Session 1aAO

Acoustical Oceanography: Acoustical Ocean Monitoring: Determination of Current and Temperature Fields I

David R. Palmer, Cochair
NOAA/AOML, 4301 Rickenbacker Causeway, Miami, Florida 33149

Iwao Nakano, Cochair
Ocean Research Department, Japan Marine Science and Technology Center, 2-15 Natsushima-cho, Yokosuka, Kanagawa, 237 Japan

Chair's Introduction—10:15

Invited Papers

10:20

1aAO1. Monitoring of the transport of the ocean current by an acoustic transceiver array. Tomoyoshi Takeuchi (Univ. of Electro-Communications, 1-5-1, Chofu, Tokyo, 182 Japan) and Keisuke Taira (Univ. of Tokyo 1-15-1, Minamidai, Nakano-Ku, Tokyo, 164 Japan)

Current velocity can be determined by measuring the difference of the sound wave traveling time between two points. The multipaths inverted echo sounder (MIES) [Takeuchi *et al.*, J. Acoust. Soc. Jpn. **49**, 543–550 (1993)] was developed in order to apply this method to the measurement of the volume transport of the Kuroshio. Measurement of mean current velocity can be made with a difference of reciprocal travel times of the sound wave along two sides of a triangle in a vertical plane, which is constructed by the acoustic paths with a base side of the mooring distance when two multipath inverted echosounders were deployed 10 km apart. The measurement of volume transport was attempted by applying the above method, where three multipath inverted echosounders were deployed so as to construct a regular triangle on the sea bottom. In this paper, the result of the measurement of the volume transport of the Kuroshio over Izu Ridge from 20 March to 25 April 1995 is presented.

10:35

1aAO2. Ocean current and vorticity measurements using long-range reciprocal acoustic transmissions. Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA 92093)

Measurements of the sum and difference of the travel times of acoustic pulses propagating in opposite directions provide powerful tomographic tools. Travel-time signals due to sound-speed perturbations cancel in the difference travel time, leaving only the much smaller signals due to currents. Reciprocal acoustic transmissions are particularly well suited to measuring large-scale barotropic flow and areal-average relative vorticity, both of which are of great oceanographic importance, but difficult to measure using other techniques. The use of reciprocal transmissions to measure ocean currents and vorticity has been tested in a series of experiments at steadily increasing ranges, from 25 to 1275 km. Acoustically derived currents and vorticity have been found to be consistent with independent measurements in all cases. The barotropic tides have provided the best test signals, because they are both well known and of large scale. Point measurements provided by current meters provide less stringent tests. The high-frequency travel-time fluctuations due to internal-wave-induced sound-speed perturbations have been found to largely cancel in the differential times, demonstrating that the ray paths of oppositely traveling signals are nearly reciprocal out to 1-Mm ranges, as is implicitly assumed when differential travel times are used to deduce ocean currents.

10:50

1aAO3. Ocean current effects on a low-frequency acoustic field and the feasibility of their use to monitor ocean dynamics. Oleg A. Godin (School of Earth and Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 2Y2, Canada)

Ray-travel-time nonreciprocity has been used in most tomography experiments to determine current velocity in the ocean by acoustic means at distances large compared to ocean depth. Although very successful in deep water, this approach is not applicable in the coastal ocean where ray arrivals are not separable and/or identifiable due to multiple bottom interactions. In this paper, other parameters of the acoustic field, including normal mode travel time, ray and mode horizontal refraction angles, and full field phase, are treated as possible data for the current velocity field. Some qualitative differences between acoustic fields in moving and motionless fluid are indicated and their importance for acoustic monitoring of ocean currents is emphasized. Existing mathematical models of low-frequency underwater sound propagation in a moving ocean are discussed. It is demonstrated that nonreciprocity of various acoustic field variables possess quite a different sensitivity to the flow velocity field and robustness with respect to unavoidable uncertainties in our knowledge of system and environmental parameters. Full-field inversion, based on an appropriate acoustic field variable, is concluded to be a promising technique for current velocity remote sensing in the coastal ocean environments. [Work supported by NSERC and RBRF.]

11:05

1aAO4. Measuring currents with acoustic propagation. David M. Farmer (Inst. of Ocean Sci., P.O. Box 6000, Sidney, BC, Canada)

Water currents refract and otherwise modify acoustic signals providing a basis for their remote detection. In recent years this concept has been explored with high-frequency propagation in the coastal environment, but the concepts have a broader application to ocean measurement. The component of flow resolved along the acoustic path may be detected through changes in effective sound speed. Phase coherent reciprocal transmission allows separation of current-induced effects from scalar contributions due to temperature or salinity, leading to measurements of great accuracy and precision. For turbulent flows this approach has been extended into the inertial subrange, permitting measurement of path averaged dissipation. In contrast, coherence of the refractive variability permits measurement of the component of flow orthogonal to the acoustic path through detection of scintillation drift with an array. Additionally, current advection of the pulse may be detected as a change in horizontal arrival angle. Resolution of both current speed and refractive variability as a function of path position can be achieved by combining multielement source and receiver arrays in a spatial aperture filter. These concepts are now being applied in the Bosphorus to the measurement of exchange with the Black Sea using a two-level reciprocal scintillation array.

11:20

1aAO5. An approach to the coastal ocean acoustic tomography. Arata Kaneko, Hong Zheng (Dept. of Environ. Sci., Faculty of Eng., Hiroshima Univ., Higashi-Hiroshima, 739 Japan), and Hideaki Noguchi (Chugoku Natl. Industrial Res. Inst., Kure, 731-01 Japan)

A reciprocal sound transmission system has been designed for long-term current measurements in the coastal sea with heavy ship traffic and fishing activities. The system was composed of two acoustic stations spaced a distance of 5.7 km on both sides of a channel in the Seto Inland Sea, Japan. Each station was equipped with a transmitter, hydrophone, and GPS receiver. Reciprocal sound transmission experiments between the two stations were successfully completed for 5 h, using a carrier of 11 kHz, modified with the M sequence of 10th order. The time coordinate at both stations was synchronized with the accuracy of 0.1 μs by the 1-Hz and 1-kHz time signals of GPS. Range-averaged current velocities, estimated from the travel time data obtained reciprocally, were in good agreement with the results of the ADCP measurement obtained along the sound transmission line. A 10.6-km sound transmission experiment using the carrier of 11.0 kHz and the M sequence of 10th order was also done successfully in an adjacent channel of the Seto Inland Sea. The present sound transmission system can easily be extended to a coastal tomography system composed of an array of acoustic stations.

Contributed Papers

11:35

1aAO6. Observation of barotropic-tide relative vorticity in the northwest Atlantic. Brian D. Dushaw, Bruce M. Howe (A. P. L., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698), Peter F. Worcester, Bruce D. Cornuelle (Univ. of California, La Jolla, CA 92093-0213), and Kurt Metzger (Univ. of Michigan, Ann Arbor, MI 48109)

Time series of reciprocal ray travel times were obtained at 350-, 410-, and 670-km ranges in the western North Atlantic during the 1991–1992 Acoustic Mid-Ocean Dynamics Experiment (AMODE). Transmissions were recorded for approximately 300 days between six transceivers in a pentagonal array. Barotropic current along each of the 15 propagation paths is derived from the difference of reciprocal ray travel times, while ten independent estimates of areal-averaged relative vorticity are found by integrating current around triangles in the pentagonal array. The estimated tidal currents are highly accurate, and tidal relative vorticity at the M_2 frequency is detected. This vorticity is induced primarily by the stretching of vortex lines by tidal elevation. Harmonic constants (amplitude, phase) of M_2 tidal vorticity are about $(4–8\pm2\times10^{-9}$ s^{-1}, $270°-320°$ $\pm20°)$, while harmonic constants of about $(2–3\times10^{-9}$ s^{-1}, $300°-340°)$ are predicted using the shallow-water equations. The measured tidal harmonic constants are compared with those derived from a global barotropic tidal model.

11:50

1aAO7. Acoustic observations of Mediterranean flow into the Black Sea. Daniela Di Iorio and Tuncay Akal (SACLANT Undersea Res. Ctr., Viale San Bartolomeo 400, 19138 La Spezia, Italy)

The physical behavior of the Mediterranean flow entering the Black Sea through the Bosphorus Strait is described using a variety of high-frequency acoustic systems. Because of the density difference between salty Mediterranean and fresh Black Sea water, a two-layer exchange is formed which is confined within a canyon in the Black Sea exit region of the Bosphorus Strait. A 307-kHz acoustical scintillation system placed 6 m from the seafloor and covering a 300-m propagation path is used to describe the mean Mediterranean current speed and the turbulent velocity fluctuations within the bottom boundary layer of the Mediterranean flow during a 4-day period when the exchange was maximal. In the idealized case of isotropic and homogenous turbulence, estimates of the turbulent kinetic energy dissipation rate leads to values ranging from 1×10^{-6} to 5×10^{-5} W/kg^{-1}. A 600-kHz broadband acoustic Doppler current profiler placed within the canyon shows that the two-layer exchange displays temporal variability over scales of a few days associated with the meteorological conditions in the Black Sea. To help interpret the oceanographic measurements, a 120-kHz high-resolution echo sounder is used to obtain two-dimensional images of the two-layer exchange.

Session 1aNS

Noise: Booms, Jets, Ducts and More

Mary M. Prince, Cochair
Centers for Disease Control and Prevention, National Institute of Occupational Safety and Health, 4676 Columbia Parkway, Cincinnati, Ohio 45226

Ichiro Yamada, Cochair
Kobayashi Institute of Physical Research, 3-20-41 Higashi-motomachi, Kokubunji, Tokyo, 185 Japan

Contributed Papers

10:15

1aNS1. Validation of launch vehicle sonic boom predictions. J. Micah Downing (USAF Armstrong Lab., Wright–Patterson AFB, OH 45433) and Kenneth Plotkin (Wyle Labs., Inc., Arlington, VA 22202)

Concern has been raised about the sonic boom levels generated by launch vehicle operations. To address this concern the USAF recently monitored the sonic boom generated by the launch of a Titan IV rocket and the reaction of the local wildlife. The sonic boom generated by this launch intersected the Channel Islands off the coast of Southern California. The sonic booms were recorded by the USAF Boom Event Analyzer Recorder and by DATs with low-frequency microphones on San Miguel, Santa Rosa, and Santa Cruz islands. Before the launch, predictions were made to assess the sonic boom impact on the islands. The prediction model used in this comparison was PCBoom3, a single-event sonic boom model developed by the USAF. This model has been updated to model sonic booms generated by launch vehicles and includes the effect of the rocket plume on the generation of the sonic boom. The measured sonic boom signatures are compared to predictions.

10:30

1aNS2. Sonic-boom noise penetration into the ocean: 1996 update. Victor W. Sparrow, Judith L. Rochat, and Tracie J. Ferguson (Grad. Prog. Acoust., Penn State, 157 Hammond Bldg., University Park, PA 16802)

Recently, there has been substantial progress in predicting the penetration of sonic boom noise into the ocean. Such noise penetration occurs for either commercial or military supersonic aircraft operating over the ocean. As previously discussed [J. Acoust. Soc. Am. **97**, 3258(A) (1995)], one can use analytical techniques to make predictions of the noise penetration, but eventually finite difference calculations become the method of choice when accounting for wind wave swell and ocean inhomogeneities. Two-dimensional calculations already indicate that typical wind wave swell can focus or defocus a penetrating rounded sonic boom waveform up to approximately ± 1.5 dB for a homogeneous ocean below a homogeneous atmosphere. Higher Mach number supersonic flight accentuates the focusing. Predictions are currently being sought for more realistic sonic boom waveform shapes, three-dimensional interactions between the incident boom and the ocean swell, and the effects of bubble plumes immediately below the ocean surface. One simulation of the focusing from ocean swell will be visualized via a videotape. [Work supported by NASA Langley Research Center, under Grant NAG 1–1638, and by Armstrong Laboratory, Air Force Material Command, USAF, under Grant F41624–96–1-0003.]

10:45

1aNS3. Reduction of noise radiated from supersonic jets. Yoshikuni Umeda and Ryuji Ishii (Div. of Aeronaut. and Astronautics, Dept. of Eng. Sci., Kyoto Univ., Yoshida Hon-Machi, Sakyo-ku, Kyoto, 606 Japan)

In the present investigation, sound-pressure levels radiated from the twin- and multijet with a square configuration were measured at various pressure ratios R from 2.00 to 6.33. The nozzle diameter was 5 mm and the center-to-center spacing of the nozzles was fixed to 1.4 times the nozzle diameter. From this experiment, it is found that the overall sound-pressure level (OASPL) radiated from the multijet with a square configuration becomes lower than that from the twin jet for the pressure ratio range $R > 4.0$. In the near sound field upstream of the nozzle exit, the difference of the OASPL between the twin- and multijet with a square configuration reaches a maximum (about 7 dB) at the pressure ratio about $R=5.0$. This decreasing of the OASPL may be caused by the shielding of the sound waves radiated from the inner part of the multijet with a square configuration by the surrounding four jets, and by stabilizing the jet oscillation due to the appropriate coupling process among four jets. [Work supported by the Ministry of Education and Culture of Japan (Grant No. C2-08650201).]

11:00

1aNS4. Exact solutions for sound radiation from a circular duct with flows. Y. C. Cho (NASA Ames Res. Ctr., MS 269-3, Moffett Field, CA 94035-1000)

Sound radiation from a circular duct is a classic problem. Exact solutions were previously reported for the case of a negligibly thin duct wall, using the Wiener–Hopf technique. Despite the elegance of its closed-form solutions, the numerical presentations have been limited to mere demonstrations of its capability. A computer program is not publicly available for its numerical evaluation. Such a numerical evaluation has recently become increasingly in demand not only for its own merits, but also for cross examination of the numerical results of computational aeroacoustics. These techniques are just starting to attract widespread attention as a potential tool in attacking important but unsolved aeroacoustic problems. This paper presents a comprehensive mathematical procedure for the numerical evaluation primarily for use in studies of noise emission of aircraft engines. Various mean flows are included for simulations of an aircraft jet engine exhaust and inlet, and aircraft cruise condition. Unlike previously published reports, this paper will include radiation of nonpropagating modes as well as propagating modes.

11:15

1aNS5. Lord Rayleigh revisited: Can vortices in tubes be sources of noise? James B. Lee (Concert Acoustics, P. O. Box 80571, Portland, OR 97205)

Energy can flow through tubes filled with air governed by the laws of fluid dynamics, even in the violent form of shock waves. Energy also can flow through such systems as oscillations of small amplitude, which are governed by the equations of linear acoustics. What happens in the regime

between? Calculations performed upon complaint of excessive low-frequency noise during alternate blasting for two long parallel railway tunnels indicated that particle velocities were a substantial fraction of the speed of sound at the tunnels' mouths—between the regimes of shock waves and linear resonances. A Strouhal calculation suggested that vortices were the likely source: A sequence of counter-rotating vortices was being propelled out of the tunnels' mouths, generating a powerful monopole source at 12 Hz. Modern noise consultants were baffled, but Lord Rayleigh had solved the system a century before, as a compound nonlinear eigenvalue problem, necessarily taking viscosity into account. The blasting engineers drove small cross-bores between the tunnels, sucking circulating fluid out of the vortices, reducing power, and quieting neighbors. The experiment should be repeated at a smaller scale with better instruments under controlled conditions.

11:30

1aNS6. Sound energy distribution at an intersection of an underground tunnel: Additional small-scale experiments. Hiroyuki Imaizumi, Sunao Kunimatsu, and Takehiro Isei (Safety Eng. Dept., Natl. Inst. for Resources and Environment, 16-3 Onogawa, Tsukuba, Ibaraki, 305 Japan)

In studies on some interactions between sound propagation characteristics and environmental factors underground, small-scale experiments on the sound energy distribution at an intersection of tunnels have been carried out in an anechoic room. The small-scale tunnels were made by acrylic plates, and the original surfaces of inner walls were assumed to have an acoustically hard condition, while surfaces of the walls covered by flannel were acoustically soft. Impulsive sound was generated electrically as a sound source was applied, and propagated sounds were measured by an omnidirectional microphone. The experiments were carried out under several kinds of conditions for angles at the intersection and positions of the sound source. Influences of intersection on the sound energy distribution were indicated by comparisons between the measuring points before and behind the intersection with different angles. In addition, the influences of the acoustical characteristics of the inner walls and positions of the source are also presented.

11:45

1aNS7. Effects of perforated pipe on the higher-order modes in an elliptical chamber. Tatsuyu Ikeda, Tsuyoshi Nishimura (Kumamoto Inst. of Technol., 4-22-1 Ikeda, Kumamoto, 860 Japan), Tsuyoshi Usagawa, and Masanao Ebata (Kumamoto Univ., Kumamoto, 860 Japan)

The role of the perforated pipe in a simple elliptical expansion chamber, especially its ability to improve noise reduction by decreasing the level of resonance caused by traverse waves, is presented. The characteristics of the perforated pipe have been studied by means of impedance which depends on the shape and porosity of a pipe. The following phenomena regarding sound-pressure distribution when the perforated pipe is located at the center of a simple elliptical expansion chamber have been observed. (1) The resonance frequencies of higher-order modes shifted to the lower frequency ranges in comparison with those when a pipe was not attached, and (2) the resonance frequencies inside a perforated pipe are similar to the ones outside; however, the resonance level inside is higher than the one outside. In addition, an arrangement of the output pipe is also studied based on the distribution of higher-order modes in a chamber. The experimental results which tend to prove the effectiveness of the perforated pipe are concretized by a mathematical formula in the paper.

MONDAY MORNING, 2 DECEMBER 1996 — WAIALUA ROOM, 10:15 TO 11:45 A.M.

Session 1aPA

Physical Acoustics: Nonlinear Acoustics I: Propagation in Solids

James A. TenCate, Cochair
Los Alamos National Laboratory, EES-4, MS D443, Los Alamos, New Mexico 87545

Akira Nakamura, Cochair
Department of Electrical Engineering, Fukui Institute of Technology, 3-6-1 Gakuen, Fukui, 910 Japan

Contributed Papers

10:15

1aPA1. Nonlinear surface wave propagation in a piezoelectric material. M. F. Hamilton, Yu. A. Il'inskii, and E. A. Zabolotskaya (Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712-1063)

Model equations were derived from first principles for nonlinear surface wave propagation in a piezoelectric material. There is no dependence on ad hoc or empirical parameters. The present work extends an earlier analysis of surface waves in crystals, described by the authors at the previous meeting [J. Acoust. Soc. Am. **99**, 2538(A) (1996)]. As in the earlier work, the model equations account for crystals having arbitrary symmetry, and for surface wave propagation in arbitrary directions in planes having arbitrary orientations with respect to the crystalographic axes. Here, elastic, piezoelectric, dielectric, and electrostrictive properties of the material are taken into account. The model is formulated as spectral evolution equations that are integrated numerically to illustrate the distortion of a finite amplitude surface wave that is sinusoidal at the source. The material we considered is $LiNbO_3$, for which measured values of all required second- and third-order elastic constants are available in the literature [e.g., the latter are reported by Cho and Yamanouchi, J. Appl. Phys. **61**, 875 (1987)]. Analysis of the nonlinearity matrix permits identification of which physical effects contribute most to the distortion of nonlinear surface waves. [Work supported by ONR, NSF, and Schlumberger Foundation.]

10:30

1aPA2. Second harmonic generation in a sound beam transmitted through an isotropic solid. B. J. Landsberger, M. F. Hamilton, Yu. A. Il'inskii, and E. A. Zabolotskaya (Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712-1063)

An ultrasonic immersion procedure for determining third-order elastic coefficients of rock via measurements of harmonic generation was described recently by Plona *et al.* [J. Acoust. Soc. Am. **98**, 2886(A) (1995)]. Here an accurate quasilinear model is presented for second harmonic generation in a sound beam transmitted through an isotropic solid immersed in liquid. With the primary beam represented as an angular spectrum, analytic solutions were derived for second harmonic generation by all pairs of plane waves in both the liquid and the solid. Numerical superposition of the analytic solutions, followed by Fourier transformation, yields the second

harmonic field anywhere in the solid or liquid. There are no restrictions on geometry or orientation of the sound source, and all combinations of compression and shear wave interactions are taken into account. At the previous meeting the accuracy of this model for the case of reflection near the Rayleigh angle was demonstrated [J. Acoust. Soc. Am. **99**, 2538(A) (1996)]. Reported here are comparisons with second harmonic diffraction patterns that were measured in a 1-MHz beam radiated by a 1.2-cm radius source and transmitted through a 10-cm-thick block of lucite immersed in water. [Work supported by ONR, NSF, and Schlumberger Foundation.]

10:45

1aPA3. Comparative measurement of second-harmonic generation in various materials. Xiaoli Hou and Corinne Darvennes (Dept. of Mech. Eng., Tennessee Tech. Univ., Box 5014, Cookeville, TN 38505)

Second-harmonic generation was measured in several man-made materials for possible application of nonlinear properties to nondestructive testing. Samples included several thicknesses of two types of polymer matrix composites, three types of concretes, and plywood. Steel and aluminum specimens were used as references and one of the composite samples was evaluated before and after fatigue cycles. A monochromatic ultrasonic signal was sent into each sample via a contact transducer placed on its top surface. The growth of the second harmonic was recorded, with a second transducer placed on the bottom face, as the amplitude of the input signal was gradually increased and for several values of the input frequency. Nonlinearity parameters could not be measured, due to the limitations of our equipment. Nonetheless, some interesting observations were made: (1) the two composites were much more nonlinear than the metals; (2) the concretes and the wood were extremely absorptive and an output signal was observed only at the lowest input frequency; and (3) fatigue cycles significantly increased the second harmonic, even though no damage was observed by C-scanning. [Work supported by NSF, TTU Manufacturing Center, and the FRG Program.]

11:00

1aPA4. Observations of nonlinearity with slow dynamics in rocks. James A. TenCate, Thomas J. Shankland (EES-4 MS D443, Los Alamos Natl. Lab., Los Alamos, NM 87545), Paul A. Johnson (Los Alamos Natl. Lab. and Univ. Pierre et Marie Curie, 75252 Paris Cedex 05, France), and Bernard Zinszner (Inst. Français du Pétrole, Rueil Malmaison Cedex, France)

A typical resonance curve—measured acceleration versus drive frequency—made on a thin bar of rock shows peak bending with a softening (nonlinear) modulus as drive levels are increased. Previous work showed the shapes of these nonlinear resonance curves depend on sweep rate, i.e., the ''slow dynamics.'' Slow dynamics in a 0.3-m-long, 50-mm-diam bar of Berea sandstone under ambient conditions have been documented for the first time. Peak strain levels during the experiments ranged from 10^{-11} to 10^{-5} at a fundamental bar resonance frequency near 4 kHz. Slow dynamics begin to appear at strain amplitudes above 10^{-6} at ambient conditions and at the onset of nonlinear peak bending. Higher strains condition the rock, altering its response for minutes to hours after the drive has been turned off. Other rocks show similar results. Physical origins of the slow dynamics lie in nonlinear effects at the microstructural level of cracks, pores, and interstitial clays. Further work examines environmental effects on conditioning and recovery as a means of relating them to physical properties and microtexture of the rock. [Work supported by OBES/DOE through the University of California.]

11:15

1aPA5. Modeling nonlinearity and slow dynamics for rock in resonance experiments. Koen E. A. Van Den Abeele (EES-4 MS D443, Los Alamos Natl. Lab., Los Alamos, NM 87545) and Robert A. Guyer (Univ. of Massachusetts, Amherst, MA 01003)

The presence of compliant features in rock causes nonlinear distortion of sound waves propagating through a sample, even at microstrain levels. When performing resonance experiments, one observes nonlinear behavior in different ways: resonant frequency shifts as function of drive amplitude, nonlinear dissipation including hysteresis, asymmetry of the resonance curves, and rich harmonic spectra. Recently, a new interesting feature has been added to these nonlinearity observations: the existence of a slow dynamics conditioning and recovering characteristic. Experimental evidence will be shown in a companion ASA contribution by TenCate *et al.* The focus is on the mathematical modeling of all of the nonlinear resonance observations, including the slow dynamics characteristics. The model uses a finite-difference time-domain solution of the one-dimensional wave equation with appropriate boundary conditions, and calculates waveforms along the sample. The key part of the model is the description of a modulus which depends on higher-order elastic constants, hysteresis strength, and its own weighted response over previous times. By integrating the history of the modulus using an exponential weighting function, interesting conditioning and recovering effects can be simulated which agree well with the nonlinear and slow time constant observations in rock. [Work supported by DOE/OBES/UCal.]

11:30

1aPA6. Theory and feasibility of breather solitons in sandstone. Miguel Bernard and Bruce Denardo (Natl. Ctr. for Physical Acoust. and Dept. of Phys. and Astron., Univ. of Mississippi, University, MS 38677)

A theoretical model and an experimental feasibility study of nonlinear Schrödinger breather solitons in a sandstone waveguide are presented. These solitons are acoustic waves that are theoretically characterized by a standing wave motion in the transverse direction, an exponential self-localization along the waveguide, and a speed of propagation that can have any value that is small compared to the speed of sound. The sole requirement in the model is that transverse standing waves *soften* (the resonance frequency decreases as the amplitude is increased). The softening of a Berea sandstone sample and the resonance frequency quality factors have been measured in the range 500 Hz–15 kHz. With the use of a composite transducer designed to optimize the drive amplitudes, frequency shifts of 7% have been measured. Comparison of this shift with that due to nonuniformities and estimation of the soliton length and decay distance reveal that the observation of the solitons is feasible in Berea sandstone. This is a first step toward the observation of nonlinear Schrödinger acoustic breather solitons in a sandstone core.

Session 1aSC

Speech Communication: Lexical Factors in the Use of Spoken Language (Poster Session)

Lynne E. Bernstein, Cochair
Spoken Language Processing Laboratory, Department of Human Communication Sciences and Devices, House Ear Institute, 2100 West Third Street, Los Angeles, California 90057

Yasuo Ariki, Cochair
Department of Electronics and Informatics, Ryukoku University, Seta, Otsu, Shiga, 520-21 Japan

Contributed Papers

All posters will be on display from 10:15 a.m. to 12:15 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 10:15 a.m. to 11:15 a.m. and contributors of even-numbered papers will be at their posters from 11:15 a.m. to 12:15 p.m.

1aSC1. The role of within-category structure in the integration of auditory and visual speech. Michael D. Hall, Paula M. T. Smeele, and Patricia K. Kuhl (Dept. of Speech and Hearing Sci., CHDD, Box 357920, Univ. of Washington, Seattle, WA 98195-7920)

The influence of visual information on auditory speech perception can be observed under conditions where the two sources of information are discrepant. One demonstration involves viewing a face producing /b/ while listening to a dubbed /g/ token, with participants reporting that they heard /bg/ (McGurk and MacDonald, 1976). This ''combination'' response reflects the contribution of both modalities. Two experiments evaluated whether differences in the perceived quality of auditory stimuli within the /g/ category influence the incidence of combination responses. Synthetic VCV stimuli ranging from ''good'' /aga/ to ''poor'' /aga/ tokens were generated by factorially combining 6 levels of $F2$, and 4 levels of $F3$, onset frequency. In experiment 1 participants identified these auditory stimuli and rated them with respect to goodness as a /g/. Goodness was found to be correlated with, but not completely predicted by, consonant identification. In experiment 2 these stimuli were separately dubbed with a visual /aga/ (''matched'') and /aba/ (''mismatched,'' which should evoke combination responses). Results will be discussed in terms of the sufficiency of consonant identification and category goodness in predicting the probability of combination responses. These data will be used to address models of auditory-visual speech integration. [Work supported by NICHD.]

1aSC2. Audiovisual integration of speech based on minimal visual information. D. H. Whalen (Haskins Labs., 270 Crown St., New Haven, CT 06511), Julia Irwin (Haskins Labs., New Haven, CT 06511 and Univ. of Connecticut), and Carol A. Fowler (Haskins Labs., New Haven, CT 06511, Yale Univ., and Univ. of Connecticut)

Two competing theories have been proposed to explain the fact that vision can dominate over audition in syllables that have been spliced so that the two modalities specify different phonemes [McGurk and McDonald, Nature **263**, 746–748 (1976)]. The first theory states that acoustic and visual information are combined in varying proportions depending on how strong the information is in each signal. The second proposes that the visual signal has linguistic value because speech gestures can be conveyed visually, and that these gestures are the primitives of speech perception for every modality. The present experiment contrasts dynamic and static visual information by reducing the visual signal to two or three video frames, synchronized with the speech in the appropriate location. Dynamic stimuli had at least two frames showing movement of the mouth, while static ones had a single frame, taken from the consonant closure, repeated to make a three frame visual image. Even these brief images were enough to elicit speech percepts that matched the visual image. The dynamic and static images were equally effective, suggesting revisions in both theories. [Work supported by NIH Grant No. HD-01994.]

1aSC3. Relationships between word knowledge and visual speech perception. I. Subjective estimates of word age of acquisition. Edward T. Auer, Jr., Robin S. Waldstein, Paula E. Tucker, and Lynne E. Bernstein (Spoken Lang. Processes Lab., Human Commun. Sci. and Devices Dept., House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057)

In individuals with normal hearing, words estimated to be learned earlier are recognized more rapidly than words estimated to be learned later. To investigate how word knowledge is related to lipreading proficiency, word age-of-acquisition (AOA) estimates were obtained from 50 hearing (H) and 50 deaf (D) (80-dB HL pure-tone average or greater hearing losses acquired before the age of 48 months) adults. Participants judged AOA for the 175 words in Form M of the Peabody Picture Vocabulary Test-Revised using an 11-point scale, and responded whether the words were acquired through speech, sign language, or orthography. The two groups differed in when (mean AOA: H = 8.9 years, D = 10.6 years) and how (H=69% speech and 31% orthography; D=38% speech, 45% orthography, and 17% sign language) words were judged to be acquired. However, item analyses revealed that the relative acquisition order was essentially identical across groups ($r=0.965$). Interestingly, within the deaf group, better lipreaders estimated that more words had been learned through speech than orthography. An implication of these results is that learning words primarily through orthography does not support highly accurate spoken language processing. [Work supported by NIH Grant No. DC00695.]

1aSC4. Relationships between word knowledge and visual speech perception. II. Subjective ratings of word familiarity. Robin S. Waldstein, Edward T. Auer, Jr., Paula E. Tucker, and Lynne E. Bernstein (Human Commun. Sci. and Devices Dept., House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057)

Word familiarity is an important factor in word recognition and lexical access for hearing individuals. Subjective word familiarity ratings are hypothesized to reflect experience with words irrespective of the modality

(i.e., spoken or written) through which exposure has taken place, and to provide an estimate of the size of the mental lexicon. To investigate how word familiarity is related to lipreading proficiency, 450 printed words were presented for rating on a seven-point scale to 50 deaf and 50 hearing participants. Preliminary results revealed that the deaf participants produced lower mean familiarity ratings than did the hearing participants, for high-, medium-, and low-familiarity words (hearing means = 6.7, 4.8, 3.0; deaf means = 6.0, 3.8, 2.6). Among the deaf participants, correlations between established familiarity ratings and individuals' ratings were reliably higher for excellent than for good lipreaders, a possible indication that perceptual experience influences the structure of the lexicon. At the same time, the performance of the excellent lipreaders provides support for the hypothesis that lexical organization does not depend on the perceptual input modality (i.e., vision versus hearing). [Work supported by NIH Grant No. DC00695.]

1aSC5. Speechreading talking faces. Dominic W. Massaro, Christopher S. Campbell, Michael A. Berger, and Michael M. Cohen (Dept. of Psych., Univ. of California, Santa Cruz, CA 95064)

It is now well established that visible speech is an important source of information in face-to-face communication. Given the valuable role of audible speech synthesis in experimental, theoretical, and applied arenas, visible speech synthesis has been developed. Research has shown that the talking head closely resembles real heads in the quality of its speech and its realism (when texture mapping is used). The talking head can be heard, communicates paralinguistic as well as linguistic information, and is controlled by a text-to-speech system. Several sources of evidence are presented which show that visible speech perception (speechreading) is fairly robust across various forms of degradation. Speechreading remains fairly accurate even when the mouth is viewed in noncentral vision; eliminating and distorting high-spatial frequency information does not completely disrupt speechreading; and speechreading is possible when additional visual information is simultaneously being used to recognize the speech input. The results are consistent with the fuzzy logical model of perception in which multiple sources of information are used to recognize patterns. Various visible feature sets are tested within the framework of the model to determine which visible features are functional in speechreading. Demonstrations of the talking head and various psychological phenomena will be provided. [Work supported by NIDCD.]

1aSC6. Sensitivity of cued speech reception to cue imperfections. Maroula S. Bratakos and Louis D. Braida (Res. Lab. of Electron., MIT, Cambridge, MA 02139)

In manual cued speech (MCS) a speaker gestures with his/her hand to resolve ambiguities among speech elements that are often confused by speechreaders. The shape of the hand distinguishes among consonants, and the position of the hand relative to the face distinguishes among vowels. Experienced receivers of MCS achieve nearly perfect reception of everyday connected speech. To understand the benefits that might be derived from the imperfect cues produced by an automatic cueing system, videotaped sentences with handshapes corresponding to the phones identified by simulated phonetic speech recognizers were dubbed. The cues dubbed on these sentences were discrete in both shape and position rather than fluidly articulated, and the speaking rate was roughly 50% faster than for MCS. When the phones identified by an ideal recognizer were used to produce the cues, performance was only slightly lower than for MCS. When cues were derived from an existing recognizer, intelligibility was reduced, but substantial benefits to speechreading were observed. Current research is aimed at developing an automatic speech recognition system with the speed, accuracy, and computational efficiency required for a real-time automatic cueing system. [Work supported by NIH.]

1aSC7. Hemispheric differences in perceiving and integrating dynamic visual speech information. Jennifer A. Johnson and Lawrence D. Rosenblum (Dept. of Psych., Univ. of California, Riverside, CA 92521)

There is evidence for a left-visual-field/right-hemisphere (LVF/RH) advantage for speechreading static faces [R. Campbell, Brain & Cognit. **5**, 1–21 (1986)] and a right-visual-field/left-hemisphere (RVF/LH) advantage for speechreading dynamic faces [P. M. Smeele, NATO ASI Workshop (1995)]. However, there is also evidence for a LVF/RH advantage when integrating dynamic visual speech with auditory speech [e.g., E. Diesch, Q. J. Exp. Psychol.: Human Exp. Psychol. **48**, 320–333 (1995)]. To test relative hemispheric differences and the role of dynamic information, static, dynamic, and point-light visual speech stimuli were implemented for both speechreading and audio–visual integration tasks. Point-light stimuli are thought to retain only dynamic visual speech information [L. D. Rosenblum and H. M. Saldaña, J. Exp. Psychol.: Human Percept. Perform. **22**, 318–331 (1996)]. For both the speechreading and audio–visual integration tasks, a LVF/RH advantage was observed for the static stimuli, and a RVF/LH advantage was found for the dynamic and point-light stimuli. In addition, the relative RVF/LH advantage was greater with the point-light stimuli implicating greater relative LH involvement for dynamic speech information.

1aSC8. Face identification using visual speech information. Deborah A. Yakel and Lawrence D. Rosenblum (Dept. of Psych., Univ. of California, Riverside, CA 92521)

Traditionally, the recovery of linguistic message and speaker identity is thought to involve distinct operations and information. However, recent observations with auditory speech show a contingency of speech perception on speaker identification/familiarity [e.g., Nygaard *et al.*, Psychol. Sci. **5**, 42–46 (1994)]. Remez and his colleagues [Remez *et al.*, J. Exp. Psychol. (in press)] have provided evidence that these contingencies could be based on the use of common phonetic information for both operations. In order to examine whether common information might also be useful for face and visual speech recovery, point-like visual speech stimuli were implemented which provide phonetic information without containing facial features [L. D. Rosenblum and H. M. Saldaña, J. Exp. Psychol.: Human Percept. Perform. **22**, 318–331 (1996)]. A 2AFC procedure was used to determine if observers could match speaking point-light faces to the same fully illuminated speaking face. Results revealed that dynamic point-light displays afforded high face matching accuracy which was significantly greater than accuracy with frozen point-light displays. These results suggest that dynamic speech information can be used for both visual speech and face recognition.

1aSC9. The effect of utterance duration on visual-speech intelligibility scores. Jean-Pierre Gagne and Lina Boutin (Ecole d'orthophonie et d'audiologie, Univ. de Montreal, Montreal, PQ H3C 3J7, Canada)

The effect of duration on the visual-speech intelligibility of talkers was investigated. The stimulus set consisted of 25 sentences. Each sentence had the same grammatical structure and contained three critical elements (subject, verb, object). Seven talkers were videotaped while they spoke a list of sentences twice using conversational and clear speech. For each talker, the mean duration of the conversational and clear sentences were measured. The recordings were digitized and dubbed under four conditions: (1) normal conversational; (2) conversational speech decelerated to a speed equivalent to the duration of the talker's average utterances of clear speech; (3) natural clear speech; (4) clear speech accelerated to a speed equivalent to the talker's average conversational speech. The test sentences (25 sentences $\times$ 2 iterations $\times$ 4 durations $\times$ 7 talkers) were randomized and shown (without sound) to a group of 18 subjects. The responses obtained during the perceptual task were used to determine each talker's speech intelligibility for the four experimental conditions. The results revealed a significant effect for sentence duration and talker, as well as a significant sentence-duration x talker interaction. The effects of sentence duration on visual-speech intelligibility will be discussed. [Work supported by NSERC.]

1aSC10. Cross-modal context effects on the perception of /r/ and /l/ in a speech and nonspeech mode. Kerry P. Green and Linda W. Norrix (Ctr. for Neurogenic and Commun. Disord., Univ. of Arizona, Tucson, AZ 85721)

Norrix and Green [J. Acoust. Soc. Am. **99**, 2591–2592 (1996)] provided evidence for cross-modal context effects on the perception of /r/ and /l/ in a stop cluster. Tokens from a synthetic /iri–ili/ continuum were dubbed onto a visual /ibi/. When presented in an auditory-visual (AV) condition, the tokens were perceived as ranging from /ibri/ to /ibli/. Results indicated a reliable shift in the AV condition relative to an auditory-only (AO) condition. This shift was in accord with acoustic consequences of articulating /r/ and /l/ in a stop cluster. In the current study, sine-wave analogs of the /iri–ili/ tokens were constructed and presented to two groups of observers in an AO and AV condition. Group One was told they would hear schematic speech sounds and instructed to identify what they heard as /r/ or /l/. Group Two made up their own criteria for classifying the tokens as nonspeech sounds. Results indicated a reliable shift in the /r–l/ boundary between the AO and AV conditions for the speech group only and suggest that the influence of the visual articulatory context depends upon listeners interpreting the auditory tokens as speech. [Work supported by NIDCD, NIH.]

1aSC11. Effect of sign complexity on speech timing in simultaneous communication. Robert L. Whitehead (Appl. Lang. and Cognition Res., Natl. Tech. Inst. for the Deaf, 52 Lomb Memorial Dr., Rochester, NY 14623-5604), Nicholas Schiavetti (State Univ. of New York, Geneseo, NY 14454), Brenda Whitehead (Natl. Tech. Inst. for the Deaf, Rochester, NY 14623), and Dale Evan Metz (State Univ. of New York, Geneseo, NY 14454)

Simultaneous communication combines spoken English with manual representations of English words by signs and fingerspelling. The purpose of this investigation was to study the effect of sign complexity on temporal features of speech during simultaneous communication (SC). The effects of three independent variables: (a) mode (speech only versus SC); (b) sign complexity (base versus elaborated signs); and (c) type of sign movement (kinetic versus morphokinetic) were studied on five dependent variables: (a) word duration, (b) sentence duration, (c) diphthong duration, (d) interword-interval before signed experimental word (IWIB), and (e) interword-interval after signed experimental word (IWIA). Audio recordings were made of 12 normal-hearing, experienced sign language users speaking experimental words that varied in sign complexity and movement under SC and speech only (SO) conditions. Results indicated longer sentence durations for SC than SO and longer anticipatory durations of IWIB and diphthong before signed words, especially those using more complex signs. IWIA only lengthened for SC vs SO with no further effect of sign complexity. These results indicate a finite effect of sign complexity on pause and segment durations before the sign but not as strong an effect as has been reported for increased fingerspelling complexity.

1aSC12. Synthesizing audiovisual speech from physiological signals. Eric Vatikiotis-Bateson and Hani Yehia (ATR Human Information Processing Res. Labs., 2-2 Hikaridai, Seika-cho, Kyoto, 619-02 Japan)

Previous examination of perceiver eye movement behavior during audiovisual speech tasks has shown that linguistically relevant visual information is distributed over large regions of the face [Vatikiotis-Bateson *et al.*, ICSLP-94 (1994)]. Furthermore, the simultaneous production of facial and vocal tract deformations suggests a single source of control for acoustic and visual components of speech production. To further examine this possibility, orofacial motion during speech has been correlated with perioral muscle activity, the time-synchronous behavior of vocal tract articulators, and elements of the speech acoustics (e.g., rms amplitude) using both linear and nonlinear modeling techniques. Not surprisingly, since small motions require small forces, linear techniques such as minimum mean square error and second order autoregression provide reasonably good estimates of the inherently nonlinear mapping between muscle EMG and orofacial kinematics. This paper assesses the relative merits of such linear models versus nonlinear, neural network estimations of the orofacial dynamics.

1aSC13. Neural processes of audio–visual speech perception. Satoshi Imaizumi, Koichi Mori, Shigeru Kiritani, Masato Yumoto (RILP, Univ. Tokyo, Bunkyo-ku, Tokyo, 113 Japan), and Hideaki Seki (Chiba Inst. Tech., Japan)

Neural processes related to audio–visual speech perception were investigated by measuring the mismatch magnetic fields (MMF), which reflect a neural activity detecting deviant stimuli randomly inserted in a stream of rapidly repeating frequent stimuli. Three audio–visual stimuli were used: AbVg (audio signal /ba/ with discrepant visual signal /ga/), AbVb, and AdVd. MMF were measured from the left hemisphere of 16 normal hearing subjects using a 37 ch SQUID magnetometer. The rate of non-/ba/ responses to AbVg (McGurk fusion effect) varied depending on the stimulus condition and subjects. It was 80% when AbVg was the frequent, but was 45% when AbVg was the deviant. Significant MMF-like fields were excited in the auditory cortex by deviant AbVg embedded in frequent AbVb. The subjects with a low-fusion rate had larger MMF-like fields than those with a high rate. These results suggest that the auditory mismatch detection process is affected by visual signal, and phonetic categorization is affected by a module which can either fuse or dissociate audio–visual information.

1aSC14. Lexical access in French: Recognizing "identical" phrases. Robert Bannert, Pascale Nicolas, and Monika Stridfeldt (Umeå Univ., Dept. of Phonet., S-901 87 Umeå, Sweden)

Information on word recognition and lexical access has been dominated by research on English. In French, with its different prosodic coding, showing the phonological processes of *liaison*, *enchaînement*, and *e-deletion*, represents an opportunity to widen the understanding of speech recognition. A listening test was carried out containing 20 utterances and ten distractors ranging from one to six syllables and forming seven pairs and two triplets of supposedly "identical" linguistic phrases. A representative sample of each type produced by a male and a female French speaker was presented five times in random order to 18 native listeners of French (including the two speakers) and nine Swedish learners of French as a foreign language. No group identified any of the original utterances correctly, and only two pairs of stimuli were identified at random. Taking into account the frequencies of the stimuli, it appears that most stimuli contain some acoustic information that guides recognition. An acoustic analysis of these utterances showed prosodic and, what might be considered unexpected, according to the literature, segmental differences in spectrum amplitude and $F0$. [Work supported by the Swedish Council for Research in the Humanities and Social Sciences.]

1aSC15. "SLIP-ing" in phonologically similar neighborhoods. Michael S. Vitevitch (Dept. of Psychol., Park Hall, SUNY, Buffalo, NY 14260-4110)

Phonological speech errors are the class of speech errors in which one or more phonemes are added, deleted, substituted, or reversed. Previous work [Dell (1990); Stemberger and MacWhinney (1986)] has found that low-frequency words are more likely to be involved in phonological speech errors than high-frequency ones. If word frequency, influences performance in speech production, do other characteristics of words, such as neighborhood density or neighborhood frequency, influence speech production? The influence of phonologically similar neighborhoods on a type of whole word speech error (malapropisms) has previously been demonstrated [Vitevitch (1996)]. The present research examined the influence of phonologically similar neighborhoods on phonological speech errors. Using the SLIPS technique [Motley and Baars (1976)], phonological speech errors were elicited from participants with real word pairs that varied in their neighborhood characteristics. The results show that low-frequency words ''slipped'' more often than high-frequency ones, replicating past results from error corpora and from other studies using this methodology. Moreover, words with sparse phonologically similar neighborhoods tended to ''slip'' more than words with dense phonologically similar neighborhoods. The implications for these findings on models of speech production, speech perception, and lexical representation are discussed.

1aSC16. 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., State Univ. of New York, Park Hall, Buffalo, NY 14260-4110)

The role of phonotactic probability in the segmentation of spoken words in continuous speech was investigated. 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 transition type (CC, CV, VC, and VV). Both intraword co-occurrence probabilities and transitional probabilities were manipulated. The implications of the results for the use of phonotactic probabilities in the identification of words in fluent speech will be discussed.

1aSC17. Frequency effects in malpropisms. William Raymond and Alan Bell (Dept. of Linguist., Univ. of Colorado, Boulder, CO 80309)

Malapropisms are lexical substitution errors resulting from a failure at the stage of accessing the phonological form corresponding to a lemma—a semantically and structurally specified lexical entry [Fay and Cutler (1977); Garrett (1980)]. An example is *tentative* for *tenable*. Since they occur naturally in utterance contexts, malapropisms are an important source of information about this stage of lexical access. A study of over 300 malapropisms confirmed prior findings that errors resemble targets closely in phonological form and in syntactic category. In addition, it was found that errors resemble targets in derivational morphology (independently of phonological similarity), and that there is a strong correlation ($r^2=0.34$) of the text frequencies of the error and target words. The partial correlations of frequency with word length, syntactic category, and segmental similarity only account for one-half of this correlation (residual r^2 about 0.17). Models which treat frequency effects as biases on the decision processes of selecting word forms [Luce *et al.* (1990)] or as form activation thresholds [Jescheniak *et al.* (1994)] cannot account for such a correlation directly. The results thus appear to support some degree of organization of the lexicon according to frequency, as suggested, e.g., by Forster (1976).

1aSC18. The role of time during lexical access. Arthur G. Samuel (Dept. of Psych., SUNY, Stony Brook, NY 11794-2500)

Speech naturally occurs over time—words unfold from ''left-to-right.'' This structure inherently confounds two logically distinct factors: the amount of phonetic information presented, and the amount of time the perceptual process has been working. For example, in the word ''acoustical,'' both the elapsed processing time and the number of processed phonemes (or syllables) would be greater for the second /k/ (''c'') than for the first one. The current study attempts to unconfound these two factors by using speech compression and expansion techniques, coupled with phonemic restoration methodology. The strength of phonemic restoration can be used as an index of the strength of lexical activation. For example, restoration is stronger for word-final phonemes than for phonemes in earlier word positions, due to the higher lexical activation later in words. The current study tests phonemic restoration when available processing time is manipulated via speech compression/expansion. An interesting additional variable that appears to interact with time is the active lexical cohort size: Additional processing time seems to be more useful when the cohort has been narrowed to a small set. The results support and constrain activation models of lexical access. [Work supported by NIMH.]

1aSC19. Sources of variability as linguistically relevant aspects of speech. Lynne C. Nygaard and S. Alexandra Burt (Dept. of Psych., Emory Univ., Atlanta, GA 30322)

Recent research has suggested that only linguistically relevant sources of variability, or those which affect the perception of spoken words, are retained in long-term memory. The present study sought to determine if the linguistic relevance of surface characteristics such as speaking rate, overall amplitude, and vocal effort would differentially affect memory retention. For each source of variability, a continuous recognition memory task was used in which listeners were presented with a list of spoken words and asked to judge whether each word in the list was ''old'' (had occurred previously on the list) or ''new.'' Results showed that listeners were better able to identify a word as ''old'' if the word was repeated at the same speaking rate (condition 1), overall amplitude (condition 2), or vocal effort (condition 3), suggesting that the individual surface characteristics which comprise each type of variability were encoded into memory. However, speaking rate and vocal effort produced a greater effect on recognition memory than overall amplitude, indicating that all sources of variability may not be processed and encoded to the same extent. Rather, memory for surface characteristics may be related to the amount of linguistically relevant information each source of variability contains.

1aSC20. Spoken word recognition in older adults: Activation and decision. Jan Charles-Luce (Dept. of CDS and Ctr. for Cognit. Sci., Univ. at Buffalo, Buffalo, NY 14260) and Paul A. Luce (Univ. at Buffalo, Buffalo, NY 14260)

Spoken word recognition may be characterized by two successive stages: (1) activation of multiple form-based representations in memory and (2) frequency-biased perceptual decision. Recent research has suggested that older adults show deficits in controlled processing in perceptual and memory tasks, suggesting that while activation mechanisms subserving spoken word recognition may remain relatively intact over time, perceptual decision processes degrade. In order to investigate the possible loci of older adults' recognition difficulties, perception of specially selected words that orthogonally varied on three dimensions was examined: word frequency, neighborhood density, and neighborhood frequency. Effects of neighborhood density are typically associated with activation mechanisms, whereas neighborhood frequency is associated with perceptual decision. The implications of these results for theories of aging and spoken word recognition will be discussed. [Work supported by grants from NIDCD.]

1aSC21. Automatic generation of word models using piecewise linear segment lattices. Hiroaki Kojima and Kazuyo Tanaka (Electrotechnical Lab., 1-1-4 Umizono, Tsukuba, Ibaraki, 305 Japan)

A framework for ''phonological concept formation'' has been proposed, aiming to generate robust speech recognition models [Kojima *et al.*, Proc. ICSLP 92, Vol. 1, pp. 269–272 (1992)]. For this purpose, a ''piecewise linear segment lattice'' model is proposed. The structure is represented as a lattice of segments, each of which is represented as regression coefficients of feature vectors within the segment. Compared with typical stochastic models like HMM, the advantages are: (1) It needs fewer samples to learn; (2) it represents objects in voluntary precision; and (3) its structure can be dynamically changed by less calculation. An outline of the generation algorithm is as follows: (1) Dividing each sample into segments using DP, where the number of segments is decided based on an MDL-like criterion; (2) matching between the sequences of segments within the same word by DP; (3) modifying the division according to their matching scores; (4) picking up similar (i.e., near) subsequences and gathering them into a phonelike cluster. Speaker-independent isolated word recognition is carried out using the proposed models which are generated in several conditions. The results show that the recognition rate is improved by forming phonelike clusters.

1aSC22. Young children's and older adults' sensitivity to semantic and pragmatic context in speech production. Jan Charles-Luce and Kelly M. Dressler (Dept. of Commun. Disord. and Sci. and Ctr. for Cognit. Sci., Univ. at Buffalo, Buffalo, NY 14260)

It is generally expected that speakers of American English neutralize the /t/-/d/ voice contrast in words like ''writer'' and ''rider'' and instead produce them as homonyms. However, it has been demonstrated that young adults adjust their articulation depending on context and do not always neutralize this contrast. In particular, young adults preserve the voice contrast in semantically biasing contexts and when speaking for a listener. In the present investigation, the interest was in determining when young children became sensitive to semantic and pragmatic context and, consequently, when they adjusted their articulation in ways similar to young adults. Moreover, there was interest in older adults' sensitivity to context. Young children (ages 7–12), young adults (college age), and older adults (ages 60–80) produced minimal pairs containing voiced and voiceless intervocalic alveolar stops in two semantic contexts (biasing and neutral) and in two pragmatic contexts (listener-present and -absent). The results showed developmental changes in speakers' sensitivity to semantic and pragmatic contexts. These results will be discussed in terms of interactive activation, involving the structure and organization of a speaker's linguistic system, and pragmatic compensation, involving a speaker's cognitive decision processes to adjust articulation for the listener's benefit. [Work supported by NIH.]

1aSC23. The role of lexical access in spontaneous speech disfluencies. Gerald W. McRoberts and Herbert H. Clark (Dept. of Psych., Stanford Univ., Stanford, CA 94305)

Pauses and hesitations in spontaneous speech are assumed to result from problems in various aspects of sentence planning. A causal role for difficulties with lexical access is suggested by theoretical accounts of speech production [Levelt, *Speaking* (1989)] and empirical studies showing that pauses are more likely before rare than common words in spontaneous speech [Maclay and Osgood (1959)]. In the present study, word frequency was manipulated in a picture-naming task in which speakers produced the names of ten high- and ten low-frequency pairs of standardized line drawings within a standard sentence frame (e.g., There is a snail to the left of the harp.). The mean frequency of occurrence for high- and low-frequency items was 171.5 (range: 50–591) and 3.5 (range: 1–8) per million, respectively [H. Kucera and W. N. Francis, *A Computational Analysis of Present-day English* (1967)]. Analyses indicate that low-frequency pairs resulted in: (1) more pauses and word substitutions and (2) longer latencies to begin speaking.

1aSC24. An exploration of listener strategies in the lexical segmentation of hypokinetic dysarthric speech. Julie M. Liss, Stephanie von Berger (Dept. of Speech and Hear. Sci., Arizona State Univ., Box 871908, Tempe, AZ 85281), John Caviness, Charles Adler, and Brian Edwards (Mayo Clinic, Scottsdale, AZ 85259)

This investigation examined listener transcriptions of phrases produced by speakers with mild to severe hypokinetic dysarthria to examine individual strategies for the identification of word boundaries in connected speech. It was hypothesized that the most efficient listeners would use syllabic strength information to guide their parsing of the continuous acoustic stream. Six-hundred transcribed phrases (10 listeners $\times 60$ phrases) were coded independently by two judges to identify (1) correct word transcriptions, (2) evidence of accurate lexical parsing, regardless of exact word identification, and (3) the proportions of accurate parsing of strong and weak syllable word onsets. Linear regression analysis of the group data revealed that correct segmentation of strong syllable word onsets predicted listener performance on segmenting weak syllable word onsets [$R=0.805$, $F(1,29)=51.440$, $p<0.001$]. Despite a wide range of listener performance on ''words correct,'' individual strategies for perceptual segmentation were evident only in the transcriptions for the most severe speaker. In this case, the poorest listeners exhibited disproportionate difficulty with the segmentation weak syllable word onsets. Results suggest that a listener's ability to use syllabic strength information in lexical parsing determines, in part, their ability to recognize word onsets. [Work supported by NIDCD, NIH.]

1aSC25. Preliminary report on syllable level organization observed in Parkinsonian speech obtained in individuals before and after posteroventral pallidotomy. Q. Emily Wang (Dept. of Commun. Disord. and Sci., Rush Univ., 1653 W. Congress Pkwy., Chicago, IL 60612) and Kathleen Shannon (Rush–Presbyterian–St. Luke's Medical Ctr., Chicago, IL 60612)

Syllable level organization has been evidenced in the articulatory movement patterns in different languages [C. P. Browman and L. Goldstein, *Producing Speech: Contemporary Issues*, 19–34 (1995); R. A. Krakow, ''The articulatory organization of syllables: A kinematic analysis of labial and velar gestures,'' Ph.D. dissertation, Yale University (1989); Q. E. Wang, ''Are syllables units of speech motor organization?—A kinematic analysis of labial and velar gestures in Cantonese,'' Ph.D. dissertation, University of Connecticut (1995)]. This study analyzed speech samples produced by nondemented individuals with idiopathic Parkinson's disease (Hoehn and Yahr stage 2–3) who underwent posteroventral pallidotomy. The data were collected with the patients on and off their medications as well as pre- and post-operatively. The preliminary results indicated that the patients were able to produce stimuli with syllable-initial nasals with less difficulty than those containing syllable-final nasals, and as the patients' motor performance improved, their ability to produce the stimuli with syllable-final nasals also improved. This may suggest that the speech motor programming and execution are different for the phonemically identical phonemes in syllable-initial and syllable-final positions. [Work supported by Rush University and NIH Grant DC-00121 to the Haskins Laboratories.]

1aSC26. Effects of discourse structure on $F0$ in Japanese: Raising versus lowering. Jennifer J. Venditti (Dept. of Linguist., Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210)

Previous studies on English and other languages have shown that discourse structure has an influence on the intonation of a string of sentences or phrases. With respect to fundamental frequency in particular, both discourse-initial raising of $F0$ and final lowering effects have been reported. The present study examines whether discourse structure also influences intonation in Japanese, and if so, to what extent initial raising and final lowering interact to cue structure. Hierarchically organized discourses were constructed in which the target sentence position was varied. These short discourses were recorded by native speakers of Tokyo Japanese, and

the fundamental frequency contours of the target sentences in each position were compared. Results indicate that there is a robust raising effect on discourse-initial phrases, as reported for other languages. The degree of raising is determined by an interaction between distance from the start of the utterance and pitch range. In contrast, there is little effect of lowering on discourse-final phrases. The few phrases that did show an effect suggest that syntactic structure may interact with the lowering process.

1aSC27. Identification, vocal rating, and acoustical measurement of acted emotion. William J. Strong (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT 84602), Bruce L. Brown, Matthew P. Spackman, Rong Wang (Brigham Young Univ., Provo, UT 84602), and Stanley Feldstein (Univ. of Maryland, Baltimore, MD 21228-5398)

A series of studies of eight acted emotions compared acoustical measurements, respondents' identifications, and vocal ratings. Predictions were made as to which of these three approaches would be most accurate in differentiating between emotions within a pair. Emotion pairs were classified as logically similar and/or perceptibly similar. Logically similar emotion pairs (such as sadness/depression or anger/hate) were better distinguished by vocal/acoustic analysis. Perceptibly similar but logically dissimilar emotions (such as anger, fear, and joy) were better distinguished by respondents' identifications. Multivariate statistics were used to compare the respondents' identification space, and vocal rating space, and the acoustical measurement space.

1aSC28. Are selective adaptation effects independent of cognitive load? Donna Kat and Arthur G. Samuel (Dept. of Psych., SUNY, Stony Brook, NY 11794-2500)

Selective adaptation occurs through the repeated presentation of a sound (the ''adaptor''), and leads to a reduction in the perception of similar sounds. Adaptation has been used to investigate the nature of early speech representations. Work in this laboratory has recently demonstrated that perceptually restored phonemes can produce reliable adaptation effects, and that these effects are occurring at relatively early levels (e.g., phonemic rather than lexical) of processing [A. G. Samuel, *Cognitive Psychology* (in press)]. The current study is designed to determine whether the adaptation effects are so low-level and automatic that they do not require cognitive resources. There are three conditions in the current study: (1) adaptation alone (control), (2) adaptation during continuous arithmetic problems, and (3) adaptation during continuous rhyming judgments (presented visually). Preliminary results indicate that continuously solving arithmetic problems does not reduce the adaptation effect, indicating no general cognitive involvement in adaptation. The rhyming task tests for any more specific involvement of language processors. Immunity to this secondary task would support a very low-level, automatic basis for adaptation. [Work supported by NIMH.]

1aSC29. Linguistic strategies in the first 6 months of life. Francisco Lacerda and Ulla Sundberg (Inst. of Linguist., Stockholm Univ., S-106 91 Stockholm, Sweden)

The ability to detect word contrasts embedded in natural sentences was studied with 62 Swedish infants whose ages ranged from 48 to 147 days, using the high-amplitude sucking (HAS) technique. The infants listened to one of four possible pairs of natural carrier sentences, produced as child-directed speech, in which a target word was inserted: (1) a pair of sentences in which the contrasting target words was inserted in focal position; (2) a pair of sentences in which the target words were out of the sentence focus; (3) a pair of sentences that differed only in the position of the sentence focus; and (4) a control pair with two identical sentences. When this group of subjects is divided into two age groups, one below and the other above the median age for all the subjects, a pattern of interaction between the age groups and the experimental conditions is observed suggesting that word discrimination capacity in the younger infants may be disrupted by dominant $F0$ variations. For the older group, a tendency to attend to the word contrasts delivered in focal position seems to start to emerge. [Research supported by The Bank of Sweden Tercentenary Foundation, Grant No. 94-0435.]

1aSC30. Contrastive emphasis in elicited dialogue. Donna Erickson (Ctr. for Cognit. Sci., 208 Stadium East, 1961 Tuttle Park Pl., Ohio State Univ., Columbus, OH 43210) and Ilse Lehiste (Ohio State Univ., Columbus, OH 43210)

Phrase level characteristics of $F0$ and their interactions with duration patterns for a set of contrastively emphasized digits in elicited dialogues are examined. Phrases consisting of three digits plus a street name were elicited in a dialogue format designed to have the speaker repeat the same correction on one of the digits up to five or six times. Perception tests determined which phrases where ''best perceived'' and ''worst perceived'' as to emphasis. Onset, offset, and peak $F0$ within the sonorant portion of each digit of those phrases ''best perceived,'' ''worst perceived,'' and the reference phrases, with no corrective emphasis, were measured. Results suggest that speakers produce well-perceived contrastive emphasis by lengthening the duration of the emphasized word, shortening the duration of the other words within the phrase, and producing an extensive pitch drop between the emphasized word and the following words in the phrase. In an elicited dialogue situation in which the speaker is forced to produce repetitively the same item with contrastive emphasis, different combinations of these two cues are used, presumably in an attempt to maximize the chance that emphasis will be perceived.

1aSC31. Social influence on the phonemic transformation effect. Verlin B. Hinsz (Psych. Dept., North Dakota State Univ., Fargo, ND 58105), Magdalene H. Chalikia (Moorhead State Univ., Moorhead, MN 56563), and David Matz (North Dakota State Univ., Fargo, ND 58105)

Earlier studies found that repeated sequences of brief steady-state vowels are heard as verbal forms, a phenomenon referred to as the phonemic transformation effect (PTE). It has also been established [Chalikia *et al.*, J. Acoust. Soc. Am. **91**, 2422(A) (1992)] that, when two listeners' responses differ, they can identify the particular stimulus corresponding to each other's verbal forms. Other research suggests that unclear stimuli can be influenced by social forces. The present study examined social influences on the PTE. Participants were first asked to describe their verbal forms for ten vowel sequences used in previous studies. Then they were presented with verbal forms reported by previous listeners and were asked to match them to the ten stimuli. Most listeners performed the matching task, indicating that they could perceptually reorganize each stimulus. Finally, they were asked to listen to the sequences again and describe their verbal forms. About 47% of these responses corresponded to those provided by previous listeners, indicating that social information influenced some of the second responses to these stimuli. Implications concerning the perceptual interpretation of speech and social impact theory will be discussed.

Session 1aSP

Signal Processing in Acoustics and Speech Communication: Nonlinear and Higher Order Techniques

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Contributed Papers

10:15

1aSP1. Detection of narrowband harmonics using higher-order spectral methods. Martin L. Barlett, G. Douglas Meegan, and Gary R. Wilson (Appl. Res. Labs., Univ. of Texas, P. O. Box 8029, Austin, TX 78713-8029)

An empirical investigation of detection performance for multiple harmonic components imbedded in controlled noise was performed using both spectral correlation and stationary bispectrum detection statistics. Conventional power spectral processing is used as a metric for detection gain obtained using the higher-order processing methods. Relative performance between power spectral and higher-order spectral methods will be presented as a function of both signal-to-noise ratio and harmonic ''falloff'' at fixed probability of false alarm. The empirical results will also be compared with analytic results obtained assuming asymptotic statistics [G. R. Wilson and K. R. Hardwicke, ''Nonstationary Higher Order Spectral Analysis,'' Appl. Res. Labs. Tech. Rep. ARL-TR-91-8, Applied Research Laboratories, University of Texas, Austin, TX (1991)]. [Work supported under the Independent Research and Development Program, Applied Research Laboratories, The University of Texas at Austin.]

10:30

1aSP2. Fundamental frequency estimation of speech signals using a nonlinear observer. Taro Yoshihama, Asako Doi, and Yoshihisa Ishida (Dept. of Electron. and Commun., Meiji Univ., 1-1-1, Higashi-Mita, Tama-ku, Kawasaki, 214 Japan)

This paper describes a new method of estimating the fundamental frequency of speech signals using an adaptive Fourier analysis (AFA) algorithm based on the presumption of signal parameters using a nonlinear observer. The recursive presumption of signal parameters such as fundamental frequency can be effectively implemented by using the proposed AFA algorithm. Nonlinear observers can be used to measure nonlinear parameters which are included in observed signals. Recursive estimation procedures make it possible to follow the slow changes of signal parameters to be measured. In real-time signal analysis, the recursive algorithm is preferred to the conventional Fourier transformation because of this property. The AFA algorithm, which is based on the recursive discrete Fourier transform (RDFT), is a recursive algorithm for the simultaneous estimation of the Fourier coefficients and instantaneous fundamental frequency of speech signals. This AFA algorithm is applied to speech signals with an arbitrary length of the transform window. Further efforts are required for better convergence speed of the proposed algorithm, but this method is mostly capable of adapting and tracking nonlinear signals such as speech sound.

10:45

1aSP3. Nonlinear spectrum estimation using a new neural network with a level estimator. Hideaki Imai, Yoshikazu Miyanaga, and Koji Tochinai (Hokkaido Univ., N13 W8 kita-ku, Sapporo, 060 Japan)

This paper proposes a new nonlinear signal processing by using a three-layered network which is trained with self-organized clustering and supervised learning. The network consists of three layers, i.e., a self-organized layer, an evaluation layer, and an output layer. Since the evaluation layer is designed as a simple perceptron network and the output layer is designed as the fixed weight linear nodes, the training complexity is the same as the self-organized clustering and a simple perceptron network. In other words, quite high speed training can be realized. Generally speaking, since the data range usually used in signal processing is arbitrarily large, the network output should also cover this range. However, it may be difficult for only one node in the network to output these data. Instead of this technique, if this dynamic range is covered by using several nodes, the complexity of each node is reduced and the associated range is also quite limited. This results in a higher performance of this network than the conventional ones. As one of the objectives, this paper introduces the spectrum envelope estimation of speech waveforms. It is shown that accurate spectrum envelopes can be obtained.

11:00

1aSP4. Feature extraction for a neural network classifier. Mark Wellman and Nassy Srour (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197)

Unattended ground sensor (UGS) networks are intended to detect and localize the presence of strategic relocatable targets in the theater of operation over several kilometers. Passive acoustic sensors, an integral part of UGS, have achieved a high level of maturity and will allow acoustic target classification for tracked and wheeled vehicles. Of primary importance in the classification problem is the selection of a robust feature extraction technique, tolerant of both the environment and the nonstationary nature of the acoustic signatures. Several feature extraction techniques were used with experimental acoustic data collected from a small baseline, circular array. Results will be presented of the classification for acoustic features using a backpropagation neural network with simple power spectrum, harmonic line association [J. A. Robertson, IIT Research Institute, in-house report], principal components [J. Mao and A. K. Jain, IEEE Trans. Neural Networks **6** (2) (1995)], and wavelet packet [K. Etemad and R. Chellappa, Proc. First Intl. Conf. on Image Processing (November 1994)] feature extraction techniques.

11:15

1aSP5. Blind deconvolution based on common zeros of the *z* transforms of two observed signals. Ken'ichi Furuya and Yutaka Kaneda (Speech and Acoust. Lab., NTT Human Interface Labs., 3-9-11 Midori-cho Musashino-shi, Tokyo, 180 Japan)

A new method is proposed for recovering an unknown source signal, which is observed through two unknown channels characterized by finite impulse response filters. Unlike conventional blind deconvolution methods, this method can recover not only the spectrum of the signal but also the phase characteristics. This method is based on a cost function that comes from a filter arrangement with a double-layer structure. The cost function is minimized when the common zeros of the z transforms of the two observed signals are extracted. If there are no common zeros between the system transfer functions of the two unknown channels, the common zeros of the observed signals represent the source signal and the noncommon zeros represent the characteristics of the two channels. Therefore, the source signal can be recovered by separating the common zeros from the other zeros, that is, by minimizing the cost function. Adaptive filters are used for this procedure. Computer simulations using room transfer functions as the two unknown channels demonstrate the effectiveness of this method.

11:30

1aSP6. On the use of the zero-crossing analysis for multi-channel signal processing. Takaaki Sugihara, Shoji Kajita, Kazuya Takeda, and Fumitada Itakura (Dept. of Info. Elec., Nagoya Univ., Furo-cho 1, Chikusa-ku Nagoya-shi, 464-01 Japan)

Since time precision in finding delays between channels is one of the most important issues in estimating direction of arrival (DOA) from multichannel signals, a zero-crossing analysis is proposed as a fine and robust time-domain preprocessing for the DOA finding. As for the preliminary evaluation of noise robustness using zero-crossing information, how the zero-crossing points are affected is investigated, i.e., how far the zero-crossing points move from the original points, under the presence of noise. For test speech, a 48-kHz sampling of a male voice (upsampled from an 8-kHz signal) is used after an adding machine generated white noise so that the SNR of the signal becomes 0 to 20 dB. The robustness of the zero-crossing analysis can be concluded from the summarized results: (1) the mean value of the distance is less than 100 μs throughout the SNR; and (2) the standard deviation of the distribution becomes large as the SNR decreases; the standard deviation is less than 0.125 even under the condition of 0 dB SNR.

11:45

1aSP7. Source identification using nonlinear signal processing. Azizul H. Quazi (Naval Undersea Warfare Ctr. Detachment, New London, CT 06320)

Traditional techniques of signal processing in both the time domain and frequency domain are often not sufficient enough to characterize the complex dynamics of underwater sources that generated, radiate, and/or reflect sonar signals. The purpose of this paper is to analyze simulated signals by means of a nonlinear technique (based on mutual information), which is a completely new method that represents a revolutionary advance. Mutual information is a measure of the amount of information that one random variable contains about another random variable. It is also a reduction of uncertainty of one random variable, which is due to the knowledge of the other. Here the results of simulated sonar signal processing are presented based on mutual information and which is then compared to the traditional correlation technique. The results indicate that the mutual information is a more powerful tool compared to the correlation technique, and the results based on mutual information provide clues leading to source identification.

12:00

1aSP8. Application of NORDEN transform to time-frequency characterization of time-varying signals. Nai-chyuan Yen (Physical Acoust. Branch, Naval Res. Lab., Washington, DC 20375-5320) and Manli C. Wu (NASA Goddard Space Flight Ctr., Greenbelt, MD 20771)

The non-orthodox decomposition (NORDEN) transform developed by NASA [N. Huang *et al.*, "The Empirical Mode Decomposition and the Hilbert Spectrum for Nonlinear and Nonstationary Time Series Analysis" (to be published)] is designed to adaptively break down a complex time-varying signal into a sum of several simple mode functions. Each single mode function can then be expressed in terms of an analytic signal whose amplitude and phase vary with time and can be displayed as a trace in a time frequency plot, the VEIN diagram, with magnitude expressed as line thickness or color coded. This methodology of signal analysis, unlike other traditional transform methods, i.e., Fourier, Wigner-Ville, and Wavelet, requires no special window, preselected kernel, or particularly selected mother wavelet and provides more detailed dynamic information about the signal under study in terms of its dominant components and their instantaneous frequency variation in the observation time. Several examples from various types of acoustic signals are examined with this algorithm to demonstrate its signal analysis capability.

Session 1aUW

Underwater Acoustics: Transducers, Hydrophones, Arrays, Systems and Related Topics

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Chair's Introduction—10:15

Contributed Papers

10:20

1aUW1. A simple constant beamwidth transducer. D. J. Allwright (Mathematical Inst., Oxford Univ., Oxford, UK) and J. Power (S. A. I. C. Ltd., Cambridge CB3 0RD, UK)

Many designs for constant beamwidth transducers have appeared over the years, but these usually involve complicated array shading techniques or "lenses" applied to ultrasonic devices. With locally reacting materials, such as PVDF for example, it is possible to manufacture a composite material which automatically produces the appropriate frequency-dependent shading of the response in a single, continuous transducer. The material makes use of the capacitive (dielectric) nature of the active element in conjunction with a specially designed, electrically resistive layer, to which a single point connection is made. The resulting CR circuit acts as an integrator, and since the resistance to the remote parts of the transducer is largest, the response of those parts progressively reduces as frequency is increased. In other words, the effective size of the transducer reduces (i.e., "shrinks") as frequency is increased. Correct design of the composite provides a constant directivity characteristic. Several applications of this "shrinker technology" are discussed, including microphones, loudspeakers, and hydrophones.

10:32

1aUW2. Broad bandwidth underwater transducer. James E. Barger (BBN, Inc., 70 Fawcett St., Cambridge, MA 02138)

A new underwater transducer design technique is described that provides for ultrabroad-bandwidth directive arrays that are lightweight and thin (for easy towing). An example of this new technique is a planar array that is 1 m high, 0.75 m wide, and 0.15 m thick. This array has greater than 60% radiation efficiency throughout the frequency band extending from 400 Hz to 4 kHz, and a nominal power output within this band of 10 kW. The array is neutrally buoyant, and uses Navy type-III PZT actuators. The broadband goal is met by designing the transducers in such a way that their own stiffness reactances are mostly canceled by their masslike radiation reactances over the entire operating band. This feat is accomplished both by properly adjusting the ratio of actuator cross-sectional area to radiation area, and by using low dynamic transducer mass. Since the principle of operation requires for each transducer that its motion be influenced by its own radiation load, it is necessary to control each transducer's motion so that the desired radiation pattern is achieved. This is done by a filtered-X feedforward LMS adaptive controller.

10:44

1aUW3. Characterize reverberation spatial distribution using an L-shape array. Yung P. Lee (Science Applications Intl. Corp., 1710 Goodridge Dr., MS T1-3-5, McLean, VA 22102)

Matched-field processing (MFP) is a generalized beamforming. Instead of using plane-wave replica vectors in the plane-wave beamforming, in MFP, an acoustic model is used to calculate replica vectors in a search space. In addition to azimuthal and elevation information, which can be obtained by plane-wave beamforming, MFP also provides range and depth information. In active systems, scattered signals are usually modeled by two-way, source-to-scatterer and scatterer-to-receiver, propagation. Assuming that scatterers are illuminated and reradiate energy as point sources, the scattering field can be characterized by using just one-way, scatterer-to-receiver propagation. Passive MFP can then be used to map the spatial distribution of the scatterers. A shallow-water experiment took place off the Gulf of Mexico in November 1995; reverberation was measured on an L-shaped array from a bottomed cw source. Passive MFP was used to characterize the spatial distribution of the measured reverberation. [Work supported by U.S. Navy.]

10:56

1aUW4. Vertical array performance in shallow water with a directional noise field. Kwang Yoo and T. C. Yang (Naval Res. Lab., Washington, DC 20375)

The performance of a vertical array for source localization and for target detection (array gain) is studied in shallow water under a directional noise field. The acoustic environment consists of a downward refractive sound-speed profile. At mid (e.g., 500 Hz) frequencies the vertical directionality of the surface generated noise exhibits a notch near the signal arrival direction, i.e., which is close to horizontal for a submerged source. The ability of the beam filter in rejecting directional noise has been previously demonstrated using conventional beamforming. The same capability is carried over into full field processing by using matched-beam processing which is an equivalent of matched-field processing conducted in the beam domain. It is shown with simulated data that matched-beam processing incorporating a beam filter of 10 deg enhances the array output signal-to-noise ratio (array gain) by more than 5 dB compared with conventional beamforming and matched-field processing. For white noise, matched-field processing yields a higher peak-to-sidelobe ratio than matched-beam processing with a beam filter. For directional noise, matched-beam processing with a beam filter yields better results than matched beam processing using a minimum variance correlator. The noise field in the background ambiguity surface (the sidelobes) has been suppressed by the beam filter.

11:08

1aUW5. A new towed array shape estimation scheme for real-time sonar systems. Feng Lu, Evangelos Milios (Comput. Sci. Dept., York Univ., 4700 Keele St., Downsview, ON M3J 1P3, Canada), and Stergios Stergiopoulos (Defence Res. Establishment Atlantic, Dartmouth, NS B2Y 3Z7, Canada)

In real-time towed array systems, performance degradation of array gain occurs when beamforming is carried out on the sensor outputs of a line array which is not straight. In this paper, a new method is proposed for array shape estimation. The procedure consists of two steps. First, the tow-point-induced motion is formulated in the time domain based on the constraints from the tow-point compass-sensor readings and from a discretized Paidoussis equation. At each time instance, the shape estimate is solved from a linear system of equations. It is shown that this solution is equivalent to a previous frequency-domain solution, while the new approach is much simpler. In the second step, the tail compass-sensor data are used to adjust the overall array shape. By noting that variations in the ship speed lead to a distortion in the normalized time axis, the predicted tail displacements are first registered with the tail sensor readings along the time axis. Then distortions in the estimated array shape over its length can be compensated accordingly. A slow-changing bias between sensor zeros is also modeled in order to remove systematic sensor errors. The effectiveness of the new algorithm is demonstrated with real sea-trial data.

11:20

1aUW6. Simulated performance of an acoustic modem using phase-modulated signals in a time-varying, shallow-water environment. Christian Bjerrum-Niese and Leif Bjørnø (Dept. of Industrial Acoust., Tech. Univ. of Denmark, DK-2800 Lyngby, Denmark)

Underwater acoustic modems using coherent modulation, such as phase-shift keying, have proven to efficiently exploit the bandlimited underwater acoustical communication channel. However, the performance of an acoustic modem, given as maximum range and data and error rate, is limited in the complex and dynamic multipath channel. Multipath arrivals at the receiver cause phase distortion and fading of the signal envelope. Yet, for extreme ratios of range to depth, the delays of multipath arrivals decrease, and the channel impulse response coherently contributes energy to the signal at short delays relative to the first arrival, while longer delays give rise to intersymbol interference. Following this, the signal-to-multipath ratio (SMR) is introduced. It is claimed that the SMR determines the performance rather than the signal-to-noise ratio (SNR). Using a ray model including temporal variations of the shallow-water environment, the performance of the acoustic modem may be estimated. Simulations indicate that optimum performance is not necessarily found at receiver depths yielding the maximum total signal level, since the SMR may correspondingly be low due to strong intersymbol interference. [Work sponsored by the Danish Technical Research Council.]

11:32

1aUW7. Development of the acoustic packet data relay communication system and the results of its sea trial. Hiroshi Ochi, Takuya Shimura, Yasutaka Amitani, and Toshio Tsuchiya (Deep-Sea Technol. Dept., Japan Marine Sci. and Technol. Ctr., Natsushima-cyo 2-15, Yokosuka, 237 Japan)

In order to recover the data obtained by various sensors deployed in the ocean, a study was made concerning the acoustic digital data communication system [Ochi *et al.*, Proc. Meeting Marine Acoust. Soc. Jpn., pp. 85–86 (1996) (in Japanese)]. The data communication system was constructed by up to 99 of the same type of equipment in the area, and its specifications are: a half-duplex communication, FSK modulation, a 2500bps data rate, HDLC (high-level data link control procedure) protocol packet transmission, CRC (cyclic redundancy check) code, and frame retransmission for error detection/correction. The sea trial of this system was done at about a 4000-m depth area. Three sets of equipment were deployed in the ocean, and one of them hangs from the ship. Then, relay data transmission, among the three sets of equipment, was carried out. The detail of that sea trial will be shown at this presentation.

11:44

1aUW8. The effectiveness of a thin wedge design anechoic lining for long-time signature measurements.. Walter H. Boober (NUWC Code 8211, Bldg. 1171, Newport, RI 02841) and Scott Emery (Vector Res. Co., Inc., Rockville, MD 20852)

A thin wedge design anechoic lining has proven to be very effective for long time signature measurements at frequencies above 10 kHz. Partial treatment on five of six surfaces in a $60\times40\times35\text{-ft}^3$ depth tank resulted in accurate transfer functions of a test transducer using late sample delays for long-time signature waveforms. Comparison of data resulting from delays prior to wall/surface reflections with delays as late as 200 ms revealed negligible contribution, if any, from reflections. The ‘‘reflection’’ data when graphed over ‘‘free-field’’ data were indistinguishable. Tests were run using a directional array projector and a horizontally omnidirectional projector. The hydrophone was an H-52; a 5.1-cm vertical line array, omnidirectional in the horizontal plane. This allowed some discrimination from surface and bottom reflections. Reflections from the untreated wall were strong contributors while using the omnidirectional projector and were revealed in radiation patterns of the directional array. These tests under real world everyday measurement conditions demonstrate the effectiveness of the panels for long pulse times with late delay sampling in constrained boundaries.