Session 1aAO

Acoustical Oceanography: Bioacoustics I

Timothy K. Stanton, Chair

Department of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Bigelow 201, Woods Hole, Massachusetts 02543-1053

Chair's Introduction—7:50

Invited Paper

8:00

1aAO1. Acoustics in fisheries in the 21st century. George A. Rose (Fisheries and Marine Inst. of Memorial Univ. of Newfoundland, St. John's, NF A1C 5R3, Canada)

The history, current state, and future potential of acoustics in fisheries research and management will be addressed. Fisheries management worldwide is undergoing a fundamental paradigm change in response to population pressures and overharvesting of marine and freshwater stocks. Future fisheries management will be more conservation based and responsive to ecosystem considerations. These trends will demand more comprehensive knowledge of the components of fisheries' ecosystems and their dynamics, and more realistic estimates of the precision of ocean measurements. Acoustics is poised to take a new and leading role in these developments. Acoustics is nondestructive of marine life, can be near-synoptic, and can measure most of the water column. However, acoustic methods are still hampered by problems with detectability, especially near boundaries, uncertainties about target strengths of the myriad of species and life stages whose scattering properties are complex and behaviorally mediated, and taxonomic identification. These problems will be discussed, and a general experimental plan will be laid out to solve them. The 21st century of fisheries belongs to acoustics and allied technologies.

Contributed Papers

9:00

1aAO2. Backscatter measurements of Atlantic herring in the Northwest Atlantic. William L. Michaels, J. Michael Jech (Northeast Fisheries Sci. Ctr., 166 Water St., Woods Hole, MA 02543), and David A. Demer (Southwest Fisheries Sci. Ctr., La Jolla, CA 92038)

Annual fisheries acoustic surveys have recently been implemented in the Gulf of Maine and Georges Bank regions to estimate the abundance and biomass of Atlantic herring (*Clupea harengus*). Research was devoted to the verification of backscatter measurements from Atlantic herring using multifrequency echo-integration, omni-directional sonar, pelagic trawling, and underwater video operations. Results indicated that individual backscatter measurements during the day were significantly higher compared to the night and twilight periods. The diurnal behavior patterns of Atlantic herring contributed to this variability because of their vertical migration to near bottom during day and to surface waters during night. Observations showed that herring shoaled in stationary schools above offshore banks during the night to feed, and tend to swim more actively during twilight and day periods. An increase in the backscatter from herring during the day may be explained by their more horizontal orientation when actively swimming. Fisheries acoustic surveys conducted during day and night must incorporate systematic adjustments to account for the diurnal bias in acoustical estimates.

9:15

1aAO3. Broadband acoustic backscattering from alewife fish: Experiment and analysis. D. Benjamin Reeder, Timothy K. Stanton, Dezhang Chu (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, breeder@whoi.edu), and J. Michael Jech (NOAA/NMFS Northeast Fisheries Sci. Ctr., Woods Hole, MA 02543)

A series of laboratory scattering measurements and associated analysis have been conducted involving alewife fish (Alosa pseudoharengus). A greater-than-octave bandwidth, shaped, linearly swept, frequency modulated signal was used to insonify live, adult alewife that were tethered while being rotated. The acoustic scattering time series were measured in 1-deg increments of orientation angle over all angles in two planes of rotation. Dominant acoustic scattering mechanisms are identified through both spectral and time-domain (pulse compression) analyses. Results demonstrate the dependence of scattering strength upon frequency and angle of orientation in the lateral (horizontal) and dorsal/ventral aspects. The pulse compression processing of the echoes from the animals temporally resolves multiple returns from an individual which are correlated with size and orientation. [Work supported by ONR, NOAA and CICOR (Cooperative Institute for Climate and Ocean Research).]

9:30

1aAO4. Three-dimensional visualization of acoustic backscattering and models by alewife. J. Michael Jech (NOAA/NMFS Northeast Fisheries Sci. Ctr., 166 Water St., Woods Hole, MA 02543), D. Benjamin Reeder, Timothy K. Stanton, and Dezhang Chu (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

The advent of nontraditional underwater acoustic instrumentation (e.g., multibeam and sector-scanning sonar) and incorporating fish behavior in quantitative fisheries assessment require that backscatter be measured and modeled at a variety of aspect angles. Acoustic backscattering measurements were obtained in a large laboratory tank using live alewife (*Alosa pseudoharengus*). An individual alewife for each series of measurements was tethered and rotated in two planes of orientation (dorsal/ventral and lateral). The alewife were insonified with a broadband (40–100 kHz) chirp signal and bistatic scattering geometry was used. Backscattering amplitudes for all angles of orientation (3-D scattering ambit) were modeled using a Kirchhoff ray-mode model [C. S. Clay and J. K. Horne, J. Acoust. Soc. Am. **96**, 1661–1668 (1994)] and digital images of the fish body and swimbladder morphometry. Visualization of the scattering ambit provides a quantitative examination of the effects of fish orientation on echo amplitude. Comparisons of backscattering amplitudes from the model and measurements along the dorsal/ventral and lateral planes are given. The

1a MON. AM

utility of broadband measurements for fish backscattering amplitude measurements and the integration of acoustic models in fisheries assessments are discussed. [Work supported by ONR, NOAA/NMFS, and CICOR.]

9:45–10:00 Break

10:00

1aAO5. Incorporating behavior in backscatter model predictions of walleye pollock target strength. John K. Horne (Univ. of Washington and Alaska Fisheries Sci. Ctr., 7600 Sand Point Way NE, Bldg. 4, Seattle, WA 98115, john.horne@noaa.gov) and Michael W. Davis (Fisheries Behavioral Ecology Program AFSC, Hatfield Marine Sci. Ctr., Newport, OR 97365)

Target strength to size relationships for fish and zooplankton are typically based on *in situ* field or *ex situ* tethered measurements. Combining backscatter measurements with species-specific backscatter models improves the translation of acoustic data to computations of type, number, and size of aquatic organisms. Variability in acoustic measurements is largely due to the choice of carrier frequency and organism behaviors. Shoaling, schooling, and orientation are three behaviors that influence the amplitude of echoes received from individuals or aggregations of aquatic organisms. Kirchhoff-ray mode (KRM) backscatter models of individual fish were combined with tilt and roll tank observation data to predict and visualize backscatter from individual and groups of walleye pollock (Theragra chalcogramma). Model predictions were compared to *in situ* target strength measurements at 38 kHz in the Bering Sea. Incorporating behavior in distributions of echo amplitudes should increase accuracy of target strength to length and abundance estimate accuracy. [Work supported by ONR and the Alaska Fisheries Science Center.]

10:15

1aAO6. Capelin TS: Effect of individual fish variability. Yvan Simard (Dept. of Fisheries and Oceans, Maurice Lamontagne Inst., Mont-Joli, QC G5H 3Z4, Canada), John K. Horne (Univ. of Washington, Seattle, WA), Diane Lavoie, and Ian McQuinn (Maurice Lamontagne Inst., Mont-Joli, QC, Canada)

Capelin (*Mallotus villosus*) is an important forage fish in northern latitudes. The effect of the individual biological variability on the target strength (TS) at various acoustic frequencies was investigated from a sample collected in the St. Lawrence estuary. The geometric properties of the fish and its swimbladder were measured from radiographs obtained from 45 anesthetized fish, from 12- to 16-cm total length. The data were input to a backscattering model exploring the effect of fish shape on TS as a function of acoustic frequency, length, and tilt angle. The swimbladder had similar cross sections in both lateral and dorsal views. It represented 5.5% (s.d. 1.1%) of lateral body cross section and 8.2% (s.d. 1.8%) of the dorsal body cross section. The swimbladder cross section was related to the fish total length but the variation around the mean for a given length was $\pm 40\%$. This large variability is equivalent to the change in cross section between tilt angles of 0 and 90 deg for an average fish. The variability in shape parameters is paralleled with changes in the modeled backscatter patterns. The TS versus frequency relation exhibits substantial peaks and troughs, notably in the range of acoustic frequencies commonly used in fisheries acoustics ($\sim$38–200 kHz).

10:30

1aAO7. *In situ* target strength measurements of a deep-water fish, orange roughy. Rudy J. Kloser (CSIRO Marine Res. Hobart, Tas, Australia 7001)

In situ target strength methods have been reported to be the best way to convert echo integration acoustic surveys into biomass. Detailed *in situ* target strength measurements were obtained by both drifting over and towing through aggregations of orange roughy at 700–800-m depths. The aggregations displayed a marked avoidance response at times to the deep-towed body and it was difficult to ascertain when single orange roughy targets were being detected. Using three frequencies, it was possible to include/exclude various targets in the selection process and have more confidence that the observed targets were the species of interest. Calibration of the data was simplified by suspending a sphere under the transducer for all the drift experiments. The extracted target strength histograms for the drift experiments were very broad with many modes, and distribution mean target strengths were sensitive to the density of the schools and also not surprisingly the concentration of other species. Towing experiments were highlighted by a marked avoidance response as the towed body neared the orange roughy schools. The extracted *in situ* target strength histograms showed that the mean target strength for diving orange roughy can decrease by 3.1 dB (50%).

10:45

1aAO8. Some problems and solutions for the measurement of fish target strength: A study case with Atlantic redfish (*Sebastes spp.*). Stéphane Gauthier and George A. Rose (Fisheries Conservation, Marine Inst., Memorial Univ. of Newfoundland, P.O. Box 4920, St. John's, NF A1C 5R3, Canada)

Potential biases in the measurement of redfish target strength (TS) were examined by comparing *ex situ* and *in situ* approaches. *Ex situ* experiments were conducted on individuals after developing a method to maintain live fish in good condition. The main problem with the manipulation of these species was the inflation and distortion of the swimbladder. To minimize this bias, fish were kept in sea cages after acclimation at depth and TS was measured in a camera-monitored apparatus set in proximity to the capture site. A series of *in situ* acoustic-trawl experiments was conducted on several aggregations of redfish in Newfoundland waters. TS were collected using a hull-mounted EK500 split-beam transducer and a deep-tow dual-beam system. The dual-beam transducer was calibrated at different depths to test if change in pressure and/or temperature affected its sensitivity. This system was used to measure aggregations of fish under different transducer depth. The data indicated that biases in *in situ* TS estimates increased with the range of observation, the density of fish, and the presence of multiple targets formed by the clustering of smaller organisms. Depending on the nature of the bias, TS can be over- or underestimated by as much as 6 dB.

11:00

1aAO9. The development of a system for remotely monitoring the three-dimensional movement of acoustically tagged fish. Tracey W. Steig and Samuel V. Johnston (Hydroacoustic Technol., Inc., 715 NE Northlake Way, Seattle, WA 98105, consulting@htisonar.com)

Over the last 3 years, a passive acoustic system was developed to monitor the behavior of acoustically tagged fish. The acoustic tag receiver was designed to simultaneously monitor up to 16 omni-directional hydrophones. The signals received at the hydrophones were synchronized to determine the arrival times for each pulse transmitted by the acoustic tag. Arrival times were used to calculate the three-dimensional position of a tagged fish as it moved through the water. Since precise three-dimensional positions of the hydrophone array was required, methods were developed to accurately determine these positions. Additional algorithms were developed to track the received signals and calculate the three-dimensional position of a tagged fish. The resulting acoustic tag system was deployed at various hydroelectric dams on the Columbia, Snake, and Cowlitz rivers. Field studies monitored downstream migrating salmonids as they approached and passed into the turbine intakes, spillways, and juvenile bypass channels. The size of the juvenile salmonids ranged from 160 to 240

mm. The tags were approximately 7 mm in diameter, 23 mm in length, and operated at a transmitting frequency of 300 kHz. Fish movement patterns were tracked over time, with submeter resolution in most cases.

11:15

1aAO10. A method for estimating the accuracy of the three-dimensional position of acoustically tagged fish. John E. Ehrenberg and Tracey W. Steig (Hydroacoustic Technology, Inc., 715 NE Northlake Way, Seattle, WA 98105, consulting@htisonar.com)

Acoustic tag systems are beginning to be used to study the behavior of fish in a region of interest. In particular, tag systems are being successfully used to study the behavior of downstream migrating salmon smolts (Oncorhynchus spp.) as they approach hydroelectric dams. While field studies have demonstrated the potential for acoustic tag systems, very little has been done to quantify the performance of these systems. This paper develops a method for predicting the accuracy of the position estimates provided by acoustic tag systems. General expressions are developed that can be applied to any particular deployment of a tag system. In addition, some specific examples are used to provide general guidelines that should be followed to achieve good performance when deploying an acoustic tag system.

11:30

1aAO11. An active fish tracking split-beam sonar to study salmon smolt behavior. John B. Hedgepeth (Tennera Environ., LLC, 3427-A Miguelito Court, San Luis Obispo, CA 93401) and Gary E. Johnson (BioAnalysts, Inc., Battle Ground, WA 98604)

An active fish tracking split-beam sonar (AFTS) is being used to study salmon smolt behavior at the Dalles Dam on the Columbia River. AFTS is based on the principle of tracking radar. Once a smolt is detected, two high-speed stepper motors align the axis of a digital split-beam transducer on the target. As the fish moves, deviation of the target from the beam axis is calculated and used to re-aim the transducer, thereby tracking the target. For each ping the target is tracked, data on fish position and target strength are recorded to disk. Fish position resolution is ± 0.5 cm within a sample volume approximately 14 m of range from the transducer. Individual fish tracks are visualized in three-dimensional plotting software. AFTS tracks are coupled with hydraulic data from a computational fluid dynamics model to determine statistical relationships between fish movements and flow conditions. Sequential fish positions are also analyzed using a Markov process. Fish states observed with AFTS include holding, swimming actively downstream, and passive drift. The information from AFTS will aid design of bypass systems to protect endangered and threatened salmon populations in the Pacific Northwest. [This research is funded by the Corps of Engineers.]

11:45–12:15

Panel Discussion

MONDAY MORNING, 4 DECEMBER 2000 PACIFIC SALON A, 7:55 A.M. TO 12:15 P.M.

Session 1aBB

Biomedical Ultrasound/Bioresponse to Vibration: Topical Meeting: Physics of Echo-Contrast Agents I

E. Carr Everbach, Chair

Department of Engineering, Swarthmore College, Swarthmore, Pennsylvania 19081-1397

Chair's Introduction—7:55

Invited Papers

8:00

1aBB1. Modeling of ultrasound contrast agents. John Allen (Dept. of Biomed. Eng., One Shields Ave., Univ. of California, Davis, Davis, CA 95616, jsallen@ucdavis.edu)

Ultrasound contrast agents differ from free-gas bubbles such that a polymer, lipid, or fluid shell encapsulates them. Many modeling efforts have focused on the role of the shell on the overall dynamics. Early efforts used semiempirical approaches to incorporate the shell into a generalized Rayleigh–Plesset equation. Church [J. Acoust. Soc. Am. **97**(3) (1995)] developed a more rigorous approach modeling the albumin shell of Albunex as a linear elastic material. This work has provided the foundation for many subsequent theoretical and experimental efforts. Ye [J. Acoust. Soc. Am. **100**(3) (1996)] examined modal scattering solutions for an Albunex agent. Recent efforts to design a high-frequency contrast agent based on dipole scattering are discussed [Allen, Kruse, and Ferrara, IEEE-Trans. Ultrason. Ferroelectr. Freq. Control, in press]. The modeling of different types of shell materials remains an outstanding issue. The potential role of both viscous and viscoelastic shell materials is outlined. Furthermore, the limitations of linear elastic and viscoelastic material models are highlighted. The modeling of possible interaction effects among groups of contrast agents has received less attention. Also, outstanding issues exist with respect to a rigorous mathematical modeling of contrast agent destruction, drug delivery schemes, and potential bioeffects.

8:15–9:15

Panel Discussion on Modeling of Echo-Contrast Agents

9:15–9:30 Break

9:30

1aBB2. *In vitro* experimental evaluation of contrast agents. Katherine Ferrara (Div. of Biomed. Eng., One Shields Ave., Univ. of California, Davis, Davis, CA 95616)

In this presentation, techniques for *in vitro* characterization of contrast agents will be reviewed. Contrast agents contain air or a lower diffusivity gas, with a shell that can be composed of albumin, polymer, a lipid monolayer, a surfactant or oil. The range of possible shell properties creates a wide range of physical properties, and resulting variations in behavior upon insonation. In order to characterize the agents, several methods have been employed, including acoustical and optical interrogation of single bubbles and ensembles. High-speed optical techniques can be employed to measure the radius-time curves, to determine the mechanisms of destruction, and to detect surface waves. The maximum radius and wall velocity, and mechanisms of destruction, are significantly affected by both the gas and shell properties. Observed mechanisms of destruction include static diffusion, acoustically driven diffusion and fragmentation. These destruction mechanisms are characterized by unique time scales which have been determined through both optical and acoustical methods. The results of acoustical techniques that have been applied to determine the linear resonance frequency and to quantify the scattering and attenuation cross sections will also be presented.

9:45–10:45

Panel Discussion on *In-vitro* Experimental Evaluation of Echo-Contrast Agents

10:45–11:00 Break

11:00

1aBB3. Imaging techniques for ultrasound contrast agents. Michalakis A. Averkiou (ATL Ultrasound, Philips Medical Systems, P.O. Box 3003, Bothwell, WA 98041-3003)

Contrast agents are used today to enhance and aid organ (myocardium, liver, brain, etc.) perfusion studies. From the early days of contrast imaging it was understood that microbubble specific imaging modalities would have to be invented to utilize the contrast effect. The two main characteristics of contrast microbubbles that have defined today's ultrasound modalities are (1) microbubbles are nonlinear scatterers that generate harmonic signals and (2) they exhibit a transient response, i.e., they are destroyed by ultrasound. Harmonic Doppler and imaging techniques were introduced at first to help with the detection of microbubbles in small vessels and capillaries. The harmonic response of bubbles and their Doppler signals increase with increasing acoustic amplitudes (mechanical index, MI), which results in fast bubble destruction. Triggered (intermittent) imaging, insonifying every n cardiac cycles (where $n = 1,2,3, \ldots$), was established to minimize destruction. Triggered imaging is a cumbersome technique. Real-time imaging techniques are now available. They are made possible by low-amplitude (MI) scanning and by more sensitive low MI techniques like pulse inversion and amplitude modulation. Both triggered and real-time techniques will be discussed in this presentation and clinical examples will be presented.

11:15–12:15

Panel Discussion on Imaging Techniques Using Echo-Contrast Agents

MONDAY MORNING, 4 DECEMBER 2000 PACIFIC SALONS C AND D, 9:00 TO 10:00 A.M.

Session 1aNSa

NOISE-CON and Noise: Plenary Session—Noise Control Engineering for the Airport Railway in Hong Kong

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

9:00

1aNSa1. Noise control engineering for the airport railway in Hong Kong—Setting the standards. Glenn Frommer (Mass Transit Railway Corp., MTR Tower, Kowloon Bay, Kowloon, Hong Kong, SAR, PROC)

The Lantau and Airport Railway (LAR) is a 34-km electrical railway that connects Hong Kong Island to the new International Airport. Noise control engineering was implemented throughout the design of the project to provide a safe, comfortable, and maintainable environment both for the passengers and railway neighbors in the most cost-effective manner as possible, and minimize land take and land sterilization nearby the railway. To do so required a unique balance of noise and vibration specification and performance in the following areas: (i) a wheel/rail interface design integrating synergistically, the track and rolling stock design, (ii) the use of a noise assurance plan to manage the noise design of the rolling stock, (iii) the use of aggressive specifications for noise emissions, reverberation times, and building services within stations, (iv) the use of RASTI prediction to ensure proper design and delivery of the public address system, (v) the use of floating track slabs and other trackforms to minimize the transfer of structure-borne noise from the railway platforms to the overlying developments, and (vi) specialist training in acoustics and vibrations. This paper will summarize the key results of the LAR project.

MONDAY MORNING, 4 DECEMBER 2000 PACIFIC SALON F, 10:20 A.M. TO 12:05 P.M.

Session 1aNSb

NOISE-CON, Noise and Architectural Acoustics: Heating, Ventilating and Air Conditioning Noise

Gary W. Siebein, Chair

Department of Architecture, University of Florida, P.O. Box 115702, Gainesville, Florida 32611

Contributed Papers

10:20

1aNSb1. Air handler sound power prediction method based on ARI Standard 260. Stephen J. Lind (The Trane Co., 3600 Pammel Creek Rd., La Crosse, WI 54601, slind@trane.com)

A method of predicting air handler sound power based on ratings for a product line is described. The method provides octave band sound power levels based on ratings obtained using Air-Conditioning and Refrigeration Institute (ARI) Standard 260 Sound Rating Of Ducted Air Moving And Conditioning Equipment. Detailed sound power information for HVAC equipment is not always available, but it is important in accurately predicting noise levels in acoustically sensitive spaces. To address this need, a rating program was undertaken using ARI 260. This standard is a reverberant room technique for sound rating ducted air conditioning equipment using a reference sound source substitution method. Since sound travels from the source to receiver along numerous paths, this standard differentiates between sound power emanating from several common paths called components. Components for this project included ducted discharge, free inlet plus casing, ducted inlet and casing. The standard provides guidance on adequate number of fan sizes, appurtenances and operating characteristics. The intent of the project was to provide a model to predict sound power by unit size, component, operating condition and unit configuration. Good agreement was found between predicted levels and measured data.

10:35

1aNSb2. Condenser fan noise reduction solutions. Michael Bobeczko (Sound International Technology, 19721 Torres Way, Portolla Hills, CA 92679)

There has been a surge in the building of multimillion dollar homes along the California Coast. Often, large commercial style air conditioners are installed with multiple axial flow propeller condenser fans. They can generate low-frequency tones that annoy highly sensitive neighbors. By replacing 3- to 4-bladed stamped metal fans with a 6-, 9-, and 12-bladed highly efficient airfoil fan with pitch compensation, by reducing motor speeds from 1140 to 870 rpm, and by installing a variable speed controller, sound levels were reduced 14 dB @ 77 Hz as well as reducing energy consumption. This noise reduction solutions have proven to be far more effective than traditional barriers.

10:50

1aNSb3. Sound attenuation of dual-wall, fiberglass-lined, circular ducts with and without airflow. Douglas Reynolds (Ctr. for Mech. and Environ. Systems Technol., Univ. of Nevada, Las Vegas, 4505 Maryland Pkwy., Las Vegas, NV 89154)

A major test program for measuring the sound attenuation characteristics of dual-wall, fiberglass-lined, circular ducts from six different duct manufacturers was completed. Octave band dynamic insertion loss values for octave band center frequencies from 63 to 8000 Hz were measured. The inside diameter of the ducts that were tested ranged from 12 to 48 in. Sound tests were conducted with and without airflow. Six airflow velocities from 1000 to 6000 fpm were examined. Regression analyses were conducted on the measured insertion loss values. Regression equations for the dynamic insertion loss values were developed that can be used to estimate the dynamic insertion loss of dual-wall, fiberglass-lined, circular ducts for zero flow and for duct airflow velocity from 1000 to 6000 fpm. The equations are valid for duct diameters from 12 to 72 in.

11:05

1aNSb4. A procedure for measuring background sound levels in rooms. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027) and Mark E. Schaffer (Schaffer Acoust., Inc., Pacific Palisades, CA 90272)

Background noise criteria for indoor spaces have been in existence for decades without an associated standard measurement procedure to verify compliance. The lack of a standard measurement procedure can lead to a significant difference in reported sound levels in a room, even from experienced professionals. The proposed procedure is intended for measuring background sound levels from continuous noise sources, and is being recommended to ASHRAE Technical Committee 2.6 for inclusion in the 2003 Applications Handbook. The purpose of the procedure is to standardize the instrumentation and method of determining the background sound levels. The procedure can be used by anyone charged with the task of comparing a room's background noise environment with established noise criteria.

11:20

1aNSb5. On HVAC duct acoustical end reflection. Emanuel Mouratidis (Vibroacoustics/BVA Systems Ltd., 727 Tapscott Rd., Scarborough, ON M1X 1A2, Canada, emouratidis@vibroacoustics.com) and Richard J. Peppin (Scantek, Inc., Silver Spring, MD)

Duct end reflection (ER) is the apparent loss of sound power resulting from an abrupt change in a cross-sectional area of the duct. In most references, the magnitude of ER is only a function of frequency and duct size, because of assumed plane-wave propagation. If, between the source and duct termination, there are short duct runs, 90-deg duct bends, offset ducts, T-sections, various end diffusers, high aspect ratio cross sections, the presence of airflow, etc., the relative effects on ER are unknown. The American Society of Heating, Refrigeration, and Air-Conditioning Engineer's Applications Handbook presents tables that provide the power loss due to ER. However, the assumptions used and the limitations of the table are lacking. This paper describes results of laboratory-measured changes in reverberation room sound-pressure level (SPL) created by common system variables that are associated with ER: duct length between noise source and reverb room termination, and several duct termination conditions were investigated using small diameter duct. In a controlled laboratory environment, these system conditions were found to create an apparent random variability (up to 10 dB) in the resulting SPL, for both a high impedance fan reference sound source and a low impedance loudspeaker source.

11:35

1aNSb6. Estimating community noise levels from outdoor condensing units. Gary W. Siebein and Hyeong-seok Kim (Dept. of Architecture, Univ. of Florida, P.O. Box 115702, Gainesville, FL 32611-5702)

Noise levels propagated from cooling towers and condensing units located in partial or fully enclosed equipment courtyards onto adjoining residential properties were studied in this paper. A combination of field measurements at existing installations, computer model studies of the installations and physical acoustical model studies were conducted to determine the acoustical effects of reflections from enclosing walls of nearby buildings and partial equipment yard enclosures on sound propagation at distances away from the units. Predictions using distance attenuation and the effects of partially enclosing walls (Harris, 1991; Miller, 1981) were used to estimate sound levels at various distances from several existing outdoor condensing units. Acoustical measurements were made at the actual installations within the chiller enclosure and at various distances in the surrounding communities. Each of the actual situations was simulated in model form at a scale of 1:10. The process of calibrating the model to the existing situations will be presented. The scale model data were within 2 dB of the actual insertion losses for full size octave bands from 63 to 2000 Hz for condensing units in the open and in various enclosures.

11:50

1aNSb7. Tips for specifying active duct silencers. Steve Wise (Wise Assoc., 1409 E. Skyline Dr., Madison, WI 53705, stevewise@att.net), Christian Carme (TechnoFirst, 48, Aubagne 13676, France), and Dwight Shackleford (American Aldes, Sarasota, FL 34234)

Low-frequency noise in buildings is receiving greater focus in new noise standards. The extraordinary performance of active duct silencers on these low frequencies provides a powerful new element for fan noise control. Acoustical consultants need to know the most appropriate applications for active duct silencers. The authors have been personally involved in the design and start-up of over 1000 active duct silencers installed in the USA and Europe to silence fans on a variety of HVAC applications. This site experience from office buildings, schools, hospitals, semiconductor manufacturing facilities, auditoria, sound test chambers, and cruise ships—both retrofit and new construction—is summarized to guide acoustical consultants in estimating the performance of active duct silencers. Fan and duct silencer system layouts are reviewed, along with performance case histories.

MONDAY MORNING, 4 DECEMBER 2000 PACIFIC SALONS C AND D, 10:20 TO 11:25 A.M.

Session 1aNSc

NOISE-CON and Noise: Classroom Acoustics

David Lubman, Chair

David Lubman and Associates, 14301 Middletown Lane, Westminster, California 92683-4514

Invited Paper

10:20

1aNSc1. Progress report on I-INCE technical initiative for schoolroom acoustics. David Lubman (David Lubman & Assoc., 14301 Middletown Ln., Westminster, CA 92683-4514), Louis C. Sutherland (Rancho Palos Verdes, CA 90275-3908), and Zerhan Karibiber (Yldz Tech. Univ., Istanbul, Turkey)

Good acoustics is an indispensable requirement for verbal learning, and therefore vital to all knowledge-based societies. The World Health Organization has identified the basic acoustical requirements for verbal learning spaces (Guidelines for Community Noise, April 1999). Some nations are well advanced in meeting these requirements while others may lack local guidelines or other resources. The worldwide acoustical design and noise control communities can help nations to achieve good schoolroom acoustics. A technical initiative for that purpose titled ''Noise and Reverberation Control for Schoolrooms'' was approved by the International-INCE General Assembly at its meeting in December 1999. The initiative is intended to be an internationally coordinated program to assist participating nations with engineering issues associated with achieving satisfactory acoustics in learning spaces. Representatives of various I-INCE member societies appointed to this Technical Study Group will be asked to submit a draft report for review by the I-INCE General Assembly within two years of their first meeting. The report is expected to provide practical guidance to school planners, designers, and code officials. The Group will first convene at the Internoise meeting at Nice, France in August 2000. This presentation highlights the results of that meeting.

Contributed Papers

10:40

1aNSc2. Determination of insertion loss for classrooms at a high school. Michael Greene (URS Greiner Woodward Clyde, 2020 E. First St., Ste. 400, Santa Ana, CA 92705)

The construction of a new freeway adjacent to an existing high school in eastern San Diego County, California, prompted the need for a rigorous analysis of the noise effects on the school. The insertion loss of structures (with windows and doors open and closed) at a high school was measured. The test setup consisted of two commercial-grade loudspeakers mounted atop a manually operated lift, associated amplifiers, pink-noise generator, a real-time noise analyzer, and sound-level meters. Noise levels were measured at equivalent distances outside and inside the rooms of interest, to derive the structure's insertion loss. This was done at incident angles of 30, 45, 60, and 75 deg to the building facade. The resultant data from these measurements required the use of specially designed spread sheets to effectively analyze the present the results. The results of the measurements

indicated that improvements to the older classrooms near the freeway would be necessary in order to meet the interior noise standard for classroom spaces.

10:55

1aNSc3. Rating and ranking university classrooms by acoustical quality. Murray Hodgson (UBC School of Occupational and Environ. Hygiene, 3rd Fl., 2206 East Mall, Vancouver, BC V6T 1Z3, Canada)

Nonoptimal classroom acoustical conditions affect speech perception and learning by students. The research reported here developed methods for predicting speech intelligibility in occupied classrooms. These were used to predict speech intelligibility at various positions in 280 university classrooms when 70% occupied, for two instructor voice levels. Classrooms were classified and rank ordered by acoustical quality, as determined by the classroom-average speech intelligibility. Classrooms with poor and bad acoustical quality were identified, so that they could be prioritized for acoustical renovation. The conclusion was that 90% of the 280 classrooms have good or very good acoustical quality with a typical instructor. However, 27 (10%) of the classrooms had poor quality, and one had bad quality. Most rooms were very good at the front and good or very good at the back. With a quiet female instructor, classrooms were poor or fair on average. Most were fair at the front, and fair, poor, or bad at the back. New classroom acoustical design and renovations should focus on limiting background noise from the ventilation systems. They should promote high instructor speech levels at the back of classrooms. This involves limiting the amount of sound absorption that is introduced into classrooms to control reverberation.

11:10

1aNSc4. A resource for creating learning environments with desirable listening conditions. Robin Glosemeyer (Russ Berger Design Group, 4006 Beltline Rd., Dallas, TX 75244), Benjamin Seep (Wrightson, Johnson, Haddon & Williams, Dallas, TX 75244), and Robert C. Coffeen (The Univ. of Kansas, Lawrence, KS 66045, rcoffeen@ukans.edu)

A discussion of this recent publication on classroom acoustics by the Technical Committee on Architectural Acoustics of the Acoustical Society of America which is directed towards school planners, school administrators, school faculty, governmental agencies, architects, and others who are engaged in the planning and construction of new classrooms and in the remodeling of existing classrooms. This publication is intended to be used as an aid in the understanding of the elements of desirable listening conditions in school classrooms.

MONDAY MORNING, 4 DECEMBER 2000 PACIFIC SALON E, 10:20 A.M. TO 12:05 P.M.

Session 1aNSd

NOISE-CON, Noise and Engineering Acoustics: Noise Control Materials and Elements

Glenn E. Warnaka, Chair
Future Technologies, 1612 South Allen Street, State College, Pennsylvania 16801

Contributed Papers

10:20

1aNSd1. A finite-element analysis of a smart foam system for passive/active noise control. Cliff L. Chin, Joshua T. Lee, and Gopal P. Mathur (The Boeing Co., 2401 E. Wardlow Rd., MC: C078-0420, Long Beach, CA 90807-5309)

The smart foam noise control system is designed to reduce sound by a combination of the passive absorbing component of the poroelastic foam and the active component of the voltage-driven PVDF (polyvinylidene fluoride) film. The hybrid noise control methodology allows the control of noise over a wide frequency bandwidth. In this paper, the finite-element (FE) method is used to solve the PVDF-foam-structure-acoustic coupled smart foam system in a two-dimensional domain. An FE model of a two-dimensional smart foam actuator with embedded PVDF film will be presented. The effect of foam properties on acoustic impedance and absorption coefficient of the smart foam system is examined using the FE model. The program also predicts the structural vibration of the foam matrix and the acoustic pressure in a cavity when an array of smart foam actuators are driven by control input voltages which are determined by the active control algorithm.

10:35

1aNSd2. Properties and applications of novel acoustical foams. Suresh Subramonian, Mae Drzyzga (Dow Chemical Co., 200 Larkin Ctr., Midland, MI 48674), Sandrine Gilg, Laurent Remy, and Chung Park (Dow Deutschland, Rhinemunster, Germany)

This paper describes the unique properties of a new line of novel acoustical materials that has been developed for sound and vibration management. The materials are macrocellular thermoplastic foams made from polyolefin resin blends. Fire retardant grades have been developed to meet a variety of building and construction fire codes. The foams have several advantages over incumbent materials, including structural integrity, moisture resistance, ease of fabrication/installation and recyclability. This paper also reviews a case study involving the use of the new foams to significantly dampen the sound level of a noisy industrial facility. The sound pressure levels and reverberation decay times were measured in the target area with an integrating, real time $\frac{1}{3}$-oct spectrum analyzer. Architectural acoustical analysis was performed using ray diagrams and Sabine calculations for determining the amount and optimal placement of the absorptive media to reduce the reverberant noise fields and sound reinforcing reflections. The thickness of the acoustic media was determined by matching the peak absorption frequencies of the foam to the peak noise frequencies of the source. After installation of the absorptive media as per specifications, the reduction in noise level was measured to validate the acoustical design.

10:50

1aNSd3. Elimination of structure-borne noise using an insulating material made from recycled rubber. Paul Downey (Dodge-Regupol, Inc., 33 Craighurst Ave., Toronto, ON M4R 1J9, Canada)

A brief overview of the concept of structure-borne noise will be given from both a theoretical and practical perspective. The methods and materials used to insulate several types of structures against vibrations and the resulting structure-borne noise will be explained. A case study approach will be used to demonstrate the results of these installations. The results will be indicated in the appropriate format, including the preceding and resulting sound pressure levels. The background technical data will be presented including the spring characteristics of the materials, the natural frequency, the dynamic stiffness, and the dynamic modulus of elasticity in dependence of the load. The behavior of the materials under load and at removal of load as well as the effect of layering materials will be examined. The required calculations to permit acoustical engineers to solve structure-borne noise problems will be illustrated.

11:05

1aNSd4. Geometric sound absorbers. Glenn E. Warnaka (Future Technologies, LLC, 1612 S. Allen St., State College, PA 16801-5915, future@vicon.net)

Acoustic absorption is a commonly used technique for noise reduction. Most acoustically absorbent treatments use fibrous or porous materials. In general, such treatments are not rugged and have limitations in many applications. For example, fibrous and porous materials can retain pollutants present in the environment. This leads to unsanitary and unhealthy conditions and to fire and explosion hazards. This paper describes novel geometric acoustic absorbers, based on closely coupled quarter-wave resonators that provide a high level of acoustic absorption over a broad range of frequency. Because the absorbers depend on their geometry for absorption, the absorbers may be made of any material and, hence, the material may be chosen on the basis of cost, environmental resistance, ruggedness, etc. Further, the absorbers can be cleaned when required to eliminate the presence of contaminants. Cleaning can be accomplished by conventional means such as washing, vacuuming, high-pressure air, etc. Two absorbers are described, Configuration ∇ and Flat Absorber. The absorbers may take many forms and may be made compact by folding them or stacking them in various ways without affecting their absorption. In other cases, the absorbers can be made very rugged to withstand heavy use.

11:20

1aNSd5. Broadband structural acoustic silencers. Sripriya Ramamoorthy and Karl Grosh (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48109-2125)

It has long been a goal of the noise control community to achieve broadband passive attenuation of sound propagating in waveguides. By using designs based on models of the mammalian cochlea, theoretical studies show that a structural-acoustic silencer can achieve the desired broadband transmission loss and anechoic termination is possible provided sufficient losses are present. The mammalian cochlea can be idealized as a compact rectangular duct system ($2.5\times0.1\times0.1$ cm) interacting with a strongly orthotropic plate of variable impedance (known as basilar-membrane). A coupled fluid-structure pressure-displacement wave propagates along the basilar-membrane resonating at a frequency-dependent location. In this process, the wave is attenuated with an accompanying reduction in duct pressure (nearly 100 dB over 20–20 000 Hz for humans). Current research intends to imitate this aspect of cochlear behavior for reducing noise level in pipes. Possible applications include use in automotive mufflers and quieting hydraulic systems. Design of fluid/structure interaction between the orthotropic plate and duct is based on numerical modeling using the 2.5-D finite element/analytic technique. Results based on interaction of a varying thickness plate made of typical orthotropic materials like unidirectional graphite-epoxy composite, with fluids like water and oil, are presented. Comparison is made against a constant thickness plate made of the same material.

11:35

1aNSd6. Wideband transmission noise reduction of smart panels featuring piezoelectric shunt circuits and sound absorbing material. Jaehwan Kim, Joong-Kuen Lee, and Jin-Young Choi (Dept of Mech. Eng., Inha Univ., 253 Yonghyun-Dong, Nam-Ku, Incheon 402-751, Korea, jaehwan@inha.ac.kr)

The possibility of a wideband noise reduction of piezoelectric smart panels is experimentally studied. A piezoelectric smart panel is basically a plate structure on which piezoelectric devices with shunt electronics are mounted and sound absorbing material is bonded on the surface of the structure. Sound absorbing material can effectively absorb the sound transmitted at midfrequency, while the use of piezoelectric shunt circuits can reduce the transmission at resonance frequencies of the panel structure. Piezoelectric shunting circuits, composed of register and inductor, are determined based on maximizing the dissipated energy through the circuit. To be able to reduce the sound transmission at low panel resonances, multimode damping with one piezoelectric device is essential and it is achieved by using the measured electrical impedance model. The noise reduction performance of the panels is tested on an acoustic tunnel. The tunnel is made of a guided tunnel that has a square cross section and a loudspeaker is used as a sound source. Panels are mounted in the middle of the tunnel and the transmitted sound pressures across panels are measured. Several panels according to different absorbing material thickness, combination of double panels, and location of piezoelectric devices are tested.

11:50

1aNSd7. Active control of sounds with large dynamic range. Anthony J. Brammer and George J. Pan (Inst. for Microstructural Sci., Natl. Res. Council, Montreal Rd., Ottawa, ON K1A 0R6, Canada)

The performance of an active noise control system employing digital signal processing is influenced by the analog signal amplitudes within the input and output analog-digital (A/D) and D/A converters, and hence is sensitive to the sound pressure being controlled. While compensation is commonly provided within the algorithm for variations in input power (e.g., by normalizing the adaptation step size), the full dynamic range of the A/D and D/A subsystems is usually not realized. An analog gain control system has been developed consisting of linked, reciprocal variable gain amplifiers, so arranged that the changes in signal amplitude at the A/Ds and D/As are smaller than the changes in sound pressure of the acoustic system being controlled. The gain control system operates on the error and secondary source signals so as to maintain the error path impulse response unchanged. In this way the gain changes are transparent to the digital controller. The application of the method to an adaptive feed-forward active noise control system for a circumaural hearing protector, or headset, will be described. [Work supported by the Defence and Civil Institute of Environmental Medicine.]

Session 1aSC

Speech Communication: All Things Prosodic (Poster Session)

Amy J. Schafer, Chair

Department of Linguistics, University of California, 3125 Campbell Hall, Los Angeles, California 90095-1543

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

1aSC1. The effect of syllabification and VOT on $F2$ onsets at the CV2 boundary. Golnaz M. Ghavami, Harvey M. Sussman (Dept. of Linguistics, Univ. of Texas, Austin, TX 78712), and Bjorn Lindblom (Univ. of Texas, Austin, TX 78712 and Univ. of Stockholm, Stockholm, Sweden)

The articulatory blending of vowels and consonants has long been described by a coproduction model of coarticulation [S. E. G. Ohman, J. Acoust. Soc. Am. **41**, 310–319 (1967)]. Variation in intergestural timing of the C closure has been shown to produce instability in locus equation coefficients as place descriptors when simulated on the DRM vocal tract model [Chennoukh *et al.,* J. Acoust. Soc. Am. **102**, 2380–2389 (1997)]. This study attempted to achieve real-world variations in CV phasing by imposing a variety of temporal intervals separating vowel-to-vowel sequences. Variation in syllable shapes and degree of aspiration were used to alter the relative timing between V1 and V2. A coproduction account predicts little change in $F2$ onsets for V2 as a function of the temporal interval separating V1 and V2. In contrast, a segment-by-segment (''deactivation'') account would predict systematic changes in V2 locus frequencies, with coarticulation decreasing as the C-interval increases. Three syllable shapes (V#CV, VC#V, VC#CV) were used with five vowel contexts /i,e,ae,u,ɔ/) surrounding the six stop consonants (/bdgptk/). Acoustic measures of both F2 offset frequencies for V1 as well as locus equations derived from the CV2 interface will be presented for three speakers of American English. [Work supported by NIH.]

1aSC2. The effect of stress on $F2$ onsets in VCV utterances. Ramin Sarraf, Harvey M. Sussman (Dept. of Linguist., Univ. of Texas, Austin, TX 78712, ramin.sarraf@mail.utexas.edu), and Bjorn Lindblom (Univ. of Texas, Austin, TX 78712 and Univ. of Stockholm, Stockholm, Sweden)

This study compares a superposition or vowel-to-vowel theoretical framework versus a segment-by-segment account to best account for the coarticulatory dynamics of VCV utterences. The ''trough effect,'' a deactivation of tongue musculature after the initial vowel of a symmetrical VCV sequence (e.g., /ibi/), is not readily explainable by traditional coproduction (vowel-to-vowel) accounts of coarticulation. A Jossian segment-by-segment programming sequence appears to be more compatible with ''deactivation'' phenomena. One way to manipulate extent of ''deactivation'' is to impose differential degrees of syllable stress on either side of the stop consonant. Locus equations can provide an acoustic window on this question as timing differences should show up as slope/intercept variations. Emphatic stress on $V1$ is expected to lessen deactivation relative to weak $V1$ stress, and consequently greater levels of anticipatory coarticulation should occur, resulting in higher locus equation slopes. Four different stress-vowel conditions were used: weak versus emphatic stress counterbalanced over $V1$ and $V2$. There were five vowel contexts for $V1$ and ten vowel contexts for $V2$, surrounding the three voiced stop consonants /b, d, g/. Results will be reported on three speakers of American English. [Work supported by NIH.]

1aSC3. Function word reduction: The role of adjacent stress. Lisa M. Lavoie (MIT, Rm. 36-511, 50 Vassar St., Cambridge, MA 02139, lisa@speech.mit.edu)

Monosyllabic function words in speech are often reduced from their full citation forms. Researchers have shown that reduction is influenced by part of speech, predictability, prosody, and disfluency. This work isolates the role of an additional factor, preceding stress, in function word reduction by analyzing the realization of ''for'' in a highly controlled corpus. Five American English speakers read pairs of disyllabic words with initial or final stress (BOOty, bouTIQUE) in the frame ''Please say [word] for me.'' Two speakers show striking sensitivity to preceding stress, always producing strong–weak alternations, such that strong ''for'' follows a weak syllable and vice versa. The rimes in the strong versions average 90 ms while the weak are just 50 ms. The strong rimes consist of a clearly segmentable vowel plus /r/. In the weak, however, /r/ is not distinguishable in the acoustics as its own segment; rather, it influences the vowel $F3$ value. The other three speakers always realize ''for'' as weak, with an average duration of 39 ms after either stressed or unstressed syllables. Though limited, these results suggest that some speakers construct rhythmic groupings in speech production. Further work will determine whether similar effects are found in spontaneous speech.

1aSC4. A role of fundamental frequencies in the perception of emphasized words. Kyoko Nagao and Shigeaki Amano (NTT Commun. Sci. Labs., Atsugi, Kanagawa, Japan, nagao@avg.brl.ntt.co.jp)

Many previous studies have shown the role of the fundamental frequency ($F0$) on focus structure in speech production. The current study examined the role of $F0$ on focus perception. Eleven subjects of Tokyo Japanese were asked to listen to 889 $F0$-manipulated sentences and to select the word that they thought the speaker emphasized. Each stimulus included a noun phrase such as ''W1-no W2,'' where both W1 and W2 are a noun with an accent on the first mora. The $F0$ contour of each stimulus was varied by changing values of $F0$ at the three points: the $F0$ minimum point (i.e., dip) in the noun phrase and the $F0$ maximum point (peak) in each word. The result showed that $F0$ functions as a phonetic cue for a listener to perceive the emphasis on the word. We found that the relative $F0$ differences between the two peaks largely determined the subject's focus perception. Furthermore, logistic regression analysis suggests that the dip plays an important role. We will discuss the relationship between the dip and peaks for the focus perception.

1aSC5. The effects of stress and final position on sounds of Jicarilla Apache. Siri Tuttle (UCLA Dept. of Linguist., 3125 Campbell Hall, Box 951543, Los Angeles, CA 90095-1543, stuttle@ucla.edu)

The study of word-level prosody in Athabaskan languages is complicated by morphology and intonation. Stems are both frequently stressed and frequently word-final or phrase-final. This means that final lengthening confounds acoustic measurements of stress. This study of Jicarilla (Eastern) Apache attempts to sort out the acoustic effects of phrase-finality from those of stress. Syllables taken from a recorded text were measured for duration of onsets, codas and peaks. Phrasal position was noted as initial/noninitial and final/nonfinal. When only nonfinal syllables are examined, stress is marked significantly by increased vowel length ($p=0.02$), controlling for phonemic vowel length. This distinction remains when stressed and unstressed final syllables are compared, but is weaker ($p=0.04$). When only stressed syllables are examined, vowels are significantly longer when the syllable is final ($p=0.0001$). However, length of codas is also significantly greater in final stressed syllables ($p=0.027$), and this effect is not associated with stress. Thus despite the large effect of final lengthening, Jicarilla Apache still distinguishes stressed from unstressed syllables using duration within syllables. It is the ratio of rhyme elements to one another which contains the cue.

1aSC6. Stress is phonetically reset word-internally in Squamish. Linda Tamburri Watt (Univ. of British Columbia, E270-1866 Main Mall, Vancouver, BC V6T 1Z1, Canada)

The purpose of this experiment was to test a phonologically based prediction suggesting that in Squamish the domain for stress is reset word-internally in certain morphologically limited cases. This is unusual because stress systems usually assign only one primary stress within a word. This prediction was tested by first identifying the phonetic correlates of stress in this language, then applying these criteria to this morphological context. One subject was tested by recording a list of stimuli containing morphologically simplex words of varying length and words containing the morphological structure of interest. Results of this study show, first, that length, $F0$, and amplitude are all indicators of primary and secondary stress and, second, that primary stress occurs twice within a single non-compound word.

1aSC7. Sentence intonation and syllable stress in speech produced during simultaneous communication. Robert Whitehead, Brenda Whitehead (Natl. Tech. Inst. for the Deaf, Carey-1309, Rochester, NY 14623-5604), Nicholas Schiavetti, Dale Evan Metz, and Deborah Gallant (State Univ. of New York at Geneseo, Geneseo, NY 14454)

This study investigated prosodic variables of syllable stress and intonation contours in speech produced during simultaneous communication. Simultaneous communication (SC) refers to speech combined with manually coded English (sign and finger spelling) for the production of each word of an utterance and is used in communication between hearing and deaf persons. Ten normal hearing, experienced sign language users were recorded under SC and speech-only (SO) conditions speaking a set of sentences containing stressed versus unstressed versions of the same syllables and a set of sentences containing interrogative versus declarative versions of the same words. Results indicated longer sentence durations for SC than SO for all speech materials. Vowel duration and fundamental frequency differences between stressed and unstressed syllables as well as intonation contour differences between declarative and interrogative sentences were essentially the same in both SC and SO conditions. The conclusion that prosodic rules were not violated in SC is consistent with previous research indicating that temporal alterations produced by simultaneous communication do not involve violations of other temporal rules of spoken English.

1aSC8. Articulation of word and sentence stress. Patricia A. Keating, Taehong Cho, Marco Baroni (Dept. of Linguist., UCLA, Los Angeles, CA 90095-1543, keating@humnet.ucla.edu), Sven Mattys, Lynne E. Bernstein, Brian Chaney (House Ear Inst., Los Angeles, CA), and Abeer Alwan (UCLA, Los Angeles, CA)

This paper reports on the articulation of two aspects of speech prosody: word and sentence stress. Word stress is examined in reiterant renditions of minimal pairs differing in the location of main stress (e.g., DIScharge–disCHARGE), with the contrastive stress difference manifested as differences in the location of the nuclear pitch accent. Sentence stress is examined in sets of minimal sentences differing in the presence or location of the nuclear pitch accent (e.g., so TOMMY gave Debby a song from Timmy). Three talkers read these materials while movements of the jaw and tongue (using the Carstens Articulograph) or of 20 reflective infrared markers on the face (using the Qualysis Motion Capture system) were recorded. Syllables with and without contrastive pitch accents are compared along several articulatory dimensions, including: jaw displacement and duration of movement, jaw peak opening and closing velocities, jaw peak acceleration, whole head displacement, and eyebrow displacement. These comparisons allow us to determine whether the talkers articulate the focused syllables/words differently. The three talkers were selected to differ in their overall visual speech intelligibility, as rated by expert deaf speechreaders, and therefore interspeaker differences in articulation are expected. [Work supported by NSF KDI Grant 9996088.]

1aSC9. Prosodic indication of syntactic structure in quasispontaneous speech. Amy J. Schafer (Dept. of Linguist., 3125 Campbell Hall, UCLA, Los Angeles, CA 90095-1543), Paul Warren (Victoria Univ. of Wellington, New Zealand), Shari R. Speer, and S. David White (The Ohio State Univ.)

This research reports sentence production and comprehension results that demonstrate that untrained speakers reliably use prosodic structures to resolve certain syntactic ambiguities in American and New Zealand English. Speakers played a board game that required them to repeatedly produce several types of syntactically ambiguous sentences as part of the game dialogue. Speakers were not informed of the ambiguities or given instructions about how to speak. $F0$ and durational measures of critical regions and phonological transcription (ToBI) data show that speakers produced a variety of intonational structures for each syntactic structure, but consistently produced prosodic phrasal boundaries that indicated the speaker's intended resolution of the syntactic ambiguity. Prosodic disambiguation occurred even when nonprosodic disambiguating information was available in the discourse setting, and even when speakers were not consciously aware of the ambiguity. Follow-up forced-choice perception experiments on utterance fragments excised from the conversational context show that a separate group of naive, untrained listeners reliably chose the speaker's intended meaning. The results indicate that prosody–syntax correspondences are pervasive in spoken sentences, and thus should be carefully addressed in psycholinguistic models. [Work supported by NIH Grants MH51768 and DC00029, NZ/USA Cooperative Science Programme Grant CSP95/01, and Marsden Fund Grant VUW604.]

1aSC10. Right boundary strengthening in American English. Stefanie Shattuck-Hufnagel (Speech Group, RLE, 36-511 MIT, 77 Massachusetts Ave., Cambridge, MA 02139, stef@speech.mit.edu)

A prominent theme in recent phonetic research is the role of constituent structure in governing the phonetic implementation of phonemic segments. For example, Fougeron and Keating (1997) describe an articulatory strengthening effect for some constituent-initial consonants which reflects the prosodic constituent hierarchy: utterance-initial /n/ is produced with greater tongue contact than phrase-initial /n/, which is in turn produced with more contact than word-initial /n/, etc. This articulatory strengthening is reflected to some extent in acoustic measures of duration. Is such strengthening limited to the left edge of constituents, or can it also occur at the right edge? Results from read laboratory speech show that the occurrence of aspiration in a word-final /k/ in American English is governed in part by the nature of the prosodic boundary following the word. Results for five speakers showed that (a) utterance-final /k/, as in ''Please say bake,'' was almost always aspirated (as determined by acoustic measures of the wave form); (b) word-final /k/ before a word-initial schwa, as in ''Please say bake again,'' was almost never aspirated, and (c) phrase-final /k/ before word-initial schwa, as in ''Please say bake, again,'' was sometimes aspirated and sometimes not. [Work supported by NIH DC02125-05.]

1aSC11. Accentual lengthening in Mandarin Chinese. Yiya Chen (Dept. of Linguist., SUNY at Stony Brook, Stony Brook, NY 11794-4376, yiyachen@yahoo.com)

This study investigated the effects of accentual lengthening under contrastive focus in Beijing Mandarin, a tone language, as a comparison to the results of similar studies in stress-accent languages. Two questions were asked. First, what is the domain of lengthening? Second, what are the structural influences on accent lengthening? Four different renditions of each test sentence were elicited from the subjects with contrastive focus on four different words to convey different pragmatic meanings. The results suggest: (1) the domain of accentual lengthening is primarily the focused word, with weak spread to the following word; (2) the spread of lengthening is sensitive to the morphosyntactic/prosodic relation of the focused word and the following word; and (3) while both onset and nucleus of the focused syllable exhibit lengthening, the spreading effect of lengthening is observed more reliably on nuclei than on onsets. These results agree to a large extent with studies on accentual lengthening in English and Dutch [A. Turk and L. White, J. Phonet. **27**, 171–206 (1999); T. Cambier-Langeveld and A. Turk, J. Phonet. **27**, 255–280 (1999)]. The similarity of accentual lengthening in Mandarin and stress-accent languages suggests that some effects of accentual lengthening may be universal. [Work supported by NSF No. SBR 9600930.]

1aSC12. Post obstruent tensing in Korean: Its status and domain of application. Sahyang Kim (Dept. of Linguist., UCLA, 3125 Campbell Hall, Box 951543, Los Angeles, CA 90095-1543, sahyang@ucla.edu)

The criteria that distinguish the phonological rules from the phonetic rules based on categoricality versus gradiency have drawn considerable attention (cf. Keating, 1990; Pierrehumbert, 1990). Korean Post Obstruent Tensing (POT) rule has been claimed, based on acoustic data, to be a phonological rule and that its domain of application is an accentual phrase (AP) (e.g., Jun, 1998). This paper questions the status of POT and its domain of application based on articulatory and acoustic data. By POT, a syllable initial lenis consonant (e.g., /t/, /s/) becomes tense (e.g., /t*/, /s*/, respectively) after an obstruent coda. It is known that lenis obstruents have a longer VOT, shorter closure durations, and shorter linguopalatal contact areas than tense obstruents. To investigate if POT is really a categorical process and if its application is definable in terms of an AP, acoustic and articulatory (i.e., EPG) data of derived and underlying tense obstruents were examined in four different prosodic positions: utterance-initial, intonational phrase-initial, AP-initial, AP-medial. Sentences containing a target sequence, /VpCV/, where C=/t/, /t*/, /s/, /s*/, were produced by two Korean speakers, at normal and fast speech rates. The results will shed light on the status of POT in the grammar of Korean.

1aSC13. Prosodic domain effects on noninitial syllables. Marie Huffman (Dept. of Linguist., SUNY at Stony Brook, Stony Brook, NY 11794-4376) and Jennifer Hataier (Southern Connecticut State Univ., New Haven, CT)

Previous literature has documented strengthening of segments adjacent to prosodic domain boundaries. This strengthening might be expected to affect only the segment next to the boundary. Yet, Fougeron and Keating [J. Acoust. Soc. Am. **101**, 3728–3740 (1997)] found that some noninitial vowels showed strengthening in a domain-initial syllable, as did some onset consonants in final syllables. To test whether effects of prosodic domain boundaries extend beyond the domain-adjacent syllable, consonant and vowel durations were measured for the second, stressed, syllable in the place names Apollo, Atami, and Acosta, comparing utterance-, intonation phrase- and word-initial position. Results for four female and one male speaker indicate that while there is rarely any effect on the consonant, the stressed vowel in the nonboundary adjacent syllable shows domain effects in more than half of the items tested, with longer durations in higher domains, particularly in Atami. These surprising nonlocal effects suggest that: (1) the place of articulation effect may be due to simple distance from the boundary, since [t]s were usually shorter than the other consonants; and (2) domain edge strengthening may involve a change in articulatory setting, the onset of which is controlled more carefully than the offset. [Work supported by NSF SBR9600930.]

1aSC14. An articulatory study of prosodically conditioned V-to-V coarticulation in English. Taehong Cho (Dept. of Linguist., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095)

Recent studies have indicated that vowels in prosodically strong positions (e.g., in stressed syllables and at edges of prosodic boundaries) are not only strongly articulated, but also resistant to coarticulation with neighboring vowels. This paper further examines vowel-to-vowel coarticulation in English by analyzing extensive articulatory data from six American English speakers, using the Electromagnetic Articulograph (EMA). It is hypothesized that vowels in prosodically strong positions are more resistant to coarticulation with their neighbors, and at the same time encroach more on their neighbors. To test this, sentences were designed so that they included /V1#bV2/ where V1 and V2 were manipulated, resulting in /a-a/, /i-i/ (control condition) and /a-i/, /i-a/ (test condition). Vowels also varied in sentence stress (accented versus unaccented) and in the intervening boundaries (# = Word, ip, IP). The vertical and horizontal positions of three tongue points and jaw are examined at five different points (onset, first quarter, middle, three quarters and end) in the vowel, to assess how much of the vowel articulation is anticipated or carried over at different points of the vowel. This shows variation in degree of V-to-V coarticulation under various prosodic conditions. [Work supported by NSF doctoral research grant.]

1aSC15. Relationships among temporal patterns of speech production in English. Bruce L. Smith (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2299 N. Campus Dr., Evanston, IL 60208)

Although various temporal patterns have been noted in English, possible relationships among these different characteristics generally have not been examined. In part, this may be because it is assumed that speakers show temporal patterns such as phrase-final vowel lengthening, vowel lengthening preceding voiced obstruents, etc., to essentially the same degree. However, recent findings suggest that there can be considerable variation among speakers in the extent to which they manifest such temporal properties. Thus, the present study investigated the possibility that subjects who exhibit a particular temporal property in their speech to a greater/lesser degree than other speakers might also tend to show other patterns to a greater (or lesser) extent. The same temporal characteristics were also examined to determine whether they varied as a result of speakers different syllable durations. A number of relationships among the various parameters were observed. For example, speakers with shorter syllable durations tended to show greater phrase-final vowel lengthening than speakers with longer syllable durations. In contrast, subjects with shorter syllable durations showed less vowel lengthening preceding voiced versus voiceless consonants than speakers with longer syllable durations. Possible reasons for these temporal variations and interrelationships will be considered.

1aSC16. Prosodic prominence on negation in various ''registers'' of U.S. English. Malcah Yaeger-Dror, Sharon K. Deckert, and Lauren Hall-Lew (Dept. of Cognit. Sci., Univ. of Arizona, Tucson, AZ 85721)

Since negatives express cognitively critical information, researchers have concluded that negatives ''should'' be prosodically prominent. Acoustic analysis of pitch tracks from isolated sentences [O'Shaughnessy and Allen (1982)] and the BU Radio News corpus [Hirschberg (1990)] support this theoretical claim. This study presents evidence of prosodic variation on ''not''-negatives used in a range of different social situations distinguished along Biber's ''register'' continuum from informative to interactive. The paper will show that both the relative quantity and the type of intonational prominence varies in different registers. Informative data discussed will include both LDC's BU Radio News and the Air Traffic Control corpus. Interactive corpora include both the Switchboard corpus of polite conversations between strangers and archived presidential debates. The percentage of prominent tokens is quite low in Switchboard, higher in debates, and much higher in the informative corpora. ToBI's H* category is dominant only in the most purely informative settings, with other contours preferred in interactive registers. [Work supported by NSF.]

MONDAY AFTERNOON, 4 DECEMBER 2000 SCHOONER/SLOOP ROOMS, 1:20 TO 5:00 P.M.

Session 1pAO

Acoustical Oceanography: Bioacoustics II

Timothy K. Stanton, Chair

Department of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Bigelow 201, Woods Hole, Massachusetts 02543-1053

Chair's Introduction—1:20

Invited Paper

1:30

1pAO1. A mini-tutorial on zooplankton acoustics. Peter H. Wiebe (Woods Hole Oceanogr. Inst., Redfield Bldg., Woods Hole, MA 02543)

The techniques for use of high-frequency acoustical systems for the study of zooplankton distribution, biomass, abundance, size distribution, and behavior have been the focus of intense effort and development during the past two decades. Significant theoretical and experimental progress has been made in understanding how these animals scatter sound. A review of the major aspects of this problem will be presented, including current field survey techniques, instrumentation, laboratory measurements, and scattering models. Issues such as confounding factors in data interpretation and determination of model parameters will be discussed.

Contributed Papers

2:30

1pAO2. Three-dimensional acoustic scattering models for elongated fluid-like zooplankton. Andone C. Lavery, Dezhang Chu, Duncan E. McGehee, and Timothy K. Stanton (Woods Hole Oceanogr. Inst., Appl. Ocean Phys. and Eng., Woods Hole, MA 02543)

Accurate acoustic scattering models that correctly incorporate organism size, shape, material properties, and orientation are a critical component for the remote classification and detection of marine organisms, such as fish and zooplankton. Zooplankton that fall into the elongated fluid-like category are of particular importance, primarily due to their high natural abundances. In order to obtain a better understanding of the scattering properties of this class of organism, extensive laboratory measurements of the frequency (50 kHz to 1 MHz) and angular (all angles in two planes with 1-deg resolution) characteristics of the acoustic backscattering from live individual decapod shrimp (*Palaemonetes vulgaris*) have been performed. An acoustic scattering model based on the distorted wave Born approximation (DWBA) has been developed for these elongated fluid-like organisms. This model makes use of progressively more realistic and complex representations of the animal shape. The most sophisticated of these representations involves fully three-dimensional digitizations of the animal exterior, obtained using high-resolution computed tomography (CT). A primary focus of this study is to determine the practical conditions under which it is necessary to make use of such high-resolution, computer-intensive digitizations of animal shape.

2:45

1pAO3. Multistatic multifrequency scattering from zooplankton. Charles F. Greenlaw, D. Van Holliday, and Duncan E. McGehee (Bae Systems, 4669 Murphy Canyon Rd., San Diego, CA 92123)

Multifrequency backscattering measurements can be inverted to estimate size abundance of zooplankton if a suitable model for backscattering is known. Physics-based models generally include the physical properties (density and compressibility) as input parameters. Computer manipulations of one such zooplankton scattering model suggest that physical properties as well as size might be estimated from simultaneous multistatic and multifrequency scattering measurements. Preliminary results from laboratory measurements of silicone rubber targets will be presented and extraction of physical properties discussed. [Work supported by ONR Code 322 BC.]

3:00

1pAO4. Seasonal variations of density and sound-speed contrasts of the krill *Euphausia pacifica*. Tohru Mukai, Kohji Iida, Hisayuki Mikami, and Ryuichi Matsukura (Grad. School of Fisheries Sci., Hokkaido Univ., 3-1-1 Minato-cho, Hakodate, Hokkaido, 041-8611 Japan)

In recent years, acoustic technology has been used extensively to estimate krill abundance. Acoustic assessment is required to gain a precise estimate of krill target strength. However, predictions of target strength from theoretical scattering models are often influenced by the swimming angles and density of the krill, and sound-speed contrasts between krill and seawater. Density and sound-speed contrasts are known to show annual cycles. In this study, seasonal variations of the specific density and sound-speed contrasts of *Euphausia pacifica* are presented. Biological sampling was carried out during twilight, when the sound-scattering layer migrates up to the surface. The specific densities of *E. pacifica* were measured, using a series of variable density glycerol solutions, within 48 h of net sampling. Sound-speed measurements were performed 2 hours after net sampling, using a T-shaped velocimeter with two transducers mounted on the ends of a horizontal tube. In 1999, the seasonal changes in specific densities and sound speed contrasts were ca. 1% and 3%, respectively. When calculated using Stanton's straight cylinder model, these seasonal changes yielded a difference of approximately 5 dB in the target strength of *E. pacifica*.

3:15–3:30 Break

3:30

1pAO5. Broadband echo spectra from euphausiids and copepods. Kenneth G. Foote (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, kfoote@whoi.edu), Tor Knutsen (Inst. of Marine Res., N-5024 Bergen, Norway), Philip R. Atkins, Claire Bongiovanni, David T. I. Francis (Univ. of Birmingham, Birmingham B15 2TT, UK), Peter K. Eriksen, Mette Torp Larsen, and Tom Mortensen (RESON A/S, DK-3550 Slangerup, Denmark)

A seven-octave-bandwidth echo sounding system [J. Acoust. Soc. Am. **105**, 994 (1999)] was used to observe diverse zooplankton along the coast of western Norway during the period 28 April–9 May 1999. Echo spectra were obtained from freshly caught, swimming specimens of the euphausiid *Meganyctiphanes norvegica* and copepod *Calanus finmarchicus ex situ* in a tank mounted on the stern deck of R/V JOHAN HJORT. Echo spectra were also obtained from scatterers *in situ* at 15-m depth in the same sea areas where the experimental animals were caught. Pulse compression was used to reduce the risk of analyzing overlapping echoes. All echo spectra were absolute, as the system was calibrated by the standard-target method using a 10-mm-diameter sphere of tungsten carbide with 6% cobalt binder. Comparison of echo spectra over the nominal bandwidth 1600–2600 kHz supports the claim of acoustic classification. Additional evidence is provided by reference to modeling computations [J. Acoust. Soc. Am. **105**, 1050 (1999); **105**, 1111 (1999)] where the bodies were assumed to be weakly scattering and fluid-like, with representative morphometries and assumed or measured properties of mass density and longitudinal-wave sound speed. [Work supported by the EU through RTD Contract No. MAS3-CT95-0031 during the project.]

3:45

1pAO6. *In situ* target tracking of individual krill (*Meganyctiphanes norvegica*) in the Oslo fjord. Anders Røstad (Oregon State Univ., Hatfield Marine Sci. Ctr., 2030 SE Marine Science Dr., Newport, OR 97365, anders.roestad@noaa.gov) and Stein Kaartvedt (Univ. of Oslo, Blindern, N-0316 Oslo, Norway)

Krill (*Meganyctiphanes norvegica*) were studied during and after their diel vertical migration to near-surface water using a Simrad EK500 echosounder with a split-beam 120-kHz hull mounted transducer. Target tracking of individual krill provided information on target strength, swimming velocity, and 3-D swimming patterns through the acoustical beam. Both prospects and limitations of the split-beam target tracking technique in behavioral studies are discussed.

4:00

1pAO7. Three-dimensional acoustic tracking of krill with a multibeam sonar. Alex De Robertis, Chad Schell, and Jules S. Jaffe (Scripps Inst. of Oceanogr., La Jolla, CA 92093, aderobertis@ucsd.edu)

The swimming behavior of individual zooplankton mediates how the animals experience their spatially heterogeneous environment, and consequently, has an important effect on population dynamics. However, current understanding of zooplankton swimming behavior is limited by conventional zooplankton sampling techniques. Here, we describe the use of a high-resolution multibeam sonar [Jaffe *et al.*, Deep Sea Res. **42**, 1495–1512 (1995)] to reconstruct the swimming trajectories of individual krill (*Euphausia pacifica*) in Saanich Inlet, British Columbia. The instrument was deployed in the deep scattering layer at depth during the day, and in near-surface strata during twilight periods of vertical migration and at night. Successive acoustically determined animal positions from these records are linked using a simple target tracking algorithm resulting in $>10^4$ trajectories of several seconds duration. The spatial positions of the tracked targets are improved by applying an algorithm [J. S. Jaffe, J. Acoust. Soc. Am. **105**, 3168–3176 (1999)] developed to localize targets within the acoustic beams. These methods permit quantitative analyses of swimming behavior of undisturbed krill in their natural environment for the first time.

4:15

1pAO8. Experimental verification of an algorithm for animal localization using a multibeam sonar system. Jules S. Jaffe, Chad Schell, and Alex De Robertis (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

Behavioral inferences of zooplankton activity from data generated with active multibeam sonar systems require high precision in localization due to the small extent of the animal movements. If target localization is limited to whether a target is within one beam or another, sonars with many beams and high complexity are required. Sonars which have multiple beams with measurable side lobes have the capability of achieving adequate resolution in situations of high signal to noise. In Jaffe [J. Acoust. Soc. Am. **105**, 3168–3175 (1999)], a maximum likelihood estimator for position was proposed. The algorithm utilizes a minimum mean square estimator in amplitude in order to estimate target position to a resolution smaller than the sonar's beamwidth. Experimental verification of the application of this algorithm to data aquired in a test tank with simultaneous verification of position with an optical imaging system is presented. The methodology employed uses a pair of cameras in order to simultaneously estimate the three-dimensional location of a target which is simultaneously being observed with a three-dimensional multibeam acoustical system [Jaffe *et al.*, Deep Sea Res. **42**, 1495–1512 (1995)]. The results indicate that the algorithm can produce an accurate estimate of three-dimensional target positional information providing accurate calibration is performed.

4:30–5:00

Panel Discussion

Session 1pBB

Biomedical Ultrasound/Bioresponse to Vibration: Topical Meeting: Physics of Echo-Contrast Agents II

E. Carr Everbach, Chair
Department of Engineering, Swarthmore College, 500 College Avenue, Swarthmore, Pennsylvania 19081-1397

Invited Papers

1:30

1pBB1. Ultrasound bioeffects associated with contrast agents. Diane Dalecki (Dept. of Elec. and Computer Eng., Univ. of Rochester, Rochester, NY 14627, dalecki@ece.rochester.edu)

Ultrasound contrast agents are stabilized microbubbles used to enhance diagnostic imaging. Various studies indicate that certain bioeffects can result from the interaction of ultrasound with tissues containing contrast agents. The presence of ultrasound contrast agents in blood has been shown to increase the extent of hemolysis resulting from exposure to pulsed ultrasound *in vitro* and *in vivo*. The presence of contrast agents in tissues also increases the susceptibility of many tissues to damage from pulsed ultrasound or lithotripter fields. For example, injection of Albunex during exposure of mice to a piezoelectric lithotripter field with amplitude of only 2 MPa produced hemorrhages in most soft tissues, such as fat, muscle, kidney, stomach, and bladder. Contrast agents in the vasculature can also decrease the threshold for the production of premature cardiac contractions resulting from exposure to pulses of ultrasound. This lecture will provide an overview of some bioeffects of ultrasound in tissues containing ultrasound contrast agents.

1:45–2:45

Panel Discussion on Bioeffects of Echo-Contrast Agents

2:45–3:00 Break

3:00

1pBB2. Sonoporation and ultrasound transfection assisted by contrast agents. Junru Wu (Dept. of Phys., Univ. of Vermont, Burlington, VT 05405, jwu@zoo.uvm.edu)

Gene therapy provides potentially effective treatment of a wide variety of diseases. Generally speaking, all gene therapy protocols require introduction of foreign nucleic acids, most often DNA, into the target cells and tissue, where gene expression is induced. However, low permeability of tissue and cells causes efficient barriers to entry of foreign DNA. Therefore, gene therapy requires transfection techniques that promote the uptake and expression of foreign DNA by a cell. There are two types of transfection techniques—viral and nonviral. Among the nonviral transfection methods, sonoporation and ultrasound-induced transfection are relatively new methods. Recent *in vitro* studies [Bao *et al.*, Ultrasound Med. Biol. **23**, 953–959 (1997); Greenleaf *et al.*, *ibid.* **24**, 587–595 (1998); Ward *et al.*, J. Acoust. Soc. Am. **105**, 2951–2957 (1999)] have shown that the presence of contrast agents dramatically enhances the efficiency of the transfection and also reduces the required acoustic pressure. This presentation will focus on reviewing the recent publications of sonoporation and ultrasound transfection using contrast agents. The possible physical mechanisms involved and main advantages of this technique will be discussed.

3:15–4:15

Panel Discussion on Novel Uses of Echo-Contrast Agents

4:15–4:30 Break

4:30

1pBB3. Future directions in ultrasound contrast agent research. Charles C. Church (Acusphere, Inc., 38 Sidney St., Cambridge, MA 02139, church@acusphere.com)

Since the introduction of Levovist® in Europe and Albunex® in the United States, interest in ultrasound contrast agents has increased enormously. This change is reflected in the ensuing rapid pace of technical development, and the depth of the discussions during this topical meeting. Before Albunex®, it was essentially impossible to convince an equipment manufacturer to devote time to the development of contrast-specific imaging modalities. After Albunex®, harmonic gray-scale imaging was in the clinic within the year, followed by harmonic color Doppler imaging, power Doppler, pulse inversion gray scale, pulse inversion Doppler, power modulation, coherent contrast imaging, and so on. The trend continues today as old techniques are improved, new techniques (e.g., subharmonic imaging) are tested and new approaches (e.g., phase-shift imaging) are conceived. Similarly, rapid advances have occurred in other areas, with models now including the details of shell viscoelasticity, *in vitro* studies now imaging single bubble dynamics to microsecond resolution, and specially designed microbubbles being used for targeted imaging, drug delivery, and gene therapy. Even bioeffects research has ''advanced'' with the discovery of imaging-modality-dependent contrast agent effects such as induction of premature ventricular contractions. Selected aspects of each of these areas will be discussed.

4:45–5:45

Panel Discussion on Modeling of Echo-Contrast Agents

MONDAY AFTERNOON, 4 DECEMBER 2000 PACIFIC SALON D, 1:20 TO 2:20 P.M.

Session 1pNSa

NOISE-CON and Noise: Plenary Session—Product Sound Quality From Perception to Design

John J. Van Houten, Chair
J. J. Van Houten and Associates, 3320 East Chapman Avenue, #323, Orange, California 92869-3811

1:20

1pNSa1. Product sound quality: From perception to design. Richard H. Lyon (RH Lyon Corp, 691 Concord Ave., Cambridge, MA 02138)

This discussion of product sound quality is concerned with the relation between the work of product designers and the perceptions of consumers/users regarding the acceptability of the sound. Designers make choices regarding structure, materials, and components in a product. The tools they need should allow them to anticipate the effect of those choices on sound quality. This presentation recounts the role of psychoacoustics in product design and product acceptability and notes the results of that work in metrics for sound quality and consumer/user perceptions about the product. The successes and drawbacks of this activity are noted. Recent work on a new paradigm, using sensory profiles for sound as an intermediary between metrics and perception, is described along with results using this procedure. Future developments of the procedure are outlined. [Work supported in part by National Science Foundation Dynamic Systems and Controls Program and by RH Lyon Corp .]

MONDAY AFTERNOON, 4 DECEMBER 2000 PACIFIC SALON F, 2:35 TO 5:10 P.M.

Session 1pNSb

Noise, NOISE-CON and Engineering Acoustics: Performance Assessment of Acoustical Test Rooms

Kenneth A. Cunefare, Chair
School of Mechanical Engineering, Georgia Institute of Technology, Graduate Box 286, Atlanta, Georgia 30332-0405

Chair's Introduction—2:35

Invited Papers

2:40

1pNSb1. Qualifying a reverberation room for sound absorption measurements. Alf Warnock (Inst. for Res. in Construction, Natl. Res. Council, Ottawa, ON K1A 0R6, Canada)

ASTM C423, the standard method for measuring sound absorption in a reverberation room, was recently revised extensively and now includes tests to verify that the room has acceptable acoustical properties. Further changes are planned by the ASTM task group. This paper will present sets of measurements made in the reverberation rooms at the National Research Council Canada illustrating the effects of some physical factors, such as number of loudspeakers, number of microphones, the need for rotating diffusers, and the number of specimen positions that should be used for different specimens. Practical problems facing laboratory operators will be discussed. The precision required for measurement in a room will be determined to some extent by the reproducibility of the test method. There is little point in reducing the precision of the measurement in individual rooms once it is significantly less than the test reproducibility. Further changes that might be made to C423 will be presented.

3:00

1pNSb2. Dynamic insertion loss measurements of sound attenuators at low frequencies. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027) and James D. Partain (PCI Industries, Inc., Fort Worth, TX 76111)

For decades the duct silencer industry has relied on ASTM E 477 as a standard for measuring the acoustic and aerodynamic properties of passive duct silencers. Historically, there has been significant concern regarding the accuracy and reliability of the measured dynamic insertion loss values for duct silencers at 125 Hz and below. The uncertainty at low frequencies has been blamed on several factors including: (1) inaccurate determination of sound level in reverberation chambers due to inaccurate spatial averaging (e.g., low modal density), and (2) varying acoustic output power from the electronic signal source due to a different impedance seen by the loudspeaker (with and without the test specimen in the duct). This paper presents a third reason for the low-frequency variability, which is likely to be more significant than all others combined. It will show that the problem is caused by acoustic resonances in the empty test duct, and these resonances will occur in every facility that is constructed in accordance with the test standard. The authors will present test data collected in a NVLAP accredited laboratory verifying this low-frequency effect, and also recommend specific changes to ASTM E477 which could eliminate this problem.

3:20

1pNSb3. Anechoic wedge design and development/anechoic chamber qualification testing. Alan Eckel (Eckel Industries, 155 Fawcett St., Cambridge, MA 02138, ae@eckelacoustic.com) and Istvan Ver (Eckel Industries, Stowe, MA 01775)

The initial research conducted at Harvard University's Electro Acoustic Laboratory for the development of the anechoic chamber will be reviewed in this presentation. These findings, which were originally presented in the Office of Scientific Research and Development National Defense Research Committee Report No. 4190 (OSRD, No. 4190), will be discussed. The research conducted by Leo L. Beranek and his colleagues during WWII established the basic design and construction for anechoic wedges and chambers for years to come. Contemporary design and materials used in anechoic wedge construction have evolved to encompass a variety of room acoustic treatments, to include fiberglass, foam, and metallic wedges. Design and construction of current commercially available chambers and wedges will be reviewed. The second half of the presentation will focus on the development of anechoic and hemi-anechoic chamber qualification procedures to current ISO and ANSI standards. The use of various sound sources will be discussed as well as the correlation between these standards and other test methods in determining chamber performance.

3:40

1pNSb4. Experiences with anechoic chamber qualification per ISO 3745 and ASA/ANSI S12.35. Kenneth A. Cunefare (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

ISO 3745 and ASA/ANSI S12.35 contain recommended procedures for the performance qualification of anechoic and semianechoic rooms. This paper presents experience with a number of different methodologies and analyses used in a series of tests configured to comply with these specifications. The methods include discrete measurements at fixed separation distances with multiple microphones, measurements taken with a single microphone repositioned between each measurement, and a computer-controlled continuous traverse system. Experience with different instrumentation chains will be briefly addressed. Also, a comparison of test results using pure tone and pink noise excitation will be presented. Finally, an algorithm will be described which, for a given continuous traverse test data set, determines the best acoustic source center and source offset distance to minimize the deviation from the permissible bounds contained within the specifications.

4:00

1pNSb5. On sound power determination: The uncertainty in sound power determined in a reverberation room based on a round-robin. Richard J. Peppin (Scantek, Inc., 916 Gist Ave., Silver Spring, MD 20910, peppinr@asme.org)

In this paper I will present the results of an extensive round-robin sponsored under the NIST National Voluntary Laboratory Accreditation Program for Acoustics. The round-robin tested the interlaboratory variability of sound power measured using the reverberation room sound power standards ANSI S12.31-33 and ISO 3741-3743. Also I will discuss the ramifications of uncertainty on sound-pressure estimation.

4:20

1pNSb6. Research into quality assessment methods for anechoic chambers. Francis Babineau, Jr. and Brandon Tinianow (Johns Manville, 10100 W. Ute Ave., Littleton, CO 80127, babineau@jm.com)

The goal in qualification testing for any anechoic or hemi-anechoic chamber is to define a volume within which acoustic measurements can be performed accurately in a free-field condition. In this study, both commonly accepted and novel techniques are employed to assess the acoustical performance of a hemi-anechoic chamber and ultimately its relevancy for qualified testing. Two recognized international test methods are discussed, as well as various alternate methods. The alternate methods include variations on the existing standards, utilizing acoustic intensity measurements and using an alternate sound source. The alternate sound source more closely approaches actual sound power testing conditions, in that it does not have a well-defined acoustic center. Considerations for performance assessment include impact on sound power measurements, ease of assessment, and comparisons of results to those of accepted methods. Recommendations for improvements to contemporary assessment techniques as well as future work will also be presented.

Contributed Papers

4:40

1pNSb7. Design and construction of a convertible hemi/anechoic acoustical laboratory for testing space flight hardware at the NASA Glenn Research Center. Beth A. Cooper (NASA John H. Glenn Res. Ctr. at Lewis Field, 21000 Brookpark Rd., M.S. 86-10, Cleveland, OH 44135, beth.a.cooper@grc.nasa.gov)

At the NASA Glenn Research Center, science experiment payloads developed for the International Space Station are designed to meet noise emission criteria that promote hearing conservation and speech intelligibility goals. A key component in the successful design of low-noise hardware is the frequent use of acoustical testing as a diagnostic and design verification tool. NASA has designed and constructed a convertible hemi/anechoic acoustical testing laboratory that provides convenient access to accurate and repeatable sound pressure level measurements as well as state-of-the art capabilities for sound power level testing per ANSI S12.35 using a multichannel PC-based data acquisition system. The laboratory consists of a 100-Hz hemianechoic chamber, vibration-isolated to 3 Hz, with $21' \times 17' \times 17'$(h) interior working dimensions and removable floor wedges. A separate, sound-isolated control room doubles as a test support equipment enclosure when testing flight hardware that requires remote connections to high-noise support equipment and utilities. Movable modular furnishings facilitate reconfiguration of the enclosure, and data acquisition and test control functions are easily relocated to an adjacent quiet area. The Glenn Research Center Acoustical Testing Laboratory also offers a full range of noise control services to assist payload developers with meeting the ISS noise emission criteria.

4:55

1pNSb8. Noise and vibration isolation design for a vibration testing laboratory. David A. Nelson (Nelson Acoustics Engineering, Inc., P.O. Box 879, Elgin, TX 78621) and Thomas Stewart (Cisco Systems, Inc., San Jose, CA 95134-1706)

Cisco Systems new Engineering Approvals Center provides an example of the successful application of noise and vibration control principles in a mixed-use building. The Vibration Laboratory portion of the building is designed for maximum throughput and comprises a number of powerful noise and vibration sources, including four hydraulic shaker tables with hydraulic pumps, an electrodynamic shaker table, two drop shock tables, and a package drop area. Because office spaces, conference facilities and sensitive high-performance EMC chambers are located directly above and adjacent to the Vibration Laboratory, a considerable degree of isolation was designed into the structure to preserve an officelike environment in these areas. Design and installed performance of critical building elements will be discussed, including the isolated slab and vibration breaks, noise-isolating control room, walls, windows, doors and ceiling.

MONDAY AFTERNOON, 4 DECEMBER 2000 PACIFIC SALON D, 2:35 TO 5:05 P.M.

Session 1pNSc

Noise and NOISE-CON: Noise Standards—Challenges to Quieter Products

Robert D. Hellweg, Jr., Chair

Compaq Computer Corporation, MR 01-3/D3, 200 Forest Street, Marlborough, Massachusetts 01752

Chair's Introduction—2:35

Invited Papers

2:40

1pNSc1. Noise labeling for consumer products. Paul Schomer (USA-CERL-CNN, P.O. Box 9005, Champaign, IL 61826-9005, schomer@uiuc.edu)

Currently, ISO 4871-1996 (Acoustics–Declaration and verification of noise emission values of machinery and equipment) uses A-weighted or spectral sound power levels as metrics of choice for noise labeling. Also included is a measure of the likely variations in levels that result from machine to machine variability, measurement to measurement variability, and site to site variability. This labeling scheme is adequate to label machinery for use in commercial or industrial settings. Here, plant engineers or audiologists, etc., can readily interpret and use data. But a two-number label using the sound power level in decibels is not optimum for conveying information to the consumer. This paper suggests an alternate form for the data. Sound power is still the physical quantity measured, and, as suggested by others, a one-number system is used. The statistical variation is included by reporting the value that will not be exceeded in 19 out of 20 measurements. But the metric reported is one that is roughly proportional to judgments of loudness, and, as such, should provide greater meaning and clarity to the user. Hopefully, with comprehensible noise labeling, the consumer will be able to include noise in his/her choice of products.

3:00

1pNSc2. The cylindrical microphone array: A proposal for use in international standards on sound power level measurement. Matthew A. Nobile, Jennifer A. Shaw (IBM Hudson Valley Acoust. Lab., Bldg. 704, M.S. P226, 2455 South Rd., Poughkeepsie, NY 12603), and Brian Donald (9 Creek Bend Rd., Poughkeepsie, NY 12603)

To determine the sound power level of a stationary noise source, a hypothetical measurement surface is defined which envelops the source under test. The sound pressure level is measured at selected microphone positions on this measurement surface, and the sound power level computed. A previous paper [M. A. Nobile and J. A. Shaw, Inter-Noise **99**, 1535–1540] introduced the concepts of the cylindrical microphone array and cylindrical measurement surface and presented results from initial measurements. The array was suggested as a more practical alternative to the current parallelepiped or hemispherical arrays for measuring the sound power level of sound sources in hemi-anechoic chambers according to standards such as ISO 3744, or the Information Technology and Telecommunications industry test code ISO 7779. It was also suggested that for certain types of machines, such as tall, rectangular computer frames, or other equipment with similar aspect ratios, the cylindrical measurement surface may yield results that are more accurate. In this paper, the results of further experiments and measurements using the cylindrical array will be presented, along with a proposal for incorporating this new array into the international standards on sound power level measurements.

3:20

1pNSc3. Method for evaluating the attenuation of acoustical covers used in high-end computer systems. Jennifer A. Shaw, Matthew A. Nobile (IBM Hudson Valley Acoust. Lab., Bldg. 704, M.S. P226, 2455 South Rd., Poughkeepsie, NY 12601), and Brian Donald (Poughkeepsie, NY 12603)

High-end computer systems, such as those used in the Information Technology and Telecommunications (ITT) industry, have become increasingly noisy in recent years, due to the amount of airflow needed to keep them functioning properly. As sound power levels increase, the systems run the risk of exceeding their noise-limit specifications. One solution is to incorporate acoustical covers as a noise control measure. By attenuating the amount of noise emitted from the machine, the overall measured sound power level will decrease, thereby allowing the system to pass specifications. However, a method for evaluating the amount of attenuation these acoustical covers provide is needed. To date, the attenuation has been measured by using actual system frames that are under development, which are often hard to acquire and introduce too many variables into the measurement. Also, both front and rear covers are needed for the measurements, so that the attenuation of just one cover is not easily obtained. This paper proposes the use of a test box as an evaluation tool for measuring the attenuation of a single cover. The test box provides a uniform and repeatable method that yields a more useful metric for rating cover performance.

3:40

1pNSc4. A comparison of two methods for the evaluation of prominent discrete tones: Phase 3. Anne C. Balant (State Univ. of New York at New Paltz, HUM 18, Ste. 6, New Paltz, NY 12561 and IBM Hudson Valley Acoust. Lab., balant@us.ibm.com), Robert D. Hellweg, Jr. (Compaq Computer Corp., Marlboro, MA 01752), and Matthew Nobile (IBM Hudson Valley Acoust. Lab., Poughkeepsie, NY 12601-5400)

This paper presents results of Phase 3 of the study of prominent discrete tones by the Inter Committee Working Group (ICWG) from Information and Technology Equipment (ITTE). Progress and earlier results were reported at INTER-NOISE 99, the 139th Meeting of the Acoustical Society of America, and INTER-NOISE 2000. Phase 3 is a round robin involving 40 signals (some from ITTE products and some artificial sounds), which were recorded and distributed to participants as .wav files. Each of several ITTE company laboratories determined the objective tone-to-noise ratios and prominence ratios of the signals and/or provided subjective ratings. The paper presents comparisons of the objective and subjective ratings from the different laboratories. The relative success of the two objective methods in predicting the subjective ratings is discussed, as are the underlying issues and recommendations of the task group.

4:00

1pNSc5. Computer sound quality: Masking, loudness and prominence. David A. Nelson (Nelson Acoustics Engineering, Inc., P.O. Box 879, Elgin, TX 78621)

Computers and business equipment were one of the first applications for standardized sound quality metrics. Various product specifications (emanating primarily from Europe) defined criteria for A-weighted sound pressure and sound power levels as well as the absence of prominent tones and impulsive noise. Standards such as ISO 7779, ANSI S12.10 and ECMA 74 have provided the means for assessing these quantities. Of particular interest at the moment is the prominent tone, for which there are two accepted metrics: tone-to-noise ratio and prominence ratio. Experience has shown that both metrics occasionally fail to track listeners perception of the prominence of tones or tonal complexes. An alternate prominence analysis is constructed based on psychoacoustical factors, using a method proposed by Zwicker for identifying the loudness of any sound masked by another. The accepted metrics are compared and contrasted with the masked loudness method using carefully selected test cases.

Contributed Papers

4:20

1pNSc6. Sound quality of contemporary personal-computers. Ya-Wen Chou and Ming-San Cheng (D300, Bldg. 64, No. 195-6, Sec. 4, Chung Hsing Rd., Chutung Hsinchu, Taiwan 310, ROC, gloria@itri.org.tw)

Development of sound quality analysis in product design and marketing has drawn much attention in recent years. Many makers nowadays would like to know their products sound quality and to improve them instead of just reducing the overall sound pressure level. The noise of personal-computers is characterized as either rumble or containing pure tone components, while sound quality analysis may reveal additional information on the market acceptance and the noise influence to customers. The prevailing industrial test standards for product rating of these products, however, have not yet included the concept of sound quality. To investigate the possibility and appropriateness of adopting sound quality analysis in the test standard to rate product noise, we used many personal-computers in accordance with prevailing test standards such as ISO 7779. The results were then analyzed and indexes of sensory pleasantness were calculated by sound quality parameters of loudness, sharpness, roughness and tonality as suggested by Zwicker and Fastl. Comparison between sensory pleasantness values and sound power level indicates a large discrepancy in terms of product rating. It is suggested that the rating of these products by sensory pleasantness may be considered in the future.

4:35

1pNSc7. Noise radiation from a riding mower. Christian M. Skinner (Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804, skinner@sabine.acs.psu.edu) and Courtney B. Burroughs (Penn State Univ., State College, PA 16804)

Riding mowers are a significant contributor to community noise. This has prompted many countries to develop standards to reduce the noise emissions from riding mowers. This research will look at the individual noise sources involved with the riding mower; these are the engine, exhaust, cooling fan, the cutting blades, and the mowing deck housing. The relative contributions to the overall noise from each component were determined by isolating each component during outdoor measurements of the sound power levels. To investigate the mechanisms of the noise generated by the rotating mower blades, a test bed was built to hold the mowing deck and electric motor. The motor was isolated from the test bed via isolation mounts to provide a quiet source of power for the mower deck. Pressure sensors have been embedded into the leading edge of the blades and also at the grass exit to measure the disturbances as the blades and deck interact with each other. Results from these measurements are presented and discussed.

4:50

1pNSc8. International space station acoustics. Jerry R. Goodman (ISS Acoust. Lead, NASA-Johnson Space Ctr., 1814 San Sebastian Ln., Houston, TX 77058, jerry.r.goodman@jsc.nasa.gov)

Control of acoustics is important in the International Space Station (ISS) because of the long-term crew exposure to noise and their confinement. ISS requirements cover modules, government furnished equipment, and payloads. The current status of compliance to these requirements will be described, including examples of where acoustics efforts have aided compliance and where difficulties were experienced. Concerns with obsolete requirements, and lack of acoustic experience and rigor to achieve compliance, will be discussed.

MONDAY AFTERNOON, 4 DECEMBER 2000 PACIFIC SALON E, 2:40 TO 5:25 P.M.

Session 1pNSd

NOISE-CON and Noise: Community Noise and Community-Noise Barriers

Louis C. Sutherland, Cochair
27803 Longhill Drive, Rancho Palos Verdes, California 90275-3908

Brigitte Schulte-Fortkamp, Cochair
Department of Physics/Acoustics, Oldenburg University, Oldenburg D-26111, Germany

Contributed Papers

2:40

1pNSd1. Community noise at Del Mar, California, during two four-day helicopter noise studies. Robert W. Young (710 W. 13th Ave., E-6, Escondido, CA 92025-5511, exy2@worldnet.att.net)

Monday, 30 August 1999, was the first day of a four-day Marine Corps demonstration of noise of helicopters flying north and south along the coast at Del Mar, California. The second four-day demonstration was 26–29 October 1999. A Computer Engineering Limited automatic noise monitor operated continuously at 421 Ocean View Ave. or at 12839 Via Grimaldi. Helicopters flew 0.5- to 5-mile intervals offshore, at altitudes 500 or 2000 feet above sea level. On 2 September 1999, at 421 Ocean View Ave., from midnight to 0800, the one-hour average sound level was less than 41 dB. From 0800 until 0951, helicopter sound levels exceeded an 80-dB threshold 15 times. During the hour ending at 1000, helicopter one-hour average sound level was 72 dB. The maximum fast A-weighted sound level MXFA was 94 dB. The A-weighted sound exposure level ASEL was 103 dB. The 421 Ocean View site is 300 feet east of the shoreline and about 200 feet above sea level. Thus when a helicopter flies 0.5 miles offshore and altitude 500 feet above sea level, it is only 300 feet above and 3500 feet distant from the monitor at 421 Ocean View Ave.

2:55

1pNSd2. The analysis of the environmental influence of drilling installation. Wieslaw Wszolek, Jan Macuda, and Tadeusz Wszolek (Univ. of Mining and Metallurgy, 30-059 Krakow, Poland)

Drilling work, due to its character, settlement, and whole-day service, has potential acoustic danger for the environment. The degree of influence depends on many factors, e.g., the type of drilling equipment, applied acoustic screens, localization of drilling and its surroundings, the type of drilled-trough rocks, and applied drilling technology. The main sources of acoustic noise are motors, pumps, and current generators. The distinction can be made between some procedures that are more noisy than the normal drilling process, for example, tripping. Usually the drilling rig is poorly equipped with noise reduction installation. Experiments were carried out on eight types of drilling assemblies. The received results show that different drilling equipment has an individual influence on the environment. The evident proof of that is the azimuthal acoustic characteristic and additionally the noise level dependence from the cycle of drilling work. The level of LAeq in the surroundings of the examined drilling rig has been made. It has been proved that, due to the noise reduction, the selection of optimal settlement of the drilling rig and optional equipment is possible. In this optimalization process, the cycle of drilling work has been considered. This reduction is possible without additional acoustic protection.

3:10

1pNSd3. Noise produced by religious pilgrimages in Monterrey, Mexico. Fernando J. Elizondo-Garza and Jose de J. Villalobos-Luna (Acoust. Lab., FIME-UANL, P.O. Box 28 ''F,'' Cd. Universitaria, San Nicolas, 66450, N.L., Mexico, fjelizon@ccr.dsi.uanl.mx)

In this paper are shown the result of measurements of the noise produced by religious pilgrimage in the context of the traditional ''Virgin of Guadalupe'' Christian celebration in the city of Monterrey, Mexico. The noise implications are discussed and conclusions are presented.

3:25

1pNSd4. Regulation of amplified sound sources. Eric Zwerling (Rutgers Noise Tech. Assistance Ctr., 14 College Farm Rd., New Brunswick, NJ 08901)

Few sources of noise generate more complaints than amplified music, which can present a difficult enforcement profile if the appropriate approach is not employed. Highly amplified vehicular sound systems are a nuisance increasing in both occurrence and intensity, requiring an enforcement standard that is objective, content-neutral and easily employed by an officer in a police cruiser. A ''plainly audible'' standard meets these criteria and has withstood court challenges. The specific language for inclusion into an ordinance is presented. The regulation, enforcement and curtailment of low-frequency nuisances arising from fixed sources such as bars or parties may prove intractable for performance codes setting A-scale permissible sound level limits. One possible approach, octave band regulation, is not cost-effective and technically feasible for most enforcement agencies which desire efficiently rapid response. An effective approach is a C-scale regulatory protocol prohibiting an amplified sound reproduction device from raising total sound levels within the dwelling of a complainant by specific time-indexed levels. There are technical and evidentiary benefits to taking these measurements within the residence, and not at the property line. This new approach has been recently adopted by St. Augustine, FL and Lafayette, LA. The ordinance language is presented.

3:40

1pNSd5. Verification of sound levels inside discotheques and public show sites. Alberto de Leo, Franco Giuliani, Concetta Fabozzi, and Renzo Tommasi (Natl. Agency for Environ. Protection of Italy (ANPA), Via V. Brancati 48, 00144 Rome, Italy)

In 1997 a decree was issued in Italy with the aim of defining the limits of noise sources inside discotheques and public show sites. ANPA (Italian Agency for Environmental Protection) has started research on about 30 sites in order to know the real levels of exposures to noise inside different types of public show sites and, after, carry out a first verification on proposed limits and methods, and also evaluate the possibility of using the automatic control systems stated by the rules. Such campaigns enable us to obtain elements that are very useful for the development of sectoral rules and for the planning of interventions.

3:55

1pNSd6. Creating noise sources for rotorcraft and aircraft noise simulation models. Kenneth J. Plotkin (Wyle Labs., 2001 Jefferson Davis Hwy., Ste. 701, Arlington, VA 22202, kplotkin@arl.wylelabs.com)

Modern computing power has made noise simulation models, i.e., prediction of actual time histories of sound levels, practical. Switzerland uses its FLULA model for airport noise contours. NASA's new RNM rotorcraft model uses simulation to address the complexities of rotorcraft noise. The Wyle/USAF NMSIM fixed-wing model has been used to address significant effects associated with complex terrain. A requirement for this type of model is the three-dimensional noise source properties of the aircraft being analyzed. This information is generally extracted from fly-over noise tests. Because noise is needed on a complete sphere (or at least a hemisphere), parts of the noise sphere are associated with long propagation distances and/or grazing incidence to measurement positions. This paper reviews the practical problems associated with turning fly-over measurements into source models. Examples from experience with RNM and NMSIM are presented.

4:10

1pNSd7. Rotorcraft noise model (RNM) and acoustic repropagation technique (ART2) validation and application using CH-146 (Bell 412) measurement data. Juliet A. Page and J. Micah Downing (Wyle Labs., 2001 Jefferson Davis Hwy., Ste. 701, Arlington, VA 22202)

A series of acoustic measurements were conducted at Moose Jaw Canadian Forces Base, Saskatchewan, Canada in June, 1998 for obtaining detailed noise data on a large array of ground- and crane-based microphones for a CH-146 helicopter. The project, conducted by the North Atlantic Treaty Organization (NATO) Committee on the Challenges of Modern Society was to define test procedures and analysis methods for the standardization of a NATO rotorcraft noise database. Four independent microphone arrays were deployed by the USAF, NASA, RAF, and DERA. Data from the USAF and NASA arrays were used to validate the Rotorcraft Noise Model (RNM). USAF data were processed via an updated acoustic repropagation technique (ART2) to define the rotorcraft broadband sound hemispheres. The source noise was then propagated to the measurement locations using RNM. Agreement with the original USAF measurements demonstrated consistency of the ART/RNM process. Agreement with the independent NASA measurements provided validation of the RNM as a prediction tool.

4:25

1pNSd8. Investigation of noise reflections using test signals. Lloyd Herman, Jeff Giesey, and Jeremy Ghent (Ohio Univ., 141 Stocker Ctr., Athens, OH 45701)

Repeatable test signals for use with a loudspeaker system were generated to investigate noise reflections from traffic noise barrier walls. Four classes of signals were considered: impulses, maximum-length sequences, pseudorandom binary sequences, and linear frequency-modulated sine waves or chirps. Overall, the chirps yielded the most information from initial experiments. The test system was applied to single and parallel barrier configurations to identify the number of significant reflections, to determine the level of the reflections relative to the direct signal and to quantify the contribution of the reflections to the overall sound level at receivers of interest. For receivers located on the community side of parallel barrier configurations the contribution of the reflections was found to increase with receiver distance from the barrier.

4:40

1pNSd9. Field measurements of a reduced-scale noise barrier's insertion loss and absorption coefficients. Todd A. Busch and Ramon E. Nugent (Acentech, Inc., 1429 Thousand Oaks Blvd., Ste. 200, Thousand Oaks, CA 91362)

In situ testing determined the insertion loss (IL) and absorption coefficients of a candidate absorptive barrier material as part of a project to abate railway noise in Anaheim, California. A 13 000-ft barrier is proposed south of the tracks, but residents of Yorba Linda to the north have had concerns that barrier reflections will increase their noise exposure. To address this issue, a 12-ft-high by 48-ft-long demonstration barrier was built in the parking lot of Edison Field, Anaheim, as part of a public open hours, thereby allowing for acoustical measurements. The barrier's insertion loss (IL) was measured in one-third-octave bands assuming 1/2-scale construction. The IL for three (scaled) railway noise sub-sources (rail/wheel interface, locomotive, and train horn) was measured at six (scaled) distances. The highest total A-weighted IL, after corrections for finite barrier and point-source speaker effects, was 22 dB(A) for rail/wheel noise and 18 dB(A) for locomotive noise. Absorption coefficients were measured using a time-delay spectrometry technique, and were found to be much less than the reverberation room results advertised in the manufacturer's literature, though the resulting increase for noise exposure to Yorba Linda residents is still expected to be less than 1 dB(A).

4:55

1pNSd10. A correction to Maekawa's curve for the insertion loss behind noise barriers. Penelope Menounou (Dept. of Mech. Eng., The Univ. of Texas, Austin, Austin, TX 78712-1063, menounou@mail.utexas.edu)

Maekawa's curve is one of the most established methods for predicting the insertion loss (IL) behind barriers. For the simple case of a barrier modeled as a half plane, the IL is given versus a single parameter, the Fresnel number ($N1$). Predictions obtained by Maekawa's curve deviate largely from experimental data and predictions obtained by analytical solutions when the receiver is either close to the barrier or at the boundary separating the illuminated from the shadow zone. It is shown that if a second Fresnel number ($N2$) is appropriately defined, the IL obtained by the rigorous analytical solutions can be expressed versus $N1$ and $N2$ for several types of incident radiation (plane, cylindrical, and spherical). Accordingly, the single curve in Maekawa's chart can be replaced by a family of curves. Each curve corresponds to a different $N2$ and provides the IL vs $N1$. The Kurze–Anderson formula (a mathematical expression of Maekawa's curve) is also modified to describe this set of curves. The new chart gives predictions that agree well with analytical solutions and experimental data for all receiver locations. Moreover, unlike Maekawa's, this chart can be used for line, point, and plane wave sources. [Work supported by ATP.]

5:10

1pNSd11. Random and periodic square wave barriers in noise control. Nesrin Sarigul-Klijn and Dean Karnopp (Professor and Co-Director of Transportation Noise Control Ctr. (TNCC), Dept. of MAE, UC Davis, Davis, CA 95616-5294, nsarigulklijn@ucdavis.edu)

Federal regulations and increased public interest in reducing transportation noise have led to the construction of miles of sound walls. Most of these walls are almost perfectly non-absorptive barriers with the shortest possible diffractive edge; a straight line. A number of alternate strategies have been studied in recent years, including T-top, Y-top and random edged barriers.Prior investigations by other researchers suggest that it might be possible to improve the performace of a barrier without increasing the average height by introducing a jagged profile. This is because the jagged geometry on the edge of a sound wall alters the sound pressure level in the shadow zone by causing the region of the barrier nearest the receiver to admit multiple paths with variable phase. The direct waves from the diffracting edges of the barrier and waves subsequently reflected from the ground plane are superposed at the receiver causing constructive or destructive interference at the receiver. This is not easily amenable to analytical methods. This led to anechoic chamber testing of the various practical jagged edge treatments. In this paper, theoretical and experimental investigation of the effectiveness of more practical random and square wave barriers are detailed. The TNCC Anechoic chamber experimental facility is utilized for one-sixth scaled barrier tests.

MONDAY AFTERNOON, 4 DECEMBER 2000 PACIFIC SALON B, 2:00 TO 3:35 P.M.

Session 1pSA

Structural Acoustics and Vibration: Methods for Control of Vibration and Radiation

Timothy W. Leishman, Chair

Department of Physics and Astronomy, Brigham Young University, Provo, Utah 84602-4673

Chair's Introduction—2:00

Contributed Papers

2:05

1pSA1. Active noise control studies using the Rayleigh–Ritz method. Senthil V. Gopinathan, Vasundara V. Varadan, and Vijay K. Varadan (CEEAM, Penn State Univ., University Park, PA 16802)

The Rayleigh–Ritz method is used to model a smart composite plate and the piezoelectric patches attached to it. Classical plate theory is used to model the composite plate and electroelastic theory is used to model the segmented piezoelectric patches. Eigenfunctions of a clamped–clamped beam are used as the Ritz functions for the panel and the rigid cavity modes are used to model the acoustic cavity. The dynamic equations of motion for the coupled smart panel-cavity system are derived using Hamilton's principle. The forcing term due to the cavity acoustic pressure is determined by using virtual work considerations. Five collocated pairs of actuator/sensor pairs are attached to the plate at a predetermined placement scheme. A (multi-input–multi-output) MIMO controller is designed for the coupled vibroacoustic system. The output feedback controller is then employed to emulate the optimal controller by solving the Riccati equations. The performance of the control system in reducing the vibration and the noise transmitted into the enclosure is studied. It is shown that the Rayleigh–Ritz procedure can be used more efficiently than the finite-element method for cabin noise control problems.

2:20

1pSA2. Physical limits on the performance of active vibration isolation systems. Vyacheslav M. Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA 92606)

Two fundamental issues in active vibration isolation are optimal actuators/sensors placement and optimal control algorithms. In this work, a general approach to active vibration isolation is pursued, with the intent of finding the limitations on the isolation performance that would be valid for all possible active force distributions independent of the particular control algorithms employed. The limiting performance is estimated in terms of isolation efficiency, frequency domain, and work performed by active forces on system displacements, considered a measure of the control effort. These estimates are derived for a general multi-DOF system and expressed as exact inequalities in explicit analytical form. The results show that physical limitations do exist on the efficiency of a generic active vibration isolation system. The principal differences in this respect are discussed between free and supported systems, absolute and relative motions, force and kinematic excitations. Limiting performance estimates can help set design goals for isolation systems. Active force distributions emerging from limiting performance analyses assist in finding rational system configurations and optimal controls. Minimum active work, ex-

pressed in terms of passive subsystem parameters, provides a criterion for optimization of a structure as a potential host for an active vibration control system.

2:35

1pSA3. Active control of structurally radiated sound. Jin Seok Hong (Dept. of Precision Mech. Eng., Grad. School, Hanyang Univ., 17 Haengdang-Dong, Sungdong-Ku, Seoul, Korea, 133-791), Heung Seob Kim (Georgia Inst. of Technol., Atlanta, GA 30332-0405), Kihong Shin, and Jae-Eung Oh (Hanyang Univ., Sungdong-Ku, Seoul, 133-791 Korea)

In this paper, active control of radiating sound (using active structural acoustic control) from a vibrating rectangular plate by a steady-state harmonic point force disturbance is experimentally studied. Two piezoceramic actuators are bonded directly to the panel surface to provide control input structure. The sensor consists of two accelerometers mounted on the plate. Estimated radiated sound signals using a vibro-acoustic path transfer function are used as error signals. The vibro-acoustic path transfer function represents a system between accelerometers and microphones. The controller is a 2 by 2 filtered-x LMS algorithm implemented on a TMS320C30 DSP. The results demonstrate good control of sound radiation using the vibro-acoustic path transfer function.

2:50

1pSA4. Noise reduction in a launch vehicle fairing using actively tuned loudspeakers. Jonathan D. Kemp (Dept. of Mech. Eng., Duke Univ., Durham, NC 27708) and Robert L. Clark (Duke Univ., Durham, NC 27708)

Loudspeakers tuned as optimal acoustic absorbers can significantly reduce damaging, low-frequency, reverberant noise in a full-scale launch vehicle fairing. Irregular geometry, changing payloads, and the compliant nature of the fairing hinder effective implementation of a passively tuned loudspeaker. A method of tuning the loudspeaker dynamics in real time is required to meet the application requirements. Through system identification, the dynamics of the enclosure can be identified and used to tune the dynamics of the loudspeaker for passive dissipation of targeted, high-intensity, low-frequency modes that dominate the acoustic response in the fairing. A loudspeaker model with desired dynamics serves as the reference model in a control law designed to tune the dynamics of a nonideal loudspeaker to act as an optimal tuned absorber. Experimental results indicate that a tuned loudspeaker placed in the nose cone of the fairing dissipates significant acoustic energy, up to 12 dB at the targeted modes, and verifies results calculated from the simulation. [Work supported by AFOSR F49620-98-1-0383 under the supervision of Dr. Daniel J. Segalman.]

3:05

1pSA5. Tuned passive vibration suppression using linear electromechanical coupling. Michael J. Anderson, Tony J. Anderson, William Zornik, Christopher Hocut (Dept. of Mech. Eng., Univ. of Idaho, Moscow, ID 83843-0902), and Jonathan D. Blotter (Idaho State Univ., Pocatello, ID 83209)

One proposed method to suppress vibrations of a mechanical structure is to couple the energy of vibration to an external electric shunt circuit. Coupling of the mechanical energy to the electric circuit is accomplished with a piezoelectric transducer attached to the structure. Significant amounts of mechanical energy can be dissipated by the shunt, but a large inductor is required. In this presentation, we describe an alternative approach that replaces the shunt with a linear passive electric circuit that is connected to another piezoelectric transducer attached to the structure. Even though the electric circuit does not contain an inductor, it is still possible to dissipate significant amounts of mechanical energy in the connecting circuit. The mechanism of energy dissipation in the electric circuit is similar to, but not identical to, that exploited by a tuned electric shunt. Inductance, however, is provided by mutual coupling of piezoelectric transducers on the structure.

3:20

1pSA6. Passive vibration suppression using nonlinear electromechanical coupling. Tony J. Anderson, Michael J. Anderson, Christopher Hocut, William Zornik (Dept. of Mech. Eng., Univ. of Idaho, Moscow, ID 83843-0902), and Jonathan D. Blotter (Idaho State Univ., Pocatello, ID 83209)

The development of a new class of passive vibration and acoustic suppression systems is presented. The approach is to transfer the energy from one mechanical system to another using reversible piezoelectric transducers connected with a passive electric circuit. Response of the first system is suppressed while exciting the response of the second system. The two mechanical systems can be physically separated structures, or different mechanical modes of a single structure. The passive electric circuit is a network containing diodes and/or transistors switched from the piezoelectric voltages. Reductions of 25% in the response of the directly excited system have been shown. This approach is an improvement over a passive shunt technique in that the typically heavy inductor in the electrical shunt is replaced with an existing system mechanical impedance. It also has the advantage over active control techniques in that an external power source is not required and there is no possibility to add energy to the system from the controller.

Session 1pSC

Speech Communication: Learning and Cognitive Processing (Poster Session)

Richard S. McGowan, Chair
CReSS LLC, 1 Seaborn Lane, Lexington, Massachusetts 02420

Contributed Papers

All posters will be on display from 2:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:00 p.m. to 3:30 p.m. and contributors of even-numbered papers will be at their posters from 3:30 p.m. to 5:00 p.m.

1pSC1. The segmental representation of consonant clusters: Constituents, cohabitants, or unitary clusters. Kathleen M. Measer and James R. Sawusch (Dept. of Psych., Park Hall, SUNY at Buffalo, Buffalo, NY 14260)

What is the form of the segmental representation that underlies auditory word recognition? In spoken word recognition, different segmental representations have been proposed for words. Most models of word recognition assume some form of phonetic representation. In a phonetic representation, consonant clusters such as the /spl/ in ''splash'' are treated as a sequence of separate, discrete elements. However, the precise nature of this representation is unknown. Clusters could be composed of abstract phonemes, context-sensitive allophones, or some other unit. A form-based priming paradigm was used with naming, lexical decision, and A–X name matching tasks to investigate the representation of syllable initial clusters. Primes that differ from the targets in the number and order of consonants (in syllable initial clusters) were presented to listeners. For example, primes such as /skrav/, /skav/, /krav/, /sav/, /kav/, /rav/, and /bav/ (control) were used with the target /skrem/. If clusters are composed from their constituent phonemes, the number of phonemes in common between prime and target should determine the magnitude of priming. The results will be discussed in terms of their implications for the segmental representation of speech. [Work supported by NIDCD Grant R01DC00219 to the Univ. at Buffalo.]

1pSC2. Acoustic and perceptual speaker normalization. Patti Adank (Dept. of Linguist., Univ. of Alberta, AB T6G 2E7, Canada), Roeland van Hout (Nijmegen Univ., 6500 HD Nijmegen, The Netherlands), and Roel Smits (Max Planck Inst., 6500 AH Nijmegen, The Netherlands)

An attempt was made to evaluate the performance of several methods for speaker normalization in both the acoustic and the perceptual domain of speech. This was done by comparing the acoustic distributions applied to the same vowel data and further by comparing the acoustic distributions to the perceptual distributions of the vowel data. The normalization methods included, among others, extrinsic (e.g., z-score transformation) and intrinsic methods (Bark transformation), and formant weighting ($F2$) and correction ($F3-F2$). To obtain the acoustic distributions, the normalization methods were applied to $F0$ and formant data from monophthong vowels in /sVs/ context of male and female speakers of Standard Dutch. The perceptual distributions were obtained through an experiment with phonetically trained listeners, whose task was to judge each vowel's height, place of constriction and amount of rounding/spreading. The acoustic and perceptual distributions were compared using correlational- and cluster-analysis techniques. When describing the results of these tests, the focus will be on the extent to which the variation and overlap are the same in the acoustic and perceptual domains, and which normalization methods show a pattern of overlap and variation most similar in both domains. [Work supported by The Netherlands Organization for Research (NWO).]

1pSC3. Talker voice and similarity affect lexical neighborhoods. Liza K. Zimack, James R. Sawusch, Kathleen M. Measer, Paul A. Luce (Dept. of Psych., Park Hall, SUNY at Buffalo, Buffalo, NY 14260, lkz2@acsu.buffalo.edu), and Rochelle S. Newman (Univ. of Iowa, Iowa City, IA 52242)

Words that are similar to a syllable can influence listeners perception of phonemes in the syllable. Known as the lexical neighborhood effect, this has been shown to be robust, but vary in magnitude across talkers. In previous studies [L. Zimack and J. Sawusch, J. Acoust. Soc. Am. **107**, 2918 (2000)], series of nonword syllables were tested in two voices. In one voice, an effect of neighborhood on phonetic perception was clearly found. In the other voice, this effect was absent. The pattern of results suggested that there may be differences in the composition of neighborhoods across different talkers. New studies have explored acoustic–phonetic factors in voices that may underlie this variation. The influence of neighborhood has been examined across syllables from 10 talkers. Measurements of syllable duration, speaking rate, and formant frequencies have also been made for these 10 talkers. Across the talkers, there is variation in the effect of neighborhood on phoneme perception. These data will be presented along with acoustic measurements of vowel spaces and other qualities that may influence the similarity of phonemes and words to one another and thus alter the influence of neighborhood upon perception. [Work supported by NIDCD Grant R01DC00219 to SUNY at Buffalo.]

1pSC4. Segment realization in relation to a carrier word's uniqueness point. Roger Billerey-Mosier (Dept. of Linguist., Univ. of California, Los Angeles, 405 Hilgard Ave., Los Angeles, CA 90095)

This paper determines whether the acoustic realization of English phones differs depending on their position relative to a carrier word's uniqueness point, defined as the earliest point at which that word differs from all the words in the lexicon that share the same initial string. As words are confidently identified at their uniqueness point [Marslen–Wilson (1987)], phonetic material past that point may play a less crucial disambiguating role and show signs of reduction akin to those reported in Wright (1997), who found that vowels in monosyllabic words with relatively few perceptual neighbors are acoustically reduced. The acoustic duration of segments located at and past a carrier word's uniqueness point is compared (e.g., -erry in blueberry and strawberry, respectively, because of blueberry's competitors bluebird, blueblood, etc.). No consistent pattern

of differences is found. This result suggests that the domain of Lindblom's (1990) hypospeech and hyperspeech model is higher than the word. The implications of this result for DeJong's (1995) characterization of stress in English as word-level hyperarticulation are discussed.

1pSC5. The importance of lexical status in word recognition of familiar words and novel items. Evanthia Diakoumakou and Jose Benki (Univ. of Michigan, Prog. in Linguist., 1076 Frieze Bldg., Ann Arbor, MI 48109-1285)

Talker variability and its effects on speech perception is an issue of central importance in word recognition research, and it is known that familiarity with a speaker's voice improves recognition of novel words [Nygaard *et al.*, Psych. Sci. **5**, 42–46 (1994)]. In an investigation of how talker variability interacts with lexical status in the recognition of both familiar and unfamiliar words, listeners performed a closed-set identification task. A visual prime of four potential responses consisting of words and nonsense words that differed minimally phonologically was presented, followed by the audio target stimulus for identification using a response box. The experiment consisted of three 100-trial blocks. The trials were blocked according to talker variability (first block, single talker; second block, multiple talkers; and third block, single talker). Lexical status of the target stimulus varied within trials. Response times were also collected. Listeners performed better when the target item was a word in both the single and the multiple talker conditions. No significant interaction between talker variability and lexical status was found. The results suggest that for listeners lexical status is far more important than talker variability during word recognition.

1pSC6. Distinctive functional load of pitch-accent in Japanese based on word familiarity. Mafuyu Kitahara (NTT Commun. Sci. Labs., 3-1 Morinosato Wakamiya, Atsugi, 2430198 Japan and Indiana Univ.) and Shigeaki Amano (NTT Commun. Sci. Labs., Atsugi, Japan)

The present paper proposes to measure the ''functional load of pitch-accent'' for Japanese words. It is calculated from (1) the ratio of the number of accentually contrastive homophone pairs to the number of noncontrastive pairs; and (2) the difference between word-familiarity ratings for each pair. The functional load is larger when there are more contrastive homophones to noncontrastive homophones for the target word, and when the difference of word familiarity scores is smaller between each homophone pair. A large scale word-familiarity database is used for the calculation of functional load and other statistical properties of pitch accent. Distribution of accent and opposition types at each word length are investigated with respect to familiarity scores. Results show that oppositions which include an unaccented form dominate in short simplex words, from low to high-familiarity range. This suggests that the role of pitch accent in distinguishing homophones is biased to the presence–absence contrast, and not to the location per se of pitch accent. Preliminary results from a perception experiment suggest that top-down information in the lexicon, such as functional load and distribution of accent and opposition types, interacts with the bottom-up process in lexical access.

1pSC7. Lexical tone in spoken word recognition: A view from Mandarin Chinese. Chao-Yang Lee (Dept. of Cognit. and Linguistic Sci., Box 1978, Brown Univ., Providence, RI 02912)

In Mandarin Chinese, segmentally identical words are distinguished by lexical tones, which are realized as distinct fundamental frequency patterns over a syllable. Since tone is used phonemically in this language, the question arises of whether tone is implicated in lexical processes in the same way as segmental structure. Three experiments evaluated the role of Mandarin tones in lexical activation and competition. The form and mediated priming experiments showed that Mandarin listeners exploited lexical tone on-line to disambiguate monosyllabic words that were segmentally identical but tonally distinct. Furthermore, acoustic similarity between tones modulated the magnitude of priming and generated opposite priming patterns between the two priming experiments. The interplay of lexical activation and inhibition, based on a neural network model of lexical access, was invoked to account for the dissociation. The gating experiment showed that more acoustic input was needed to recognize words with tonal minimal pairs, indicative of lexical competition. In conclusion, this study showed that lexical tone in Mandarin Chinese participates in lexical processes, and that the mapping of tonal information onto the lexicon is based on acoustic similarity between tones. The author thanks Sheila Blumstein, Philip Lieberman, James Morgan, Peter Eimas, and Molly Homer for their helpful comments.

1pSC8. Development of prevocalic, intervocalic, and postvocalic /r/ in young American children. Richard S. McGowan (CReSS LLC, 1 Seaborn Pl., Lexington, MA 02420), Susan Nittrouer, and Carol Manning (Boys Town Natl. Res. Hospital, Omaha, NE 68131)

The development of children's /r/ production is the subject of a current investigation of a longitudinal database of children's speech. Nine children were recorded interacting with a parent in approximately 2-month intervals from the age of about 15 months to the age of about 31 months. It would be expected that many of these children would not produce /r/ in an adult manner [e.g., M. M. Vihman and M. Greenlee, J. Speech Hear. Res. **30** (1987)], but the properties of children's production of /r/ as a function syllable position and the production of r-colored vowels has not been examined closely. We have examined two children from this database for their ability to produce /r/ and r-colored vowels in various syllable positions. These children more readily produce postvocalic /r/ and r-colored vowels than they produce prevocalic /r/. Further, it may be that practice with intervocalic /r/ may help in future productions of prevocalic /r/'s. These observations will be quantified using formant frequency and amplitude measures and compared with the productions of /r/ by other children in the database.

1pSC9. Coarticulatory information in natural speech stimuli is crucial for infant recognition of syllable sequences. Suzanne Curtin (Dept. of Linguist., USC, 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693), Toben Mintz, and Dani Byrd (USC, Los Angeles, CA 90089-1693)

Recent research has determined that coarticulatory information in speech provides important cues to early word segmentation. This experiment investigates whether 7-month-old infants' ability to recognize a string requires the presence of appropriate coarticulatory information in the speech familiarization stream. Following familiarization to a string of CV syllables, infants were tested to determine if sequences that co-occurred in the familiarization string were preferred over those in which the syllables did not appear adjacently during familiarization. Further, the test phase was conducted so that the items had either appropriate or inappropriate coarticulation information. The results indicate that infants tested on items with appropriate coarticulation listened significantly longer to strings that had appeared during familiarization than to the appropriately coarticulated control strings that never occurred together during familiarization. Interestingly, when presented with inappropriate coarticulation test items, infants showed no preference for previously familiarized strings over the non-co-occurring syllable strings. We conclude that infants are sensitive to coarticulation in recognizing sequences in a speech stream. Furthermore, coarticulatory cues, in combination with other cues to segmentation, greatly enhance recognition of syllable sequences. These results suggest that coarticulation plays an important role in early word segmentation. [Work supported by NIH Grants DC-03172 and SSHRC 752-98-0283.]

1pSC10. Infant sensitivity to lexical neighborhoods during word learning. George J. Hollich III, Peter W. Jusczyk (Dept. of Psych., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218), and Paul A. Luce (Univ. at Buffalo, Buffalo, NY 14269)

Competition from existing lexical items that share similar phonotactic and phonetic properties could inhibit a child's ability to encode new items. Two studies are reported that examine infant's abilities both to detect the similarity among these ''lexical neighbors,'' words that differ by a single phoneme, and to learn a referent for a novel neighbor after an exposure to a high number of these similar sounding words. In study 1, 15-month-old infants exhibited a novelty preference for a neighborhood prototype after being familiarized with twelve lists of twelve neighbors. This suggests that infants are capable of detecting neighborhood similarity among words. In study 2, 17-month olds were tested on their ability to learn the referents of two novel prototypes after being exposed to their respective lexical neighbors. In the high-density condition, six lists of twelve neighbors were used. The low-density condition consisted of six lists of three neighbors plus nine filler items. Results indicated that word learning was significantly better in the low density condition, both in overall looking times and in infant reaction times to the targeted word. These results fit well with current models of spoken language recognition that suggest a competitive effect for words arising from dense lexical neighborhoods.

1pSC11. 12-month-olds show evidence of a possible-word constraint. Elizabeth K. Johnson, Peter W. Jusczyk (Dept. of Psych., Johns Hopkins Univ., Baltimore, MD 21218, zab@jhu.edu), Anne Cutler (Max-Planck-Inst. for Psycholinguist., Nijmegen, The Netherlands), and Dennis Norris (MRC Cognition and Brain Sci. Unit, Cambridge, UK)

The possible-word constraint (PWC) limits the number of lexical candidates considered for a given input by stipulating that the input should be parsed into a string of feasible words [Norris *et al.*, Cogn. Sci. **34**, 191–243 (1997)]. Any segmentation resulting in impossible word candidates (i.e., an isolated single consonant) is disfavored. Four experiments using the head-turn preference procedure investigated whether 12-month-olds observe the PWC to aid them in word recognition. In the first two experiments, infants were familiarized with lists of words (e.g, ''rush''), then tested on lists of nonsense items containing these words in ''possible'' or ''impossible'' positions [e.g., ''niprush'' (nip+rush—possible) or ''prush'' (p+rush—impossible)]. In the other experiments, 12-month-olds were similarly familiarized with lists of words, but test items occurred in sentential contexts; this condition more readily taxed online segmentation abilities. In the first two experiments, 12-month-olds listened significantly longer to targets in ''possible'' versus ''impossible'' contexts when targets occurred at the end of nonsense items, but not when they occurred at the beginning. In experiments 3 and 4, infants listened significantly longer to words in the ''possible'' condition regardless of target location. These results suggest that 12-month-olds, like adults, use the PWC during online word recognition.

1pSC12. Speech perception and production as evidence for the role of phonological representation in the development of phonological awareness. Virginia A. Mann (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Science Plaza, Irvine, CA 92697-5100, vmann@uci.edu) and Judith G. Foy (Loyola Marymount Univ., Los Angeles, CA 90045)

Previous research has shown a clear relationship between phonological awareness and early reading ability. The factors underlying the development of phonological awareness are not well understood; both exposure to literacy and the adequacy of underlying phonological representations are cited as antecedents. To test the hypothesis that phonological awareness associates with strength of phonological representations, this study examined speech perception, production, and phonological awareness in 40 four- to six-year-old children attending private preschools. An extensive questionnaire was administered as a probe to home literacy environment; vocabulary letter knowledge and reading ability were also determined. The data indicate that both literacy environment and the adequacy of phonological representation are factors in phonological awareness. They further suggest that strength of phonological representations may be more closely associated with the awareness of onset-rime than with true phoneme awareness. Children with a less developed sense of rhyme had less mature speech perceptual skills and a less mature pattern of articulation independent of age, vocabulary, and letter knowledge. Children with weaker phoneme awareness also showed less mature speech perception and articulation than children with strong abilities, but this appeared to turn on age, letter knowledge, and vocabulary knowledge.

1pSC13. The role of one's own voice in auditory virtual environments. Christoph Pörschmann (Institut für Kommunikationsakustik, Ruhr-Universität Bochum, 44780 Bochum, Germany)

The sound of our own voices contributes significantly to the way we perceive real and virtual environments and has strong implications on the way we speak. It allows speech to be controlled, which is of particular relevance in acoustically ''difficult'' situations, for example when speaking in a noisy environment. Starting from a model of the different pathways relevant to the perception of one's own voice, the implementation of the model in an auditory virtual environment generator is described. The results of an auditory validation experiment show that the naturalness of one's own voice is significantly increased when the model is applied. Furthermore, this system can serve as a tool for psychoacoustic research regarding the perception of one's own voice. In a second experiment the implications of a plausible presentation of one's own voice on the sense of presence in an auditory virtual environment were studied. For this purpose, the virtual environment generator was enhanced to a multiuser system so that it could be used as an auditory teleconferencing system where two or more participants were perceptually placed in one virtual room. Finally, possible commercial applications for the implemented model are identified and briefly described.

1pSC14. Visual influences on internal structure of phonetic categories. Lawrence Brancazio, Joanne L. Miller, and Matthew A. Paré (Dept. of Psych., 125 NI, Northeastern Univ., Boston, MA 02115, brancazio@neu.edu)

We investigated effects of visual place of articulation on internal category structure for voiceless consonants. Volaitis and Miller [J. Acoust. Soc. Am. **92**, 723–735 (1992)], measuring category goodness for tokens from extended auditory voice-onset-time (VOT) /bi/-/pi/-*/pi/ and /gi/-/ki/-*/ki/ continua, found that the best exemplars of /ki/ fell at longer VOTs than those of /pi/, consistent with the well-known difference between /b/-/p/ and /g/-/k/ voicing boundaries. To explore the basis of this best-exemplar shift, we changed perceived place of articulation of auditory stimuli by pairing them with an incongruent visual token (i.e., the McGurk effect). First, we established that the best-exemplar region for /t/ (like /k/) falls at longer VOTs than that for /p/ for auditory bilabial and alveolar continua. Second, subjects rated auditory /bi/-/pi/-*/pi/ stimuli with visual /pi/ (congruent) and /ti/ (incongruent) tokens as exemplars of /pi/ or /ti/, respectively. The best-exemplar region fell at longer VOTs for /ti/ than /pi/, consistent with previously reported visually induced voicing boundary shifts [Green and Kuhl, Percept. Psychophys. **45**, 34–42 (1989); Brancazio *et al.*, J. Acoust. Soc. Am. **106**, 2270 (1999)]. Thus, perceived place of articulation influences internal category structure even when auditory properties are unchanged, suggesting that the structure is not strictly governed by psychoacoustic factors. [Work supported by NIH/NIDCD.]

1pSC15. Auditory-visual context effects on the perception of voicing. Linda W. Norrix (Dept. of Cognit. Sci., Univ. of Arizona, Tucson, AZ 85721) and Connie Keintz (Univ. of Arizona, Tucson, AZ 85721)

Perception of an auditory continuum can be changed by a preceding visual context [K. Green and L. Norrix, in press, JEP: HPP]. This study examines if a visual context for /l/, as in /abla/, influences the voicing distinction for an auditory /aba/-/apa/ continuum. Voice onset time (VOT) and $F1$ onset frequency were measured during production of /aba/, /apa/ (singleton conditions) and /abla/, /apla/ (cluster conditions). Findings indicate longer VOTs in /apla/ compared to /apa/ and lower $F1$ onset frequencies for the cluster compared to singleton conditions. An auditory-only (AO) continuum (/aba/–/apa/) was created and presented to listeners for identification. In an auditory-visual (AV) condition each auditory token was paired with a face saying /abla/ and presented for identification. If subjects use knowledge about coarticulation and its effects on VOT during production, then they should make more /b/ responses in the AV compared to AO condition. In contrast, if they recover $F1$ information from the visual articulation, then more /p/ responses should be obtained. Findings indicate more /p/ responses in the AV compared to AO condition, suggesting that perceivers are recovering information from visual articulations (perhaps tongue height) rather than using stored knowledge about coarticulatory effects on VOT. [Work supported by NSF.]

1pSC16. Identity priming using McGurk stimuli as primes. Iris Oved, Linda W. Norrix, and Merrill F. Garrett (Cognit. Sci. Prog., Univ. of Arizona, Tucson, AZ 85721)

Research indicates that auditory and visual information is integrated during the perception of speech. Conflicting auditory and visual stimuli can result in an illusory experience known as the McGurk effect (e.g., auditory /bav/ dubbed onto a face saying /gav/ results in a perception of ''dav''). This study used a priming paradigm to investigate whether a phonemic representation for the auditory portion of a McGurk stimulus is active *after* the illusory phoneme is experienced. Subjects were given (nonword) prime-target conditions, including: (1) McGurk (e.g., Prime auditory /bav/ + visual /gav/ = ''dav;'' Target auditory /bav/); (2) Incongruent (e.g., Prime auditory-visual /mav/, Target auditory /bav/); (3) Identity (e.g., Prime auditory-visual /yav/, Target auditory /yav/). Results show that mean reaction times to repeat targets were fastest in the identity condition. Response times for the McGurk and incongruent conditions were indistinguishable from one another and significantly slower than the identity condition. This finding suggests that once the auditory and visual information is combined and a phonemic representation is made, the actual auditory signal is no longer available to affect processing of the target. [This work is based on ideas developed by the late Kerry P. Green and supported by NSF.]

1pSC17. Time-varying information for vowel identification in visual speech perception. Deborah A. Yakel (Orange Coast College, 2701 Fairview Rd., Costa Mesa, California, dyakel@occ.cccd.edu) and Lawrence D. Rosenblum (Univ. of CA, Riverside, Riverside, CA 92521)

The relative salience of time-independent and time-varying dimensions during vowel identification accuracy was examined for visual speech perception. Six different conditions were created using nine vowels /o,I,i,e,ae,a,u,U,uh/ in the consonantal context /bVb/: control syllables, variable center syllables, silent center syllables, initial syllables, and final syllables. Control syllables were full syllables. Variable center syllables were produced by deleting the beginning and end transitions, leaving 50%–60% of the vowel. In silent center syllables, 50%–60% of the vowel was removed, leaving the beginning and end transitions. Initial syllables contained the beginning transition, and final syllables contained the end transition. Results revealed that silent center vowels were identified as accurately as the control syllables. All other conditions showed a significant decrease in visual vowel identification accuracy. A follow-up experiment revealed that the silent center syllable accuracy was greater than the variable center syllables even when their durations were similar. These results are analogous to findings in auditory speech perception [J. J. Jenkins, W. Strange, and T. R. Edman, Percept. Psychophys. **33**, 441–450 (1983)] and suggest that the time-varying aspects of the syllable may be the primary source of information for visual vowel identification.

1pSC18. Is what you hear what you see, even in a foreign language? Seung-Jae Moon (Ajou Univ., Suwon, Korea)

Hearing someone's voice, we immediately recognize the owner of the voice if we know that person. Even when we do not know him, we make up a certain mental image of the person. How accurate is the image triggered by the voice alone? Is the relationship between voice and the image triggered by the voice language-specific or universal? These are the issues this study addresses. Korean speech samples from 8 males and 8 females were recorded. Two pictures were taken for each speaker: one showing the whole body with the reference background so that physical characteristics may be easily compared, and the other showing only the face close-ups. 151 non-Asian subjects without any knowledge of Korean language were asked to match the voices with the corresponding pictures. The results were compared with the previous results obtained from 361 native speakers of Korean. While Korean subjects showed highly accurate matching patterns, non-Asian subjects showed much lower accuracy. The tendency to match a certain voice with a certain picture, regardless of being correct or not, was very high with Korean subjects, but much lower with non-Asian subjects. These results seem to suggest that the perception of voice is language-, or culture-specific.

Session 1pSP

Signal Processing in Acoustics: Signal Processing Techniques

David I. Havelock, Chair
National Research Council, IMS/ASP, M36 Montreal Road, Ottawa, Ontario K1A 0R6, Canada

Chair's Introduction—1:25

Contributed Papers

1:30

1pSP1. The role of the time-reversal processor in acoustical signal processing. James Candy and David Chambers (Univ. of California, Lawrence Livermore Natl. Lab., P.O. Box 808, Livermore, CA 94551)

Time-reversal (T/R) processing of noisy ultrasonic sensor array measurements is an established technique for focusing energy in both homogeneous and inhomogeneous media. It can effectively be used to detect flaws or scatterers by utilizing its primary attribute, the ability to focus on the strongest scatterer. In this paper, temporal techniques are discussed that exploit this focusing property to construct signal processing algorithms. First, the underlying theory is briefly discussed illustrating the focusing property and then the discussion of a technique to detect when the T/R processor has focused on the dominant scatterer or flaw follows. Once focus detection is accomplished, that particular scatterer is removed from the measured data enabling a recursive detection of the next scatterer until all have been detected and removed. Next the localization problem is attacked using the dominant scatterer focusing property of the T/R processor. Here a simple nonparametric model of the wave-front is matched with the measured by varying the assumed flaw (scatterer) location until the best match occurs.

1:45

1pSP2. Multiple eigenvalues of the time reversal operator for a small spherical scatterer. David Chambers (Lawrence Livermore Natl. Lab, P.O. Box 808 L-154, Livermore, CA 94551) and Arthur Gautesen (Iowa State Univ., Ames, IA 50011)

Time reverse mirrors have been used to discriminate and focus energy on individual scatterers embedded in difficult media. A time reverse mirror consists of an array of transceivers, each of which records the incident pressure field then re-emits a time reversed image of it. Prada [J. Acoust. Soc. Am. **97**(1) (1995)] described this operation in terms of a time reverse operator whose eigenvalues are associated with individual scatterers in the medium. In particular, it was shown that there was a one-to-one correspondence between the eigenvalues and scatterers for the case of well separated scatterers whose density is identical to the medium. In this talk, Prada's approached is generalized to scatterers of arbitrary densities where the scattered wave is no longer spherical. It is shown that the one-to-one correspondence between scatterers and eigenvalues of the time reversal operator is broken. For the specific case of a small spherical scatterer, there are three eigenvalues. Physical interpretations of the three eigenvalues and the implications for applications of the time reverse method will be discussed. [Work performed under the auspices of the Department of Energy by the Lawrence Livermore National Laboratory under Contract No. W-7405-Eng-48.]

2:00

1pSP3. A talker tracking strategy for teleconferencing. David I. Havelock (Inst. for Microstructural Sci., Natl. Res. Council, Ottawa, ON K1A 0R6, Canada, david.havelock@nrc.ca)

Sound pickup during teleconferencing is adversely affected by room reverberation. A microphone array steered to the talker can significantly improve the quality of sound pickup but, to follow a conversation in the room, the array must be rapidly steered to alternate talkers. Furthermore, inaccurate and spurious estimates of the source direction must be ''smoothed'' to achieve accurate and stable steering. These two requirements tend to conflict in terms of system response time. One method of achieving both of these performance requirements is to maintain multiple talker trackers simultaneously and to switch between these trackers based on the dynamics of speech and conversation. The array steering performance of a talker tracking system is discussed and results for both simulated and real data are presented.

2:15–2:30 Break

2:30

1pSP4. Analysis filterbank with arbitrary frequency and bandwidth using a multitaper technique. Ryuji Suzuki, John R. Buck (Dept. of ECE & CMaST, UMass Dartmouth, 285 Old Westport Rd., N. Dartmouth, MA 02747), and Antonio H. Costa (UMass Dartmouth, N. Dartmouth, MA 02747)

Acoustical signals are often analyzed for power spectral density. The computational efficiency of the fast Fourier transform (FFT) for the short-time Fourier transform (STFT) is attractive in many contexts, but some applications require nonuniformly spaced analysis frequencies with different bandwidths. Two common examples are the mel frequency scale for speech signals and the constant Q or logarithmic frequency scale analysis of musical signals. These analyses are usually implemented with polyphase filterbanks, wavelets, or an *ad hoc* technique. This research presents a new method that allows arbitrary analysis frequencies and variable bandwidths. The method is based on the multitaper approach from nonparametric spectral estimation. Consequently, this method preserves the benefits of the multitaper approach, namely improved bias and variance properties when compared with the regular STFT-based approach. The multitaper method requires $O(NM)$ computations, where N is the data segment length and M is the number of analysis frequency channels. This is the same order of growth exhibited by the straight discrete Fourier transform with the same N and M. [Work supported by NSF Ocean Sciences.]

2:45

1pSP5. Experimental evaluation of leaky LMS algorithms for active noise reduction in communication headsets. David A. Cartes, Laura R. Ray, and Robert D. Collier (Thayer School of Eng., Dartmouth College, 8000 Cummings Hall, Hanover, NH 03755, david.a.cartes@dartmouth.edu)

An adaptive leaky LMS algorithm has been developed to optimize stability and performance of active noise cancellation systems. The research addresses performance issues related to insufficient excitation, nonstationary noise fields, and signal-to-noise ratio. The algorithm is based on a Lyapunov tuning approach in which three candidate algorithms, each of which is a function of the instantaneous measured reference input, measurement noise variance, and filter length, provide varying degrees of trade-off between stability and performance. Each algorithm is evaluated experimentally for reduction of low-frequency noise in communication

1p MON. PM

headsets and compared with that of traditional LMS algorithms. Acoustic measurements are made in a specially designed acoustic test cell which is based on the original work of Shaw, Brammer and co-workers and which provides a highly controlled and uniform acoustic environment. The stability and performance of the ANR system, including prototype communication headsets, are investigated for a variety of noise sources ranging from stationary white noise to highly nonstationary measured F-16 aircraft noise over a 20-dB dynamic range. Results demonstrate significant improvements in stability of Lyapunov-tuned LMS algorithms over traditional leaky or nonleaky normalized algorithms while providing noise reduction performance equivalent to that of the NLMS algorithm for idealized noise fields.

3:00

1pSP6. Real-time noise canceling based on spectral minimum detection and diffusive gain factors. Hyoung-Gook Kim, Klaus Obermayer (Dept. of Computer Sci., Tech. Univ. of Berlin, Franklinstr. 28/29, 10857 Berlin, Germany), Mathias Bode, and Dietmar Ruwisch (Cortologic AG, Berlin, Germany)

In this paper, we propose a very simple but highly effective psychoacoustically motivated real-time approach on the basis of spectral minimum detection and diffusive gain factors without a speech activity detector. The first processing step is the calculation of the short-time power spectrum of the noisy speech signal. Estimating the background noise, the system calculates diffusive gain values in real time being obtained in a two-layer structure: Each node of a layer is responsible for a single mode of the power spectrum. The first layer, called the ''minimum detection layer,'' holds the present noise level derived from the minimum of the input power spectrum which is detected within frames smaller than the FFT window. The minimum is transformed into a gain factor function using a signal-to-noise ratio control parameter. The diffusive gain factor interaction of neighboring modes is performed in the second layer, called the ''diffusion layer,'' in order to avoid ''musical tones.'' In the frequency domain, a filtering operation is performed by multiplying the noisy speech power spectrum by the diffusive gain factors to yield the filtered signal spectrum. This latter is transformed to the time domain by an inverse Fourier transform with original noisy phase.

3:15

1pSP7. An improved broadband active noise compressor. Hui Lan, Ming Zhang, and Wee Ser (Ctr. for Signal Processing, School of EEE, Nanyang Technolog. Univ., Singapore 539798)

As the extension of active noise equalization (ANE), an active noise compressor (ANCP) has been proposed recently in the literature to nonlinearly compress the noise power to a certain desired range instead of shifting it linearly. The algorithm uses a variable gain factor to control the dynamic range of the residual noise power. However, the controllability provided by that gain factor is inadequate. In this paper, we propose a new way to represent the gain factor. The new method takes into account the desire to accurately maintain the noise power when it is tolerable and suppress it while exceeding the unbearable level. Simulation and implementation results show that the new method outperforms ANCP in controllability and additionally in computational efficiency [J. W. Feng and W. S. Gan, IEEE Signal Process. Lett. **5**, 11–14 (1998)].

MONDAY AFTERNOON, 4 DECEMBER 2000 PACIFIC SALON C, 1:25 TO 4:45 P.M.

Session 1pUW

Underwater Acoustics: Scattering

Dalcio K. Dacol, Chair
Naval Research Laboratory, Washington, DC 20375-5320

Chair's Introduction—1:25

Contributed Papers

1:30

1pUW1. Standard-target calibration of broadband sonars. Kenneth G. Foote (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

The object of a standard-target calibration is specification of the overall frequency response function of the system. In terms of the transmit and receive signal spectra S_T and S_R, standard-target far-field form function F, and two-way acoustic path loss P for an on-axis far-field calibration; the frequency response function is $H=S_R/(S_T F P)$, where each of the displayed functions is frequency-dependent. In practice, a broadband calibration may be hindered by (1) a vanishing denominator value, causing a divergence in H, or (2) a small numerator value or large denominator value, effectively amplifying the influence of noise. The general solution to these problems is to use multiple dissimilar targets to be able to avoid extrema in F, change the transmit signal waveform to avoid nulls in S_T, and ensure that the signal-to-noise ratio is high over the frequency band of interest. The near-field case can be addressed by adjusting the target range. Given these measures, the standard-target technique demonstrated by Dragonette *et al.* [J. Acoust. Soc. Am. **69**, 1186–1189 (1981)] for scattering measurements may be rendered into general practice for active broadband sonars.

1:45

1pUW2. Bistatic target detection, interferometry, and imaging with an autonomous underwater vehicle platform. Joseph R. Edwards, Henrik Schmidt (Dept. of Ocean Eng., Rm. 5-204, MIT, 77 Massachusetts Ave., Cambridge, MA 02139, jre@mit.edu), and Kevin D. LePage (SACLANT Undersea Res. Ctr., 19138 San Bartolomeo (SP), Italy)

The acoustic detection and classification of completely and partially buried objects in the multipath environment of the coastal ocean presents a major challenge to the underwater acoustics community. However, the rapidly emerging autonomous underwater vehicle (AUV) technology provides the opportunity of exploring entirely new sonar concepts based on mono-, bi-, or multistatic configurations. For example, the medium frequency regime (1–10 kHz) with its bottom penetration advantage may be explored using large synthetic apertures, where acoustic information is accumulated over a series of sonar pings. The performance of such approaches is highly dependent on accurate platform navigation and timing, which poses a significant challenge to AUV developers, particularly because the navigation procedures are themselves dependent on the complicated multipath acoustic environment. Using experimental data from the GOATS'98 SACLANTCEN/MIT experiment, this paper describes an investigation into the feasibility of combining seabed scattering data from consecutive pings of a fixed parametric source to form a bistatic synthetic aperture for target localization and imaging with an AUV-based receiving platform. The paper describes different levels of bistatic processing including both incoherent and coherent beamforming and very large aperture interferometric approaches, and the associated performance trade-offs are discussed. [Work supported by ONR and SACLANT.]

2:00

1pUW3. Experimental validation of numerical models of 3-D target scattering and reverberation in very shallow water. Irena Veljkovic and Henrik Schmidt (MIT, Cambridge, MA 02139)

OASES 3-D, a wave theory model of seabed insonification, three-dimensional target scattering, and rough seabed reverberation, has been developed to investigate characteristics of multistatic scattering and reverberation from rippled shallow-water seabeds with buried targets. The model combines a scattering theory based on the method of small perturbations with a seismoacoustic propagation model, allowing for field simulations in arbitrarily stratified ocean environments. The validation of this new modeling framework was a major objective of the GOATS'98 experiment carried out in 12–15-m deep water at Elba in May 1998. Spherical and cylindrical targets were buried at different depths within a 10×10 m area of sandy bottom and insonified by a parametric source. The unique dataset provided by the experiment has been used for both experimental validation of numerical models and identification of features of the 3-D acoustics which distinguish targets from reverberation. The methodology used for estimating the bistatic scattering strength of the targets and the background seabed reverberation from data collected by a moving AUV platform will be described. The performance of the model for predicting scattering from proud and buried targets will be discussed and illustrated by examples including both isotropic and aspect-dependent targets. [Work supported by ONR and SACLANTCEN.]

2:15

1pUW4. The Fermi pseudo-potential and acoustical scattering. Dalcio K. Dacol and Dilip G. Roy (Naval Res. Lab., Washington, DC 20375-5320)

The Fermi pseudo-potential was introduced in quantum mechanics as a means of simplifying problems involving scattering by a multicentered potential including applications to many-body problems. As such, it has direct applications in acoustics. It also provides a physically intuitive basis for discussing scattering by complex objects. In this presentation the pseudo-potential concept is briefly reviewed. Then, applications to scattering by complex objects and to acoustic scattering by multiple objects are discussed. The case of multiple object scattering in an oceanic waveguide will also be discussed in detail. [Work supported by ONR.]

2:30

1pUW5. Extinction theorem for object scattering in a stratified medium. Purnima Ratilal and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

A simple relation for the rate at which energy is extinguished from the incident wave of a far-field point source by an obstacle of arbitrary size and shape in a stratified medium is derived from wave theory. This relation generalizes the classical extinction theorem, or optical theorem, that was originally derived for plane wave scattering in free space and greatly facilitates extinction calculations by eliminating the need to integrate energy flux about the obstacle. The total extinction is shown to be a linear sum of the extinction of each waveguide mode. Each modal extinction involves a sum over all incident modes that are scattered into the extinguished mode and is expressed in terms of the object's plane wave scatter function in the forward azimuth and equivalent plane wave amplitudes of the modes. The only assumptions are that multiple scattering between the object and waveguide boundaries is negligible, and the object lies within a constant sound-speed layer. Calculations for a shallow-water waveguide show that the extinction cross section is highly dependent on measurement geometry, and medium stratification, as well as the scattering properties of the object and may be significantly modified by the presence of absorption in the medium.

2:45

1pUW6. A spectral formulation for the Doppler-shifted field scattered by an object moving in a stratified medium. Yisan Lai and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

A spectral formulation for the 3-D field scattered by an object moving in a stratified medium is derived using full-field wave theory. The derivation is based on Green's theorem for the time-domain scalar wave equation and accounts for Doppler effects induced by target motion as well as source and receiver motion. The formulation is valid when multiple scattering between the object and waveguide boundaries can be neglected and the scattered field can be expressed as a linear function of the object's plane wave scattering function. A normal mode formulation that is more computationally efficient but less general is also derived from first principles. The advantage of the spectral representation is that it incorporates the entire wave number spectrum, including evanescent waves, and so can potentially be used at much closer ranges to the target than the modal formulation. The Doppler effects are illustrated through a number of canonical examples.

3:00–3:15 Break

3:15

1pUW7. On scattering effects due to the proximity and relative position of two bubbles in a sound field. George Kapodistrias and Peter H. Dahl (Appl. Phys. Lab. and Dept. of Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, georgek@apl.washington.edu)

Recently the authors published results from a theoretical and experimental investigation on scattering of sound from two bubbles symmetrically arranged about the combined beam axis of a set of transducers [J. Acoust. Soc. Am. **107**, 3006–3017 (2000)]. In this presentation the investigation is extended to two additional geometries, with the two bubble array placed at angles of 0° and 45° from the combined beam axis. For each angle, the half interbubble distance d is varied such that the dimensionless variable kd ranges from 0.2–5 (for the 0° case) and 0.2–10 (for the 45° case), where k is the acoustic wave number. Modeling is accomplished by using a closed-form solution derived from the multiple scattering series, with the bubble scattering function expressed in terms of spherical harmonics. Experimental data are obtained by symmetrically arranging two bubbles, each of radius $a \approx 425$ μm, on a fine nylon thread, with the bubbles insonified by tone bursts with a center frequency of 120 kHz. The data closely agree with the simulations, and it is verified that, regardless of the geometry, for $kd \leqslant 1$ the response of the two bubble array drastically departs from the one due to single scattering. This departure is attributed to multiple scattering and is manifested as a reduction in backscattered radiation.

3:30

1pUW8. Rough sea surface limitations on high-frequency SAS imaging. Enson Chang, Ralph E. Chatham, David S. Marx, Matthew A. Nelson, Angela Putney, and L. Kieffer Warman (Dynam. Technology, Inc., 21311 Hawthorne Blvd., Ste. 300, Torrance, CA 90503)

It is a long-held notion that medium-induced signal fluctuations render long-range synthetic aperture sonar (SAS) imaging unfeasible. Recent experimental results in shallow water, however, indicate that near-theoretical SAS performance can be achieved with the aid of (sonar data driven) motion compensation and autofocus algorithms, even for surface- and bottom-reflected paths. This study examined in detail the role of these adaptive compensation algorithms in overcoming rough surface scattering-induced phase errors. Imaging and compensation algorithms were applied to sonar data that was numerically modeled for moderate to severe surface conditions. We report here the predicted imaging performance as a function of surface conditions and grazing angles. [Work supported by the Office of Naval Research.]

3:45

1pUW9. Modal conversion by rough surface scattering: The key to the *T*-phase. Robert I. Odom, Minkyu Park, and Darin J. Soukup (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

The amplitudes of the propagating acoustic modes associated with *T*-waves decay exponentially below their ray equivalent turning points, and cannot be excited directly by an earthquake. The modal decomposition for a *T*-wave producing earthquake that occurred near the western tip of the Blanco TFZ has been computed. The directly excited higher order modes are characterized by relatively large amplitudes in the ocean crust, significant water-borne components, and often strong interface components at the ocean-bottom boundary. Employing the modal scattering theory of Park and Odom (1999), it is found that energy has been transferred from higher order modes to the Stoneley fundamental and the lower order modes have significant amplitude at the water-bottom interface. Scattering from irregular ocean bottom bathymetry is the mechanism for the energy transfer. The lowest order acoustic modes, modes 1 and 2, are only very weakly excited because they have very small amplitudes at the bottom. This is consistent with the interpretation of de Groot-Hedlin and Orcutt (1999). [Work supported by the NOPP Program and ONR.]

4:00

1pUW10. Integration of numerical scattering functions with the ocean wave-guide. Chee K. Lim and J. T. Goh (DSO Natl. Labs., 20 Science Park Dr., Singapore Science Park, Singapore 118230, lcheekho@dso.org.sg)

Based on the generalized modal formulation of acoustic scattering by F. Ingenito [J. Acoust. Soc. Am. **82**, 2051–2059 (1987)] in a stratified ocean, we present a method for integrating a numerical scattering function obtained from a commercial boundary element code, SYSNOISE, with an acoustic propagation code to study scattering from arbitrary objects. We benchmarked the hybrid method by comparing the scattered fields for a rigid spherical shell obtained using this approach with an analytical solution. We discuss issues related to the numerical implementation of the approach, such as the extraction of the scattering function for an unbounded ocean medium, correct extraction of multiple modal scattering kernal using a single scattering function (for axis-symmetrical object) and numerical phase compensation procedures associated with numerical source location. The methodology developed here would allow the study of acoustic scattering problems in the ocean for arbitrary-shaped elastic objects, which are not amenable to analytical means but solvable only by numerical methods.

4:15

1pUW11. Comparison of the shallow water acoustic scattering from a submarine-shaped object with that from canonical shapes. Chee K. Lim (DSO Natl. Labs., 20 Science Park Dr., Singapore Science Park, Singapore 118230, lcheekho@dso.org.sg)

The scattering from an arbitrary-shaped object like a submarine is a complex process. In practice, for detection studies, canonical shapes such as a cylinder have been widely used to emulate a submarine-shaped object. However, the validity of this approximation is not easily quantified. In a previous paper [Chee K. Lim, ''Integration of numerical scattering functions with the ocean wave-guide,'' 139th ASA, Nov. 2000], the approach to integrating scattering functions computed using commercial codes with a propagation code was presented. The approach allows convenient calculation of scattering from very complex elastic objects. This presentation will compare the acoustic scattered field of eight different targets—a rigid sphere, rigid cylinder, rigid cone, rigid submarine-shaped object, elastic cylinder, stiffened elastic cylinder, stiffened elastic cone and, lastly, a stiffened elastic submarine-shaped object. Five aspects source angles (0, 45, 90, 135 and 180 deg) and their corresponding forward- and backward-scattered field will be compared using the results of the submarine-shaped object as the reference solution. The results would provide useful insights for future underwater acoustic scattering modeling and model-based signal processing.

4:30

1pUW12. Scattering of acoustic waves by a prolate body in plane-stratified waveguide. Boris G. Katsnelson, Valery A. Grigoryev (Voronezh Univ., 1 Universitetskaya Sq., Voronezh 394593, Russia), Venedict M. Kuzkin, and Valery G. Petnikov (General Phys. Inst., Moscow 117942, Russia)

Technique for calculation of diffraction field by spatially localized inhomogeneity is proposed. This technique is based on the fact that the scattering matrix of the waveguide mode is expressed through the scattering amplitude in free space. This permits separation of the diffraction problem in free space and the propagation problem in the waveguide (the problem of calculation of waveguide modes). Requirements for sound-speed variation and for distance between scatterer and waveguide boundaries are formulated. On the basis of the energetic approach, the concept of the cross section in the waveguide is proposed. The coupling modes and sound absorption are discussed as well. The space-time structure of a sound field scattered by the moving localized inhomogeneity in the direction of incident wave (the forward scattering) is given and discussed. As an example, structure of the sound field for a scatterer modeled by spheroid prolate for different parameters of waveguides is calculated. [Work supported by RFBR, Grant 99-02-17671.]

Additional registration fee required to attend this Tutorial.

Session 1eID

Interdisciplinary: Tutorial Lecture on Virtual Musical Instruments

Joseph Pope, Chair
Pope Engineering Company, P.O. Box 590236, Newton Centre, Massachusetts 02459-0002

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. Virtual musical instruments. Julius O. Smith III (Ctr. for Computer Res. in Music and Acoust., Dept. of Music, Stanford Univ., Stanford, CA 94305)

Virtual musical instruments are interesting from multiple points of view. The listener may, or may not, know that the music is computer generated. The composer can use the computer to produce sonorities that cannot be obtained in a real environment, such as strings many meters long, or instruments that gradually change form. The performer can benefit from virtual instruments because they are usually easier to play than real instruments. The scientists, being able to reproduce the sonority of a particular instrument, show that the physics of the instrument itself has been understood. For these reasons and more, virtual musical instruments based on mathematical acoustics have been successful for many years in the computer music research community. In this tutorial we will cover some of the more effective virtual instrument algorithms, and how they can be played and varied in real time. Examples will be described and sound examples played. Starting from the simplest cases of plucked and struck strings, we will show how simple configurations of delay lines, digital filters, and nonlinear elements can be used to synthesize realistic sounding musical instruments. Examples will work up to bowed strings and other ''self-sustained oscillators'' such as in the flute and clarinet. The richness and breadth of the timbral spaces offered by these models will be illustrated.