as $z \rightarrow \infty$. For most implementations of parabolic equation (PE) solvers, this condition is approximated by appending an absorbing layer to the compu-

tational mesh. Such a layer acts to attenuate any energy that reaches the grid boundary before the waves undergo reflection and return to the ocean region. As shown by Papadakis [J. Acoust. Soc. Am. **92**, 2030–2038 (1992)], this approximate treatment can be replaced with a nonlocal boundary condition (NLBC) that *exactly* transforms the semi-infinite PE problem to an equivalent one in a bounded domain. Papadakis' method involves evaluating a spectral (wave number) integral of a singular kernel that is inversely proportional to the impedance of the subbottom medium. In this paper, an alternate procedure is described for obtaining NLBCs directly from the z -space Crank–Nicolson formulations of both the Tappert and Claerbout PEs. Formulas for the boundary field at range $r + \Delta r$ are derived in terms of the known field along $0 \rightarrow r$ by expanding the appropriate "vertical wave number" operator in powers of $\mathcal{R} = \exp(-\Delta r \partial_r)$ and applying $\mathcal{R}^j \psi(r, z) = \psi(r - j\Delta r, z)$. The effectiveness of these NLBCs is compared to Berenger's matched-layer technique [J. Comp. Phys. **114**, 185–200 (1994)] for several numerical examples relevant to one-way underwater sound propagation.

5:00

4pUW14. The PSTD algorithm: A time-domain method combining the pseudospectral technique and perfectly matched layers. Qing-Huo Liu (Klipsch School of Elec. and Comput. Eng., New Mexico State Univ., Las Cruces, NM 88003)

Simulations of underwater acoustic wave propagation require solutions for large-scale multidimensional problems. The traditional finite-difference time-domain (FDTD) method needs a fine discretization of 8–20 cells per wavelength in order to give accurate results. On the other hand, the pseudospectral method, even though efficient, suffers from the wraparound effect due to the use of discrete Fourier transform. This effect severely limits the applications of the pseudospectral method to large-scale problems. Hence, a popular way of simulating underwater acoustic wave propagation is to use the parabolic equation (PE) methods which neglect back-scattering. In this work, Berenger's perfectly matched layers (PML) are used in the pseudospectral method to eliminate the wraparound effect. To achieve a high accuracy, this method requires only two cells per wavelength which is dictated by the Nyquist sampling theorem. As a result, it can solve at least 64 times larger 3-D problems than the FDTD method with the same requirement in computer memory and CPU time. Hence, full-wave solutions of long-range underwater acoustic wave propagation become possible. Numerous simulations show the superiority of the PSTD method for large-scale problems. [Work supported by a Presidential Early Career Award for Scientists and Engineers through EPA and by Sandia National Laboratories.]

5:15

4pUW15. Perfectly matched layers for acoustic waves in viscous media: Applications to underwater acoustics. Qing-Huo Liu and Jianping Tao (Klipsch School of Elec. and Comput. Eng., New Mexico State Univ., Las Cruces, NM 88003)

Berenger's perfectly matched layer (PML) has recently been proved to exist for elastodynamic equations [Chew and Liu, Schlumberger-Doll-Res. Rep. (1995)]. This fictitious material absorbs all waves with an arbitrary incident angle and an arbitrary frequency without giving rise to any reflections. Therefore, when used as an absorbing boundary condition (ABC) at the computational edge in a finite-difference method, the PML provides orders of magnitude higher absorption than other existing ABCs. In this work, the PML is further extended as an ABC for a finite-difference simulation of acoustic waves in viscous media. For such attenuative media, an additional term involving the time-integrated wavefield is introduced to

account for the coupling between the attenuation from the PML and the normal viscous attenuation. This ABC is highly effective in absorbing outgoing waves at the computational edge even when a dipping interface intersects the outer boundary. This new material ABC is ideal for parallelization on multiprocessor computers. The algorithm is validated by ana-

lytical and numerical solutions. Various two- and three-dimensional numerical simulations will be shown to demonstrate the applications in underwater acoustics and other areas. [Work supported by a Presidential Early Career Award for Scientists and Engineers through EPA and by Sandia National Laboratories.]

THURSDAY AFTERNOON, 19 JUNE 1997

ROOM H, 1:30 TO 3:00 P.M.

Meeting of Accredited Standards Committee S3 on Bioacoustics

to be held jointly with the

U.S. Technical Advisory Group (TAG) Meetings for ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock

T. A. Frank, Chair S3

Pennsylvania State University, Speech and Hearing Clinic, 110 Moore Building, University Park, Pennsylvania 16802

R. F. Burkard, Vice Chair S3

Hearing Research Laboratory, State University of New York at Buffalo, 215 Parker Hall, Buffalo, New York 14214

P. D. Schomer, Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics
U.S. CERL, P.O. Box 4005, Champaign, Illinois 61820

J. Erdreich, Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 108/SC4,
Human Exposure to Mechanical Vibration and Shock
Ostergaard Acoustical Associates, 100 Executive Drive, West Orange, New Jersey 07052

H. E. von Gierke, Vice Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics and ISO/TC 108/SC4,
Human Exposure to Mechanical Vibration and Shock
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, Building 233, Room A149, Gaithersburg, Maryland 20899

Accredited Standards Committee S3 on Bioacoustics. The current status of standards under preparation will be discussed. In addition to those topics of interest, including hearing conservation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years. Open discussion of committee reports is encouraged. The international activities in ISO/TC 43 Acoustics, and IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock, will also be discussed. The Chairs of the U.S. Technical Advisory Groups for ISO/TC 43 (P. D. Schomer), IEC/TC 29 (V. Nedzelnitsky), and ISO/TC 108/SC4 (J. Erdreich) will report on current activities of these international Technical Committees and Subcommittees.

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

4p THU. PM

Meeting of Accredited Standards Committee S1 on Acoustics

to be held jointly with the

U.S. Technical Advisory Group for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics

J. P. Seiler, Chair S1

U. S. Department of Labor, Cochran Mill Road, P.O. Box 18233, Building 038, Pittsburgh, Pennsylvania 15236

G. S. K. Wong, Vice Chair S1

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

P. D. Schomer, Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics

U. S. CERL, P.O. Box 4005, Champaign, Illinois 61820

H. E. von Gierke, Vice Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics

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, Building 233, Room A149, Gaithersburg, Maryland 20899

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on their preparation of standards on methods of measurement and testing, and terminology, in physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Work in progress includes measurement of noise sources, noise dosimeters, integrating sound-level meters, and revision and extension of sound level meter specifications. Open discussion of committee reports is encouraged. The international activities in ISO/TC 43 Acoustics, and IEC/TC 29 Electroacoustics, will also be discussed. The chairs of the respective U.S. Technical Advisory Groups for ISO/TC 43 (P. D. Schomer) and IEC/TC 29 (V. Nedzelnitsky), will report on current activities of these international Technical Committees.

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.