Session 4aAAa

Architectural Acoustics: Recent Authors of Books on Architectural Acoustics

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

Ronald R. Freheit, Cochair
Wenger Corporation, 555 Park Drive, Owatonna, Minnesota 55060

Chair's Introduction—9:00

In connection with the Acoustical Society's first ''Book Fair'' ongoing at this Spring 2000 Atlanta Meeting, the Technical Committee on Architectural Acoustics invited authors and editors of recent publications of interest to share their thoughts, motives and objectives in producing their works and the extent to which they feel these objectives had been achieved. This session will provide an excellent opportunity to explore the ''state-of-the-art'' in architectural acoustics at the conclusion of this most productive 20th century and to think about where we might be headed in the century ahead. We hope for a lively discussion to prepare ourselves for Red Wetherill's lecture in the following session (4aAAb) on the remarkable history of the field. Authors and/or editors representing the books listed below have been invited to participate. Authors who are unable to attend the Atlanta meetings have been invited to submit short remarks which will be read at the panel discussion.

Apfel, R. E., *Deaf Architects and Blind Acousticians....A Guide to the Principles of Sound Design*, Apple Enterprises Press, New Haven, Connecticut (1998)

Barron, M., *Auditorium Acoustics and Architectural Design*, E&FN Spoon, an imprint of Chapman and Hall, London and New York (1993)

Beranek, L. L., *Concert and Opera Halls....How They Sound*, Acoustical Society of America, Melville, New York (1996).

Cavanaugh, W. J. and Wilkes, J. A., Ed., *Architectural Acoustics....Principles and Practice*, Wiley, New York (1998)

Cowan, J. T., *Architectural Acoustics* (CD-ROM), McGraw-Hill, New York (1999)

Crocker, M. J., *Handbook of Acoustics*, Wiley, New York (1998)

Egan, M. D., *Architectural Acoustics*, McGraw-Hill, New York (1988).

Harris, C. M., *Noise Control in Buildings...A Practical Guide for Architects and Engineers*, Ins. Noise Control Engr., Poughkeepsie, New York, 1997 (orig. 1994, McGraw-Hill) and *Handbook of Acoustical Measurements and Noise Control*, Acoustical Society of America, Melville, New York (1998)

Irvine, L. K. and Richards, R. L., *Acoustics and Noise Control Handbook for Architects and Builders*, Krieger, Melbourne, Florida (1998)

Izenour, G. C., *Theater Design/Theater Technology/Roofed Theaters of Classical Antiquity*, Yale University Press, New Haven, Connecticut (1996)

Kopec, J. W., *The Sabines at Riverbank*, Acoustical Society of America (by Peninsular Press), Melville, New York (1997)

Knudsen, V. O. and Harris, C. M., *Acoustical Designing in Architecture*, Acoustical Society of America, Melville, New York (1980) (orig. 1950, Wiley)

Lubman, D. and Wetherill, E. A., Eds., *Acoustics of Worship Spaces*, Acoustical Society of America, Melville, New York (1985)

McCue, E. R. and Talaske, R. H., Eds., *Acoustical Design of Music Education Facilities*, Acoustical Society of America, Melville, New York (1990)

Mehta, M., Johnson, J. and Rocafort, J., *Architectural Acoustics...Principles and Design*, Prentice Hall, Upper Saddle River, New Jersey (1999)

Ollswang, J. and Ambrose, J., *Simplified Design for Building Sound Control*, Wiley, New York (1995)

Pelton, H., *Noise Control Management*, Van Nostrand Reinhold (1993)

Sabine, W. C., *Collected Papers on Acoustics*, Acoustical Society of America (by Peninsular Press), Melville, New York (1994) (orig. 1921, Harvard Univ. Press; 1964, Dover)

Salter, C. M., Ed., *Acoustics, Architecture, Engineering, the Environment*, William Stout Publishers, San Francisco, California (1998)

Sendra, J. J., Ed., *Computational Acoustics in Architecture*, WIT Press, Southampton, UK (1999)

Talaske, R. H. and Boner, R. E., Eds., *Theatres for Drama Performance: Recent Experience in Acoustical Design*, Acoustical Society of America, Melville, New York (1987)

Talaske, R. H., Wetherill, E. A., and Cavanaugh, W. J., *Halls for Music Performance: Two Decades of Experience*—1962–1982, Acoustical Society of America, Melville, New York (1982)

Uzzle, T., Bushnell, R. A., and Bouliane, T. G., *Technical Fundamentals of Audio*, Intertec Publishing, Overland, KS, 1999

FRIDAY MORNING, 2 JUNE 2000 TOWER 1206, 11:00 A.M. TO 12:05 P.M.

Session 4aAAb

Architectural Acoustics and Committee on Archives and History: History of Architectural Acoustics

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

Chair's Introduction—11:00

Invited Paper

11:05

4aAAb1. The flowering of architectural acoustics in the twentieth century. Ewart A. Wetherill (Paoletti Assoc., 649 Mission St., San Francisco, CA 94105)

Arising from a single building deficiency, the art and science of architectural acoustics came into being as a practical discipline at the start of the twentieth century through the painstaking exploration and profound intuition of Wallace Clement Sabine and other pioneers in acoustics. It now faces the dawning of the twenty-first century as a universal scientific discipline and an integral component of advanced building design. For the past 70 years of its often-fragile coexistence with the building professions, of architecture, much of the growth of understanding and education in architectural acoustics has found a focus in the Acoustical Society of America in collaboration with its companion societies throughout the world. While the understanding of building design for good hearing is still uneven and far from complete, the story of this development, as seen in the history of the Acoustical Society, indicates rich promise for the years ahead.

A NOTE ABOUT THE ASA HISTORY LECTURE SERIES

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

With the cooperation of the technical committees, distinguished individuals are selected to review the history of their particular technical specialty and present a lecture which shows how that activity has developed and has contributed to the Society at large and to the broad field of acoustics as well. At the 138th meeting in Columbus, Ohio, the first two History Lectures were presented: Gabriel Weinrich on Musical Acoustics and Robert Beyer and David Blackstock on Physical Acoustics. At each subsequent meeting two additional lectures will be scheduled including those in Architectural Acoustics and Engineering Acoustics at this meeting in Atlanta.

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

Session 4aEA

Engineering Acoustics: Acoustic Barriers and Absorbers

Kirk E. Jenne, Cochair
Naval Undersea Warfare Center, Newport, Rhode Island 02841

Joseph Vignola, Cochair
Naval Research Laboratory, Washington, DC

Chair's Introduction—8:00

Contributed Papers

8:05

4aEA1. Acoustic barrier: BEM prediction and experimental verification. Samir Gerges and Calza Arlinton (Mech. Eng. Dept., Federal Univ. of Santa Catarina, Cx.P. 476, Florianopolis, SC, Brazil)

This paper presents the noise attenuation of a barrier by BEM prediction, analytical formulas, and also experimentally. The analytical formulas used consider the diffraction of semi-infinte barrier, based on source image and phase change caused by the reflection of the sound waves from rigid ground surface. Numerical simulation by BEM and analytical calculation of the barrier model are compared and later validated by experimental tests with a barrier built in a semianechoic chamber. The results obtained by BEM prediction, analytical formulas, and experimental measurements are compared and the divergences between them are discussed.

8:20

4aEA2. Acoustic characterization for quality or damage evaluation in concrete structures. Hasson M. Tavossi (Dept. of Eng. Sci. and Mech., The Penn State Univ., 227 Hammond Bldg., University Park, PA 16802-1401)

Concrete structures such as building columns, parking platforms, concrete highways, and bridges are under continuous action of two major types of cyclic stresses. First, thermal stress cycles due to seasonal temperature variations, resulting in cyclic thermal expansion and contraction. Second, mechanical stress cycles due to cyclic load variations; for example, resulting from the passage of a vehicle on a concrete platform. The cyclic stress, after a certain number of cycles, causes damage in concrete by fatigue. In this study, fatigue damage in concrete is evaluated by acoustic techniques, using wave speed and attenuation. Acoustic sensors of 50-kHz center frequency are coupled to the concrete, as transmitter and receiver, to evaluate concrete quality after concrete samples are subjected to thermal stress cycles. Results obtained in the laboratory will be presented, showing a significant weakening of the concrete strength after only a few stress cycles. The cyclic stress causes a significant drop in acoustic wave speed of the concrete, represented by a low value of its Youngs modulus. A quantitative relationship between fatigue damage and acoustic characteristics of concrete can be established, to evaluate the quality of the concrete structures.

8:35

4aEA3. Effective wide-band, low-frequency multiple resonator sound absorber. Fathy B. Shenoda and Mohammed Abd-Elbasseer (Dept. of Acoust., Natl. Inst. for Standard ''NIS,'' P.O. Box 136, Giza 12211, Egypt)

Resonator-type sound absorbers are incorporated to provide relatively large sound attenuation in narrow bands. They are applied in several environments and have great practical value for the attenuation of low frequencies. A detailed experimental study leads to the construction of a multiple resonator sound absorber. It is made of metal sheet and has a total volume of 0.0016 m^3. The acoustic performance of the designed device, namely, the sound absorption coefficient and the normalized input acoustic impedance, were measured in 20-Hz steps at normal sound incidence in room temperature and under heating conditions up at 4000 °C. The designed device provides good sound-absorption-frequency characteristics not only in the high values of the sound-absorption coefficient, but also in covering a wide frequency band (60 to 1000 Hz) without attenuation breaks. Also, its acoustic characteristics are changed very little under heating conditions. Applying earlier results obtained by Lord Rayleigh and simplified by Crandall for the propagation of a sound wave in a tube, and recently applied by Maa to characterize microperforated sound absorbers, the acoustic performance of the designed multiple resonator was predicted. The measured and predicted results are in good agreement.

8:50

4aEA4. Sound transmission through perforated panel with internal Helmhotz resonators. Pedro E. Solana Quiros, Miguel A. Picard Lopez, and Juan V. Arizo Serrulla (Universidad Politcnica de Valencia; E.T.S.I. Industriales, Edfs D4-D5; Camino de Vera 14; 46022 Valencia, Spain)

The study of the passive control technique of noise through a perforated panel with resonators is useful in acoustic barrier design for both industrial and traffic noise. The loss mechanisms that reduce the energy of sound have been considered in the method for predicting the TL of compound flat panels containing Helmholtz resonators. A transmission coefficient is obtained for the system that relates the resonance frequency of the resonators for nonlinear resistance at the nozzle. The outer and the interior panels are assumed to act as simple mass reactances. In order to examine the validity of the analytical results, comparisons were made with experimental data. To provide acoustic excitation, a loudspeaker array was used. The excitation and transmitted sound fields were measured with a microphone array located on each side of the panel. The model predicts that a significant increase in TL can be expected at or near the mean resonance frequency of the system multilayer-panel–resonator array. In addition, a significant decrease in the TL is predicted at higher frequency. Also, it has been observed that as the ratio of the nozzle area divided by the resonator surface area increases, the frequency at which the peak TL occurs also increases.

9:05

4aEA5. Wave propagation experiments using point load excitation in a multilayered structure. Chunnan Zhou (General Electric Co., 534 Griffin Rd., Bangor, ME 04401), S. John Popovics (Drexel Univ., Philadelphia, PA 19104), and D. Jan Achenbach (Northwestern Univ., Evanston, IL 60201)

Transient wave propagation measurements on a single plate, a two-layer plate, and a two-layer-on-a-half-space structure are reported. The experiments are performed using point load excitation and point detection at the surface of the structure. The experimental results are used to verify

the accuracy of a new model for wave propagation in layered structures. The basis of the model is first introduced. Experimental measurements using different wave sources, plate materials, and wave sensors are compared with each other and with theoretical results. The actual source functions for pencil-lead-break wave sources are obtained. Next, experiments performed on an aluminum plate bonded to stainless steel plate are reported. Both cases of stainless steel atop aluminum and aluminum atop stainless steel are considered. The experimental results show good agreement with the theoretical predictions. The effects of the second layer on the transient displacement of the top surface are discussed. Finally, a two-layer-on-a-half-space structure consisting of a stainless steel plate as the top layer, an aluminum plate as the second layer, and a thick acrylic resin block as the half space is studied. The presented experimental transient measurements agree with the predicted results, thus verifying the accuracy of the model.

9:20

4aEA6. Study of acoustic emission in a plate using laser ultrasonics and the finite element method. Zhiqiang Shi, Jacek Jarzynski (Dept. of Mech. Eng., Georgia Tech, Atlanta, GA 30332-0405), and Laurence Jacobs (Georgia Tech, Atlanta, GA 30332-0355)

Acoustic emission (AE) is the spontaneous release of transient energy due to changes in localized stresses or incremental crack growth. This research studies AE generation, propagation, and detection in platelike components. The modeling of a variety of AE sources (such as microcrack extension) and their resulting wave propagation signature, in a two-dimensional component, are investigated. The experimental technique features a laser-ultrasonic system for the point source generation and broadband, point-wise detection of elastic waves. The numerical algorithm uses the finite element method (FEM), which has the flexibility to simulate a variety of AE source types, and thus quantify source and geometric effects—geometric effects are specific component effects such as boundaries—on the predicted AE waveforms. These results provide a quantitative understanding of AE mechanisms, and assist in the interpretation of field-measured AE signals.

9:35

4aEA7. Noise and vibration control for land mine detection. R. Daniel Costley (Miltec, Inc., Natl. Ctr. for Physical Acoust., University, MS 38677), James M. Sabatier, and Ning Xiang (Univ. of Mississippi, University, MS 38677)

Land mines can be detected by scanning the ground with a laser doppler vibrometer (LDV) to measure the motion (velocity) of the ground as it is being sonified. However, the system at times suffers from a low signal-to-noise ratio, especially when trying to detect deeply buried mines. If the signal-to-noise ratio could be improved, these mines could be more easily detected. One contribution to the noise is due to the acoustically induced vibration of the LDV. This noise is difficult to filter digitally or electronically since it occurs at the same frequency as the ground motion being measured. Passive noise and vibration control remedies have been incorporated with some success. These include enclosing the LDV within a box and mounting it on damped springs. However, it is impossible to make the LDV completely stationary at all frequencies. The translation of the LDV beam over anything other than a completely smooth surface will appear to the LDV as an out-of-plane motion of the surface. The noise and vibration control strategies will be presented and discussed, along with measurements showing both their effectiveness and limitations.

9:50–10:05 Break

10:05

4aEA8. Mode analyses in a fluid-filled borehole surrounded by a concentrically layered formation. Xiuming Wang and Kevin Dodds (CSIRO Petroleum, P.O. Box 1130, Technology Park, Bentley, WA 6102, Australia)

Acoustic wave modes including normal modes and leaky modes in a fluid-filled borehole surrounded by a concentrically layered medium are analyzed. Dispersions and excitation spectra of various modes related to monopole, dipole, including hexapole, octepole, and decapole sources, are calculated for typical well logging environments. The numerical results demonstrate that the cutoff frequencies of nonsymmetrical modes vary greatly with the orders. A Stoneley mode has no cutoff frequency in a fast formation, while at a very slow formation, it possesses a cutoff frequency. Multiple pseudo-Rayleigh modes, flexural modes, and the other higher modes have their correspondent cutoff frequencies. As frequency decreases from the cutoff frequencies, these modes correspond to the complex pole on the other Riemann sheets and become leaky modes. Also, followed by our previous work (Wang *et al.*, 1994, 1995), it is shown that head compressional and shear arrivals cannot be totally described by contributions of vertical branch cuts in a wave field representation when the Poisson's ratio is larger than 0.36 for dipole and quadrupole sources. The numerical results also show that the radial penetration of the normal modes is very shallow. The Stoneley modes are limited to 10 cm for a typical logging environment for a lower frequency than 3 kHz.

10:20

4aEA9. Recent developments in the theory of ground vibration boom from high-speed trains. Victor V. Krylov (Dept. of Civil and Structural Eng., The Nottingham Trent Univ., Burton St., Nottingham NG1 4BU, UK, Victor.Krylov@ntu.ac.uk)

Generation of ground vibration boom by high-speed trains was theoretically predicted in 1994 by the present author. The first experimental observation of this phenomenon was reported in 1997 by C. Madshus, who worked with his team on the assessment of the newly opened high-speed railway line from Gothenburg to Malmo in Sweden. The ground on the site of observation was very soft, with Rayleigh wave velocity of only 45 m/s. Therefore, for train speeds as low as 160 km/h, this Rayleigh wave velocity could be exceeded and the ground vibration boom observed. It is now well understood that ground vibration boom represents a serious hazard for the built environment, especially in the cases where high-speed lines are built on very soft soil. The present paper reviews the current status of the theory of ground vibration boom from high-speed trains. Among the problems to be discussed are contributions of different generation mechanisms, effect of track wave resonances on generated ground vibrations, effects of layered geological structure of the ground, waveguide effects of the embankments, and focusing of generated waves due to the track curvature. The results of theoretical calculations are compared with the existing experiments. [Work supported by EPSRC.]

10:35

4aEA10. New geometric sound absorbers. Glenn E. Warnaka (Future Technologies, L.L.C., 1612 S. Allen St., State College, PA 16801)

In a previous paper of the same name, two new geometric sound absorbers were discussed. The geometric absorbers use closely coupled quarter-wave resonators to absorb sound. Because the structures absorb sound by their configuration, they do not require the use of conventional porous or fibrous sound-absorbing materials and can be made of nearly

any substance such as plastic, wood, metal, etc. The absorbers can be made very rugged and long-lasting for industrial, military, and outdoor use. The previous paper discussed two basic sound absorbers, configuration delta and flat absorber, but the paper concentrated on the former sound-absorber configuration. This paper will review configuration delta but will concentrate on describing the structure and performance of the flat absorber. This absorber can be made in a thin treament, less than 25 mm thick, that provides high acoustic absorption. A number of different designs will be shown, and the results of standing wave tube and reverberation room tests will be given. Combining the two treatments described in this paper provides a broad frequency range absorber that is both compact and rugged. Helmholtz resonators can also be used with the flat absorber concept, and this is discussed.

10:50

4aEA11. Noise attenuation using a parabolic muffler. Jonathan Zalben (Yale Univ., New Haven, CT 06520-5893, jonathan.zalben@yale.edu)

This project is a patent-pending muffler design with paraboloid baffle chambers to reduce noise pollution from internal combustion engines (especially those of lawn mowers and leaf blowers). After theorizing the most reasonable design, experiments were conducted on wood and steel (18 gauge) models. The experiments show that for a wood model, white noise produced by a radio is reduced from 80 to 64 dB (97.5% intensity reduction). The wood model amplifies sound for pure frequencies around 1000 Hz and attenuates at higher frequencies. The steel model attenuates engine noise by 75% with varying rates of attenuation for pure frequencies. [Work supported by Residents for a More Beautiful Port Washington, the Nassau–Suffolk Landscapers Association, and Port Washington School System.]

11:05

4aEA12. Improvement of one- and two-layer liner design with bias flow. Jesse Ian Follet (Virginia Consortium of Eng. and Sci. Universities, 303 Butler Farm Rd., Ste. 101, Hampton, VA 23666, jfollet@vces.larc.nasa.gov)

The coupling of the resistance and reactance in acoustic liners has in the past limited aircraft engine liner design performance. This is due to the parametric dependence of the impedance on the perforated sheet geometry and cavity depths both having a significant effect on both parts of the impedance. For normal incidence liner impedance, designing a liner with its reactance approaching zero and resistance approaching unity is essential to the goal of producing maximum broadband absorption. Introducing a mean flow through the liner has shown that the liner resistance can be changed with minimal effect on the reactance. Liner geometry can be designed to achieve a broadband reactance that is as close to zero as possible. The resistance can then be increased to near unity by allowing a mean bias flow through the liner to produce the maximum broadband absorption. With sufficient modeling of liner impedance, an optimization routine can be used to find the liner characteristics that produce maximum absorption over a broad range of frequencies and sound-pressure levels. It will be shown that using bias flow can improve normal incidence liner absorption by 17% for single layer liners and 7% for double layer liners.

11:20

4aEA13. Variational formulations for sound in enclosed spaces and ducts. Mario Zampolli, Allan D. Pierce, and Robin O. Cleveland (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215)

The present formulation differs from those appearing in earlier literature in that the boundary conditions are explicitly incorporated, so that not all admissible trial functions need to satisfy them. If the class of considered trial functions is restricted so that each is a finite sum of simple functions with arbitrary coefficients, then the answer selected by the variational principle attempts to satisfy the boundary conditions plus the partial differential equations imposed in the interior simultaneously, but only achieves a trade-off. An example is a waveguide of variable cross section, with one end open and the other closed. The Rayleigh integral solution for sound radiated by a baffled aperture with an arbitrary distribution of velocity amplitude on the aperture yields the open-end boundary conditions. Individual trial functions are combinations of cross-sectional mode-shape functions with axially dependent coefficients. Numerical instabilities encountered in previous direct integration of the coupled-mode equations are avoided by taking the axial coefficient functions to be piecewise linearly along the axis. The amplitudes at the nodal points satisfy a sparse system of linear algebraic equations, this system resulting directly from the variational principle. [Work supported by DARPA/USNWRC.]

FRIDAY MORNING, 2 JUNE 2000 PEACH ROOM, 10:00 A.M. TO 12:00 NOON

Session 4aED

Education in Acoustics: Take Fives—Sharing Ideas for Teaching Acoustics

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

Do you have a novel demonstration, a new laboratory experiment, a favorite video, a recorded sound example, or a new idea for teaching acoustics which you are willing to share with your colleagues? At this session a sign-up board will be provided for scheduling presentations. No abstracts are printed. Presenters ar encouraged to have handouts to distribute. Multiple presentations are acceptable (not consecutively). Presentations are limited to 5 minutes. Keep them short! Keep them fun!

Session 4aMU

Musical Acoustics and Signal Processing in Acoustics: Audio Signal Data Compression for Musical Applications

James W. Beauchamp, Chair
School of Music, Department of ECE, University of Illinois Urbana-Champaign, 2136 Music Building, 1114 West Nevada, Urbana, Illinois 61801

Chair's Introduction—8:25

Invited Papers

8:30

4aMU1. Psychoacoustics as the basis for modern audio signal data compression. Tilmann Zwicker (Valpichlerstr. 70, D-80686 Munich, Germany, tzwicker@epo.org)

To almost everyone involved in music, whether they create it or listen to it, storing and transmitting music signals is of great importance. Recent progress makes it possible to store audio signals in small files and transmit them with very small transmission bandwidths, so that standardized data compression schemes, such as ATRAC (Adaptive Transform Acoustic Coding) or MP3 (Motion Picture Expert Group Layer 3 Format), are now ubiquitous. Successful development of audio storage and transmission technology has always taken into account how the final receiver of audio signals and final judge of their quality, the human auditory system, can actually appreciate them. For example, standards for frequency response and wow and flutter for analog studio tape recorders were originally designed to satisfy known limits of hearing. The use of digital signal processing to exploit newly discovered intricacies of the auditory system has increased the accuracy with which stored and transmitted audio signals can be tailored to the requirements imposed by our hearing. Such accurate matching provides the basis for highly effective audio-signal data compression. This talk provides an overview of the related idiosyncrasies of human hearing, in particular, masking effects in the frequency and temporal domains.

9:00

4aMU2. Three approaches to the perceptual evaluation of audio compression methods. Steven van de Par and Armin Kohlrausch (Philips Res. Labs. Eindhoven, Prof. Holstlaan 4, NL-5656 AA Eindhoven, The Netherlands and IPO-Ctr. for User-System Interaction, 5600 MB Eindhoven, The Netherlands

The compression of raw digital audio data is commonly accomplished by coding the audio data into a representation where part of the information present within the original data is lost. Upon playback this loss of information reveals itself in terms of various distortions, whose audibility depends on the coding method used. Therefore, the success of a coding method is determined by the human auditory perception of coding distortions. Three approaches to the perceptual evaluation of coding methods will be discussed. The first deals with the audibility of coding errors and is useful for methods that aim to achieve audio that is indistinguishable from the original. The second approach measures the degree of perceived degradation of coded audio, allowing comparison of different coding schemes which introduce different types of clearly audible distortions. The third approach deals with perceived intrinsic quality. For the assessment of quality, the observer is assumed to have no knowledge of the original signal. This approach is useful in situations where the original is not available to the observer. Here a judgment about quality is reached on the basis of the observers' expectations with regard to the type of audio signal heard.

9:30

4aMU3. A malleable audio representation for data compression. Scott N. Levine (Liquid Audio, 2221 Broadway, Redwood City, CA 94603)

The boundaries between audio data compression and audio synthesis/modifications are blurring as research in both fields progresses. New results using a sines + transients + noise parametric representation allows for coding gains as efficient as the most recent nonparametric transform coding schemes. Because the audio input is decomposed into separate sinusoidal, transient, and noiselike signals, aggressive quantization can be performed independently on each signal in a perceptually meaningful manner. While the sines + transients + noise representation allows for good compression rates, it also allows for high-quality time-scale modification in the compressed domain. The multiresolution sinusoidal modeling and the Bark-band noise modeling utilized in this representation have been used in both the music synthesis and audio/speech coding arenas. The transient modeling in this representation uses similar algorithms to those used in transform coding systems.

10:00

4aMU4. New approaches to sound compression that use algorithmic synthesis. Eric D. Scheirer (Machine Listening Group, MIT Media Lab., E15-401D, Cambridge, MA 02139-4307, eds@media.mit.edu)

Research into the use of perceptual models to compress sounds has achieved fruitful success in recent years. However, quantized filterbanks and linear-predictive coding are not the only techniques that can be used for audio compression. Recent research has tightened the connection between sound processing normally conceived as ''compression'' and that normally conceived as ''synthesis.'' By making this connection explicit, it becomes possible to apply methods taken from the broad literature on the synthesis of sound signals to applications in sound compression [B. L. Vercoe *et al.*, Proc. IEEE **85**, 622–640 (1998)]. New techniques that use audio synthesis to enable compression have recently emerged. One of these is termed *algorithmic structured audio*, and builds on research into software synthesis and sound-description languages. A particular implementation of this method forms the new MPEG-4 Structured Audio standard, in which the software-synthesis language SAOL is used to represent sound algorithmically. Recent research also suggests that the development of so-called *generalized audio coding* techniques, in which algorithmic-structured-audio formats carry customized perceptual decoders, might increase the overall efficiency of the marketplace of coding by allowing a more flexible approach to the selection of coding models.

10:30

4aMU5. Internet audio as seen by the producer/musician. Eric Somers (Dept. of Performing and Visual Arts, Dutchess Community College of the State Univ. of New York, Poughkeepsie, NY 12601)

In the battle over Internet audio transmission schemes, each format seeks to become the universal standard of the internet. Yet the author, a sound designer, composer, and producer, asserts that no one compression technology can best serve the various needs of the musician. The paper classifies these needs, suggests which schemes serve each best, and suggests musical and production issues that designers of audio compression technologies need to keep in mind.

11:00

4aMU6. Encoding considerations for MP3 and MPEG-2/MPEG-4 advanced audio coding. Karlheinz Brandenburg (Fraunhofer Institut Integrierte Schaltungen, 91058 Erlangen, Germany and Ilmenau Tech. Univ., 98684 Ilmenau, Germany)

Perceptual coding of high-quality audio has found widespread applications for broadcasting, internet delivery of music and storage of music as used in portable music players. The success of formats like MPEG-1 Layer-3 (AKA MP3) can be attributed to the standardization philosophy of MPEG: A universal format is specified and available for licensing for everybody. This leaves the encoder up to technical improvements. The presentation will (i) introduce the basics of high-quality audio coding, (ii) give an overview on MP3 and MPEG-2/4 AAC, (iii) explain encoding strategies, and (iv) give special consideration to parameters affecting coding quality.

FRIDAY MORNING, 2 JUNE 2000 TOWER 1205, 9:00 TO 10:45 A.M.

Session 4aNS

Noise: Topics in Noise—Noise and Vibration Effects and Noise Exposure Modeling

Richard L. McKinley, Chair
Air Force Research Laboratory, AFRL/HECB Building 441, 2610 Seventh Street, Wright–Patterson Air Force Base, Ohio 45433-7901

Contributed Papers

9:00

4aNS1. The effects of airborne vibration on human body vibration response. Suzanne D. Smith (Air Force Res. Lab., AFRL/HECB, 2610 Seventh St., WPAFB, OH 45433-7901, suzanne.smith@wpafb.af.mil)

Aircraft ground operations and maintenance personnel can be exposed to whole-body vibration via the airborne transmission of acoustical energy. The purpose of this study was to characterize human body vibration response during exposures to airborne vibration generated by military fighter aircraft during high-power ground engine runs. Miniature triaxial accelerometers were mounted on the head (bitebar), chest, spine, and lower leg of the subject. Measurements were made for selected locations along a line parallel to the longitudinal axis of the aircraft at specified engine-power settings. The highest accelerations occurred in the fore-and-aft (X) chest response. One-third-octave analysis showed a distinct peak in the chest between 50 and 100 Hz not observed in the noise levels, strongly suggesting the presence of a chest resonance. These peaks increased as the subject moved aft of the aircraft. While the noise levels also increased, it was difficult to determine a relationship between acceleration and noise without additional data due to differences in the results between power settings. The subject did report an increased sensation of vibration in the upper torso, which coincided with the increased noise levels. These data will be used in the development of human airborne vibration exposure guidelines.

9:15

4aNS2. A study of human response to helicopter interior noise. Brenda M. Sullivan (NASA Langley Res. Ctr., M/S 463, Hampton, VA 23681, b.m.sullivan@larc.nasa.gov), Mark W. Davis, David L. Young, Douglas G. MacMartin (United Technologies Res. Ctr., East Hartford, CT 06108), and Thomas A. Millott (Sikorsky Aircraft Corp., Stratford, CT 06615)

Two studies investigating low-frequency tonal content of rotorcraft noise were performed in a joint project between NASA Langley Research Center, United Technologies Research Center, and Sikorsky Aircraft. The first study investigated preference reported by subjects exposed to noise representing that experienced inside helicopters passenger cabins or crew compartments. The second study examined the use of computer-based tools to measure possible fatigue caused by long-term exposure to helicopter interior noise. In the first study subjects heard sounds with modified low-frequency rotor-generated tones, high-frequency gear-mesh tones, and mid-frequency broadband noise. Sounds were presented in pairs and subjects reported degree of preference. The rank ordering of preference shows the perceived benefit of reducing low-frequency tones versus higher-frequency broadband noise. In the second study subjects were exposed for one 6-h period to a helicopter-type sound environment at 88 dBA and for one 6-h period of exposure to a control condition of minimal noise exposure. During each of these test-exposure days, the subjects were required to complete computer-based tests at specified intervals throughout the day, to monitor any change in fatigue or alertness levels. The most relevant conclusions from the tests will be presented in the paper.

9:30

4aNS3. The investigation of vibroacoustical activity of the combined fuel pump. Valeri V. Lenchine (Yale Univ., P.O. Box 208206, New Haven, CT 06520-8206, valeri.lenchine@yale.edu), Alexander N. Kruchkov, Andrei B. Prokofiev, and Evgeniy V. Shakhmatov (Samara St Aerosp. Univ., Samara, 443086, Russia)

A theoretical and experimental investigation of the fuel pump for a gas turbine engine is described. The combined pump consists of blade and gear units. Under the pump exploitation the graphite bearing of the blade unit was being destroyed due to the high level of vibroacoustical loading at the resonant working conditions. The complex dynamic model of the pump was created to estimate pulsation and vibration productivity of the pump. The mechanism and sources of intensive vibroacoustical stresses were identified. The complex test of the fuel system allowed us to define parameters of pulsing liquid, vibration of the pump (including natural frequencies), oscillation of torque and rotation of the driving shaft, and sound power of the pump. The test's results have good coincidence with the modeling data and confirm the hypothesis regarding the reason for the bearing destruction. The facilities to decrease vibroacoustical stresses were proposed after analyzing the dynamical characteristics of the fuel system. The effectiveness of the facilities was proved by modeling and natural experiments. As a result of the research, vibration, pulsation, and noise activity of the pump were reduced significantly, and the defect connected with the destruction of the bearing was eliminated.

9:45

4aNS4. Psychoacoustic comparison of two methods for the evaluation of prominent discrete tones. Anne C. Balant (Dept. of Commun. Disord., State Univ. of New York at New Paltz, New Paltz, NY 12561 and IBM Hudson Valley Acoust. Lab., balanta@matrix.newpaltz.edu), Kristin Barringer (State Univ. of New York at New Paltz, New Paltz, NY 12561), and Matthew Nobile (IBM Hudson Valley Acoust. Lab., Poughkeepsie, NY 12601-5400)

Two objective methods have been developed to evaluate the potential annoyance of prominent discrete tones in the noise emissions of products: (1) the tone-to-noise ratio (TNR) method [ECMA-74-1997, ANSI S1.13-1995] and (2) the prominence ratio (PR) method [ANSI S1.13]. A task group of the Inter-Committee Working Group on noise from information technology and telecommunications equipment (ITTE) is studying both procedures, with the goal of optimizing a single method. In some cases the procedures give conflicting results, especially when there are harmonics and/or multiple tones within a single critical band. ECMA-74-1997 employs a modification of the TNR method for cases in which there are two tones within the critical band. This talk will present the results of two pilot studies comparing naive listeners' subjective ratings of signals containing discrete tones to the measured TNR and PR values. In the first pilot study, stimuli were a series of recorded machine noises. In the second study, stimuli were constructed by mixing the noise produced by axial fans with a pure tone close to the fundamental frequency of the fans. The level and frequency of the tone was varied. The results of these pilot studies and recommendations for future work will be discussed.

10:00

4aNS5. Designing process plants to meet a noise limit based on verifying by a noise model rather than measurements. Frank Brittain (Bechtel Corp., 50 Beale St., San Francisco, CA 94105, fhbritta@bechtel.com)

Normally, in the US process plants are designed to meet a noise limit based on verifying the limit has been met using noise measurements. An alternative approach is to design the plant based on confirming the noise limit has been met using a noise prediction model. The latter approach is often used in Europe and Australia. The strategy used in designing the plant depends in part on how meeting the noise limit is to be verified. Verification using a model has some distinct advantages and some disadvantages. The advantages of verification using a model include independence from variations in atmospheric conditions, and ambient levels (absolute and changes) at locations where the limit must be met. Verification using a model necessitates very close advance agreement among all parties on how the model is developed and approved. Further, the issues of experimentally verifying the model and updating the model with actual noise levels of equipment need to be addressed by all parties. These issues will be discussed, and alternative design strategies identified.

10:15

4aNS6. Use of structured, interactive interviews in retrospective noise exposure assessment in an occupational epidemiologic study. Mary M. Prince, Martha A. Waters (CDC/NIOSH, 4676 Columbia Pkwy., R-16, Cincinnati, OH 45226, mmp3@cdc.gov), Robert R. Anderson, and Richard R. James (James, Anderson, & Assoc., Okemos, MI 48864)

As part of a NIOSH study examining factors affecting hearing conservation program (HCP) effectiveness, a job-noise exposure matrix was constructed using these data sources: (1) task-based sound level survey data; (2) noise exposure data (dosimetry and sound level surveys) provided from plant historical reports; (3) information on process changes and engineering controls; (4) interviews with plant personnel (in engineering and safety departments); and (5) detailed work history data from personnel records for each employee in the plant HCP. For plants in which changes in exposures have occurred due to engineering control and process changes, exposure estimation becomes a challenge when there are data gaps in exposure and process changes. This paper discusses how data from structured employee interviews can be used in conjunction with available quantitative, records-based data to reconstruct processes and machinery/layout history. The goal was to characterize how noise exposure determinants (manufacturing environment, equipment, processes, shift lengths) have changed and to estimate exposure by department, job, and era. The first step was to identify employees for interview whose jobs were located in departments where exposure data were sparse or nonexistent. Data collection efforts were then geared towards reconstructing noise exposure for specific departments and jobs over time.

10:30

4aNS7. Using Caltrans noise analysis protocol methodology to determine insertion loss of 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 using the recently published California Department of Transportation (Caltrans) Noise Analysis Protocol. Both the school district and Caltrans agreed upon the details of the measurement methodology prior to the tests. 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 in the absence of and then inside the room 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 façade. The resultant data from these measurements required the use of specially designed spreadsheets to effectively analyze and 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 indoor noise standard for classroom spaces.

FRIDAY MORNING, 2 JUNE 2000 SPANISH ROOM, 9:00 TO 11:35 A.M.

Session 4aPA

Physical Acoustics: Acoustics of Sand, Paper and Foam

Julian D. Maynard, Jr., Chair

Department of Physics, The Pennsylvania State University, 104 Davey Laboratory, State College, Pennsylvania 16802

Chair's Introduction—9:00

Invited Papers

9:05

4aPA1. Impulse propagation in granular media. Surajit Sen (Dept. of Phys., State Univ. of New York at Buffalo, Buffalo, NY 14260-1500)

Grains can be described as elastic objects. When two grains are pushed against one another, they repel via Hertz' nonlinear force law, $F \propto \delta^n$, $n > 2$, $\delta \geqslant 0$, being the overlap between the grains. We show that the propagation of an impulse of any magnitude in a 1D chain of grains at zero loading can be described as a solitary wave. The width of the solitary wave, $L(n) \rightarrow 1$ as $n \rightarrow \infty$, $L \sim 5$ grain diameters for typical granular contacts and $L(n) \rightarrow \infty$ when $n \rightarrow 2$. The condition $\delta \geqslant 0$ leads to the formation of secondary solitary waves when two identical solitary waves propagating in opposite directions collide. Randomness in grain densities and sizes and restitutional losses lead to approximately exponential decays in distance of the energy of a propagating solitary wave. It turns out that impulses can be exploited to fingerprint a buried impurity mass in a 1D chain with and without gravity. In closing, we shall discussthe backscattering of an impulse from a buried object in 3D beds at shallow depths. Implications of this work with respect to humanitarian demining and related applications will be presented. [Work supported in part by U.S. Army Corps of Engineers and Sandia National Laboratories. Work done in collaboration with Marian Manciu (SUNY-Buffalo) and Alan J. Hurd (Sandia National Labs).]

9:25

4aPA2. The noise from a crumpled candy wrapper as a probe of a disordered system. Eric M. Kramer (Simons Rock College, Great Barrington, MA 01230)

We discuss the origin and properties of the crackling sound emitted by a crumpled sheet of Mylar as it is strained. These sheets possess many qualitative features of a traditional disordered system, including frustration and discrete memory. The sound can be resolved into discrete clicks, which are emitted during rapid changes in the conformation of the sheet. Observed click energies range over six orders of magnitude. The measured energy autocorrelation function for the sound is consistent with a stretched exponential, $C(t) = A \exp[-Bt^{0.35}]$. The probability distribution of click energies has a power law regime, $p(E) \sim 1/E$, independent of sheet size and material. We also discuss future directions for this research.

9:45

4aPA3. Avalanches, Barkhausen noise, and disorder-induced critical behavior. Karin Dahmen (Dept. of Phys., Univ. of Illinois at Urbana—Champaign, Urbana, IL 61801-3080)

Hysteresis loops are often seen in experiments at first-order phase transformations when the system goes out of equilibrium. They may have a macroscopic jump, roughly as seen in the supercooling of liquids, or they may be smoothly varying, as seen in most magnets. We have studied the nonequilibrium zero-temperature random-field Ising model as a model for hysteretic behavior at first-order phase transformations. As disorder is added, one finds a transition where the jump in the magnetization (corresponding to an infinite avalanche) decreases to zero. At this transition the model exhibits power law distributions of noise (avalanches), universal behavior, and a diverging length scale, which should be detectable in acoustic emission and other noise measurements. We study universal properties of this critical point using renormalization group methods and numerical simulations. Connections to experimental systems such as athermal martensitic phase transitions (with and without ''bursts'') and the Barkhausen effect in magnetic systems will be discussed. Similar ideas can also be applied to the interpretation of the Gutenberg–Richter scaling law in the statistics of earthquakes.

10:05

4aPA4. Attenuating stress waves during the fracture of a brittle carbon foam using ferrofluid damping. L. C. Krysac (Dept. of Phys., Univ. of the Pacific, Stockton, CA 95211)

Brittle fracture is a complex problem where details of defects and disorder on the microscopic scale affect the macroscopic strength of a material. We have found that during the fracture of disordered brittle carbon foam samples, stress waves produced by the breaking of individual struts may initiate avalanches of further strut-breaking events. If conditions are right for a macroscopic, sample-sized avalanche to be produced in this manner, the sample ruptures. If the stress waves are damped, by acoustic damping techniques, it may be possible to delay the onset of fracture in this material. The effect of damping the stress waves using viscous and magnetic ferrofluids on the measured fracture strength will be discussed.

10:25–10:35 Break

Contributed Papers

10:35

4aPA5. A laser-based ultrasonic technique to measure the dependence of the bending stiffness of copy paper on moisture and temperature. David A. Griggs and Yves H. Berthelot (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, yves.berthelot@me.gatech.edu)

A noncontact system is used to generate and detect, optically, a broadband ultrasonic pulse propagating in paper under controlled temperature and moisture conditions. The generation laser (Nd–Yag) produces a short optical pulse which is directed through an optical window on the paper sample inside an environmental chamber where temperature and relative humidity are controlled. Ultrasonic Lamb waves propagating in the paper are detected by a fiberoptic Mach–Zehnder interferometer (argon–ion laser) where the Doppler-shifted light exits the chamber through a small-core multimode fiber. The moisture content is measured by monitoring the weight of the paper sample relative to a dry sample. The dispersive nature of the A0 Lamb mode of the ultrasonic signal contains information about the bending stiffness normalized by the density and thickness of the paper averaged over the propagation path. The frequency dependence of the group velocity is found from the analytic wavelet transform of the signal (complex Morlet wavelet). [Work supported by the Department of Energy and the Institute of Paper Science and Technology.]

10:50

4aPA6. A measure of the dynamic tortuosity in stratified fibrous porous materials. Pedro E. Solana Quiros, Miguel A. Picard Lopez, and Juan V. Arizo Serrulla (Universidad Politecnica de Valencia, E.T.S.I. Industriales Edfs. D4-D5, Camino de Vera 14, 46022 Valencia, Spain, psolana@fis.upv.es)

Fibrous porous materials are an important element for noise control. The dynamic tortuosity is a magnitude that contains phenomenological and physical information of these materials and represents a tool of great usefulness for acoustical models and studies of fluid mechanics within them. From a phenomenological approach, this dynamic parameter includes the density of the fluid in the pores and the dynamic flow resistance and, from a microscopic point of view, contains information of the structure shape and the dynamic structure factor. It can be obtained theoretically through microscopic approximations, but it can also be calculated through acoustic procedures. From a phenomenological approximation in this work, an acoustic method of calculation is proposed to obtain it based on the measurement of the complex flow impedance of the thin samples. This will be more accurate when the sample is more rigid. Finally, the experimental procedure by a two-microphone method and the existence of relationships to microscopic parameters can facilitate the studies of acoustic behavior and optimization of these materials for the manufacturer as well as for the technician that employs them in projects of noise control.

11:05

4aPA7. The use of compressional and shear wave velocities to measure the shear strength of soil. F. Douglas Shields and Jim Sabatier (Natl. Ctr. for Physical Acoust., Coliseum Dr., University, MS 38677, dshields@olemiss.edu)

Compressional and shear wave velocities have been measured in soil contained in a rigid-walled cylinder in both the axial and radial direction. This oedometer cell was fitted with a piston so that the soil could be axially compressed. The velocities and axial strain in the soil were measured as the axial stress was increased from zero to 35 lbs./sq. in. Measurements were made on both dry and wet soil. The axial and radial velocities were used to determine the radial stress. The ratio of the radial to axial stress gave the coefficient of earth pressure at rest and the angle of shearing resistance. For both the wet and dry soil, this angle was approximately 33 degrees. The velocities in wet and dry soils had approximately the same dependence upon axial stress, but the wet soil was much more compressible than the dry soil. Both wet and dry compressed soils maintained their strain and strained velocities upon decompression. However, when water was introduced into the decompressed soil, it expanded significantly and the velocities dropped to their precompression values. [Work supported by the USDA.]

11:20

4aPA8. The effect of moisture on compressional and shear wave speeds in unconsolidated granular materials. F. Douglas Shields, Jim Sabatier (Natl. Ctr. for Physical Acoust., Coliseum Dr., University, MS 38677, dshields@olemiss.edu), and Mark Wang (Sensys Instruments Corp., Sunnyvale, CA 94089)

The effect of water vapor on the shear and compressional wave speeds in two different kinds of glass beads and in Ottawa sand has been measured. The nominal diameter of the glass beads was 125 μm and of the sand, 500 μm. The measurements were made as the water vapor was introduced slowly into the evacuated material. The vapor pressure isotherm for the beads made of titanium-barium glass was fit reasonably well by the simple BET theory. For the Ottawa sand the BET theory fit the vapor pressure isotherm if the surface area of the grains was assumed to be three times the area calculated, assuming all of the grains were spheres with a diameter of 500 μm. In these two materials the vapor had little effect on the wave speeds. For beads made of lime glass, however, the wave speeds approximately double with the introduction of water vapor, and the vapor pressure isotherm had the BET shape only if the saturated vapor pressure was assumed to be lowered by 20%. These results have been explained by assuming that a chemical reaction occurred between the lime glass and the water to form a gel. [Work supported by the USDA.]

FRIDAY MORNING, 2 JUNE 2000 — STATE ROOM, 8:30 TO 11:45 A.M.

Session 4aPP

Psychological and Physiological Acoustics: Complex Sound Perception

Jennifer J. Lentz, Chair

Army Audiology and Speech Center, Walter Reed Army Medical Center, Washington, D.C. 20307-5001

Contributed Papers

8:30

4aPP1. Joint detection-recognition of amplitude modulation. Stanley Sheft and William A. Yost (Parmly Hearing Inst., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626, ssheft@luc.edu)

Joint detection-recognition ability was measured for amplitude modulation. On each trial there were two subject responses, one for detection and the other for recognition. Stimuli were samples of wideband noise sinusoidally modulated at either 4, 16, 64, or 256 Hz. Conditions included all pairwise combinations of the four AM rates. Rate uncertainty led to only a slight decrement in detection ability with little effect of the extent of the separation between the two rates. Recognition performance was always poorer than detection ability with recognition near chance on trials in which the detection response was incorrect. When one of the AM rates was 4 Hz, the detection-recognition theorem [Starr *et al.*, Radiology **116**, 533–538 (1975)] provided a reasonable prediction of recognition ability. For the three other pairwise combinations of AM rate, recognition performance was poorer with the theorem failing to account for the extent of the decrement. The theorem requires orthogonality of signals. Results then suggest independence of 4 Hz versus higher rate processing, and also correlation in processing the higher rates. This pattern of orthogonality is consistent with two rate-dependent cues in modulation processing. [Work supported by NIH.]

8:45

4aPP2. An effect of auditory-filter envelope modulation variability on masking. William Treurniet (Communications Research Ctr., P.O. Box 11490, Sta. H, Ottawa, ON K2H 8S2, Canada, bill.treurniet@crc.ca)

In an earlier report [W. C. Treurniet and D. R. Boucher, J. Acoust. Soc. Am. **106**, 2146 (1999)], the threshold for a noise probe masked by a complex masker was shown to be lower when the masker partials were harmonically related than when the frequencies of the partials were perturbed by small amounts. The results suggest that the improved detection may arise from differences in the pattern of modulations across auditory-filter outputs in response to harmonic and inharmonic maskers. That is, envelope modulations are identical across filters for a harmonic masker, but the modulations vary for an inharmonic masker or a noise probe. The variable modulation rates across filters due to the noise probe are easier to detect against the invariant background rates generated by the harmonic masker than against the variable background rates resulting from the inharmonic masker. In support of this hypothesis, the difference in masked threshold was strongly related to the variance of the dominant period of envelope modulations measured across auditory-filter outputs. Further, the masking level differences were minimal when the average variance of modulation amplitude was small, as in the case of a single partial per auditory filter.

9:00

4aPP3. The effect of speech reading on masked detection thresholds for spoken sentences. Ken W. Grant (Walter Reed Army Medical Ctr., Army Audiol. and Speech Ctr., Washington, DC 20307-5001)

Detection thresholds for spoken sentences in steady-state noise are reduced by 1–3 dB when synchronized video images of movements of the lips and other surface features of the face are provided. In a previous report [K. W. Grant and P. F. Seitz, J. Acoust. Soc. Am. **103**, 3018 (1998)], we showed that the amount of masked threshold reduction, or *bimodal coherence masking protection* (BCMP), depended on the degree of correlation between the rms amplitude envelope of the target sentence and the area of lip opening. In the present study, we extend these results by directly manipulating this cross-modality correlation through either bandpass filtering or amplitude adjustments of selected words contained in the target sentences. A control condition was also included in which visual orthography was provided to explicitly identify the target sentence prior to each test trial. Results showed that orthographic information reduced detection thresholds by about 0.5 dB for all target sentences. Preliminary results for filtered and amplitude-adjusted sentences suggest that the magnitude of the BCMP depends primarily on the cross-modality correlation between lip-area function and rms amplitude envelope computed over windows of approximately 300 ms. [Work supported by NIH and the Department of Clinical Investigation, Walter Reed Army Medical Center.]

9:15

4aPP4. Masking period patterns in listeners with cochlear hearing loss. Magdalena Wojtczak, Anna C. Schroder, and David A. Nelson (Clinical Psychoacoustics Lab., Univ. of Minnesota, 516 Delaware St. SE, Minneapolis, MN 55455)

Masking period patterns (MPPs) were measured in listeners with hearing loss of cochlear origin, using a short 6-kHz probe presented at different times within the envelope of a 100% sinusoidally amplitude-modulated (AM) masker (4-Hz modulation rate). The carrier frequency of the AM masker was 3 kHz (off-frequency masking) or 6 kHz (on-frequency masking). MPPs obtained from hearing-impaired listeners were compared to those obtained from normal-hearing listeners using the same stimuli. A model used previously to simulate MPPs from normal-hearing listeners [Wojtczak *et al.*, J. Acoust. Soc. Am. **106**, 2147 (1999)] was used to predict data from the hearing-impaired listeners. In contrast to MPPs from normal-hearing ears, MPPs from ears with moderate to profound hearing loss did not exhibit large differences in MPP shapes for on- and off-frequency maskers, i.e., they did not exhibit longer MPP valleys for the off-frequency masking situation. Results of the modeling suggest that the similar shapes for on- and off-frequency MPPs reflect a lack of compression (or reduced compression) in the hearing-impaired listeners. The implications of this effect for speech understanding in the presence of competing talkers will be discussed. [Work suported by NIH NIDCD DC00149 and the Lions 5M Hearing Foundation.]

9:30

4aPP5. Monaural phase effects: Timing versus level cues. Martin P. Law, Mark A. Stellmack, and Neal F. Viemeister (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN 55455, mpl@nextear.psych.umn.edu)

As shown in previous studies, detection thresholds for a 500-Hz signal vary as a function of its phase relative to a 250-Hz masker. Due to differences in detectability, phase discrimination between suprathreshold 250- and 500-Hz components may be possible using the difference in effective stimulus level as a cue. Alternatively, phase discrimination may be based on changes in temporal fine-structure. If level were the cue, then the psychometric function [P(C) versus level of the 500-Hz component in the signal interval] should be V-shaped with a minimum at chance when the effective levels of the 500-Hz components in the signal and nonsignal intervals are equal. While the data did exhibit the predicted V-shaped function, performance dropped to chance only when the masker intensity level was below 70 dB, suggesting that fine-structure information can be used, but only above 70 dB. In another experiment, phase discrimination was assessed when the overall level of the two-tone complex was roved on each presentation. Small rove ranges significantly degraded performance, consistent with the use of a level cue. Overall, these data suggest that even at low frequencies monaural phase effects are not primarily based on fine-structure timing. [Work supported by NIDCD DC00683.]

9:45

4aPP6. Monaural phase effects in the discrimination of the spectral shape of transients. Gregory H. Wakefield (Dept. of Elec. Eng. and Comput. Sci., The Univ. of Michigan, 1301 Beal Ave., Ann Arbor, MI 48109-2110, ghw@eecs.umich.edu), Laurie M. Heller (Naval Submarine Medical Res. Lab., Groton, CT 06349-5900), Laurel H. Carney (Boston Univ., Boston, MA 02215), and Maureen Mellody (The Univ. of Michigan, Ann Arbor, MI 48109)

Limits on temporal resolution in auditory perception range from 25 ms, for the judgment of temporal order, to a lower limit of 3–4 ms for the discrimination of monaural phase. Results are presented on the discrimination of spectral shape for signals concentrated within this lower bound. A wideband, 4-ms noise was compared with spectrally smoothed versions. The phase spectrum was controlled by assigning the same random phase spectrum to both the original and smoothed signals. Depending on the choice of phase spectrum, discrimination thresholds for spectral shape were found to vary from 2–3 dB to as much as 15–17 dB, at which point the smoothed spectrum is essentially flat. This dependence on phase can be eliminated by presenting a train of transients, rather than a single transient. Such findings do not appear to be accounted for by synthesis artifact, nor by any simple features of either the complex spectrum or waveform envelope of these wideband signals. Phase effects are observed, however, in a computational model of the auditory nerve. [Work supported by the following grants from ONR: 61153.04114.00, MURI Z883402, and US DOD N66604-96-C-H366.]

10:00

4aPP7. Growth of spectral contrast enhancement in Schroeder-phase harmonic complexes. Laura E. Dreisbach, Marjorie R. Leek, and Jennifer J. Lentz (Army Audiol. and Speech Ctr., Walter Reed Army Medical Ctr., Washington, DC 20307, ldreis@attglobal.net)

Discrimination of spectral differences in harmonic complexes is better for waveforms that are highly modulated and presented at high levels relative to lower-level stimuli with flatter temporal envelopes [M. R. Leek and V. Summers, J. Acoust. Soc. Am. **94**, 2074–2082 (1993)]. This may reflect an enhancement of contrast between spectral peaks and valleys related to nonlinear cochlear processing. To further explore this relationship, discrimination of harmonic complexes with slightly different peak frequencies was measured at several stimulus levels. Peak-to-background differences required for discrimination were taken as a measure of the spectral contrast preserved in the auditory system. Phases were selected to generate either flat or peaked internal waveforms (negative or positive Schroeder phase), and peak frequency regions were near 2, 3, or 4 kHz. In normal-hearing listeners, the positive Schroeder waveforms produced lower spectral contrast thresholds than the negative Schroeder stimuli at all tested intensities and peak regions, with the greatest differences at moderate stimulus levels. Generally, hearing-impaired listeners demonstrated smaller threshold differences due to phase selection. These results are consistent with normal cochlear processing that is nonlinear at moderate levels, but more linear at low and high levels, and with more linear processing in impaired ears. [Work supported by NIH.]

10:15–10:30 Break

10:30

4aPP8. Hearing discrete elements of a temporal sequence: Learning and generalization. Robert S. Schlauch, Jeffrey J. DiGiovanni, and Dennis T. Ries (Dept. of Commun. Disord., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN 55455, Schla001@tc.umn.edu)

The ability to hear discrete elements of a temporal sequence was assessed. Subjects judged whether the first element of a three- or four-burst sequence of 50% duty cycle noise that alternated in level from low to high (or high to low) began with a low-level (60 dB SPL) or a high-level (80 dB SPL) burst. The rate was varied from trial-to-trial [selected values between 5 to 40 pulses per second (pps)] to prevent absolute duration and loudness cues. Initially, listeners were at chance performance for rates above roughly 20 pps. However, after 6 h of practice with feedback performance was near 90% for the highest tested rate (45 pps). After training with the stimulus that alternated in level, subjects were presented with a new condition; the three- or four-pulse sequences began with either a high-level or a low-level burst of noise, as before, but the level of each subsequent burst in the sequence was selected randomly between 60 and 80 dB SPL. Performance remained high for this condition, suggesting that listeners did not learn patterns of sequences but, rather, were able to hear discretely the first element in the sequence. [Work supported by NIH.]

10:45

4aPP9. Specificity of learning in an auditory temporal-order task. Beverly A. Wright and Julia A. Mossbridge (Audiol. and Hearing Sci. Prog., 2299 North Campus Dr., Northwestern Univ., Evanston, IL 60208-3550)

The ability to determine the onset order of two tones at different frequencies is a measure of temporal acuity and an analog of voice-onset time. Learning in this temporal-order task and its generalization to untrained frequencies and temporal-acuity tasks was explored. Temporal-order thresholds of six listeners trained 1 h per day for 6 days with tones at 0.25 and 4 kHz improved from 58 ms before to 12 ms after training, yielding a proportional improvement greater than on all untrained conditions. Comparisons between proportional improvements of trained and control listeners revealed that trained listeners learned considerably more than controls on the trained condition, and only slightly or no more than controls on: (1) the trained task at two sets of untrained frequencies (0.5 and 1.5 kHz, 0.75 and 1.25 kHz), and (2) three untrained temporal-acuity tasks (offset temporal order, onset asynchrony, offset asynchrony) at the trained frequencies. The lack of robust generalization to untrained frequencies and tasks suggests learning in onset temporal-order tasks is mediated by a frequency-dependent mechanism different from the mechanism(s) underlying performance in other temporal-acuity tasks. By increasing our understanding of plasticity in temporal-acuity mechanisms, these data may guide the treatment of related disorders. [Work supported by NIDCD.]

11:00

4aPP10. Attention and the build-up of auditory stream segregation. Robert P. Carlyon, Rhodri Cusack, Jessica M. Foxton, and Ian H. Robertson (MRC Cognition and Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 2EF, UK)

In the baseline condition of experiment 1, subjects were presented with a repeating ABA–ABA sequence of 125-ms tones in their left ear, and indicated, throughout the entire 20-s sequence, whether they heard one or two streams. The proportion of ''two stream'' responses increased during the course of each sequence. This build-up of stream segregation was greatly reduced when, for the first 10 s of each sequence, subjects performed a competing task on a train of noise bursts presented to the right ear. The competing task involved identifying the amplitude envelope of each noise burst as either rising or falling. The noise bursts had no effect in a third condition, in which subjects were instructed to ignore them. Experiment 2 showed that a substantial build-up of streaming can still be observed when subjects switch tasks halfway through the tone sequence, provided that both tasks require them to continually attend to the tones. We conclude that attention is crucial for the build-up of stream segregation, and that our results are inconsistent with models of streaming based purely on peripheral, ''automatic'' neural mechanisms.

11:15

4aPP11. The influence of noise, harmonic structure, and temperament on tonal consonance judgments. James Kondash and Rodney Hallgren (Dept. of Psych., Wright State Univ., Dayton, OH 45435, jkondash@sdl.psych.wright.edu)

Although a large body of research exists on tonal consonance in which dyadic musical intervals have been systematically rank ordered via psychophysical experiments and models, none of these studies has addressed how tonal consonance judgments may change in the presence of a masking stimulus. Presenting musical intervals and maskers simultaneously could reduce elements potentially contributing to dissonance, such as beats between upper partials or distortion products. This study investigated consonance ratings of musical dyads as influenced by (1) the level of broadband and band-limited noise, (2) the number and level of upper partials in the tones, and (3) the use of equal or just intonation. In a randomized factorial experiment, participants rated musical dyads in the various conditions on a consonance/pleasantness scale. Preliminary results for equal-tempered dyads presented in the quiet show good agreement with consonance ratings reported in similar studies [e.g., Kameoka and Kuriyagawa, J. Acoust. Soc. Am. **45**, 1460–1469 (1969)]. The presence of noise decreased consonance ratings across most intervals. [Work supported by a NASA fellowship.]

11:30

4aPP12. Extended perceptual spaces for pitched and percussive timbres. Stephen Lakatos (Washington State Univ., 14204 NE Salmon Creek Ave., Vancouver, WA 98686, lakatos@vancouver.wsu.edu) and James Beauchamp (Univ. of Illinois at Urbana—Champaign, Urbana, IL 61801)

A series of listening tests attempted to better isolate higher-order perceptual dimensions of timbre by building on past findings showing that spectral centroid and rise time represent principal acoustic correlates of the primary timbral dimensions. Listeners with varying levels of musical training rated the timbral similarity of three sets of pitched and percussive instrument sounds that were equated for centroid and rise time using signal processing techniques. Multidimensional scaling analyses yielded two- and three-dimensional perceptual spaces for the three stimulus sets. Several aspects of the spectral fine structure of the timbres correlated with the dimensions of these spaces, but the nature of the acoustic correlate varied somewhat depending on the stimulus set in which the timbre was presented. The results suggest that additional perceptual dimensions of timbre exist, but that their precise acoustic correlates are context dependent and therefore less perceptually ''primary'' than centroid and rise time. [This research was funded by Air Force Office of Scientific Research Grant No. F49620-99-1-0293.]

Session 4aSA

Structural Acoustics and Vibration: Acoustics of Uncertain Structures

Joseph W. Dickey, Cochair
Center for NDE, Johns Hopkins University, 3400 North Charles Street, Baltimore, Maryland 21218

Alison B. Flatau, Cochair
Aerospace Engineering and Engineering Mechanics, Iowa State University, 1200 Howe Hall, Ames, Iowa 50011

Contributed Papers

8:00

4aSA1. A hybrid finite-element formulation for midfrequency analysis of systems with excitation applied on short members. Xi Zhao and Nickolas Vlahopoulos (Dept. of Naval Architecture and Marine Eng., Univ. of Michigan, 2600 Draper Rd., Ann Arbor, MI 48109-2145)

The hybrid finite-element analysis combines conventional finite-element analysis with energy finite-element analysis for midfrequency computations. A hybrid FEA fomulation has been published for systems with excitation applied on long members. [N. Vlahopoulos and X. Zhao, ''A Basic Development of a Hybrid Finite Element Method for Mid-Frequency Computations of Structural Vibrations,'' AIAA J. **37**(11), 1495–1505 (1999).] In this presentation, theoretical developments associated with applying the excitation on short members will be discussed. Validation cases will be presented for several systems by comparing analytical solutions to hybrid FEA results. It will be demonstrated that the resonant behavior of the short members and the interaction between long and short members are captured correctly by the hybrid FEA. [Work supported by the Automotive Research Center established at the University of Michigan by the US Army Tank Automotive Command.]

8:15

4aSA2. Investigation of power flow in the midfrequency range for systems of colinear beams based on a hybrid finite-element formulation. Xi Zhao and Nickolas Vlahopoulos (Dept. of Naval Architecture and Marine Eng., Univ. of Michigan, 2600 Draper Rd., Ann Arbor, MI 48109-2145)

A hybrid finite-element analysis (hybrid FEA) is employed for investigating power-flow characteristics for systems of colinear beams in the midfrequency range. The importance of capturing power reinjection and power reradiation effects in the solution is demonstrated. The dependency of the power-flow characteristics of a system in the midfrequency range on the rigidity, mass, and damping properties of its components is determined. Both the hybrid FEA and analytical solutions are employed for analyses in order to establish the viabilty of the hydrid FEA as a simulation technology in the midfrequency range. [Work supported by the Automotive Research Center established at the University of Michigan by the US Army Tank Automotive Command.]

8:30

4aSA3. Structural-acoustic analysis of 3-D curved shell with a discontinuity using analytical/numerical matching (ANM). Christopher Park, Linda Franzoni, and Donald Bliss (Duke Univ., Box 90300, Durham, NC 27708-0300)

Analytical/numerical matching (ANM) is used to determine the structural vibration of a curved shell driven at a single support. The ANM solution decomposes the problem into global, local, and matching subproblems. The global problem addresses large-scale effects with structural discontinuities replaced by smooth distributed forces. The local problem models the rapidly changing region around a structural discontinuity. These constituent problems are solved independently by the most efficient method available. Here, the local problem is solved numerically using 3-D solid finite element analysis (FEA) and the matching problem is solved analytically, using the Love–Timoshenko shell equations. The global problem is decomposed modally in azimuth, leaving a corresponding FEA problem in the axial direction for each mode. Due to the smoothness of the global problem, relatively few modes are required for the modal decomposition and the affiliated FEA problem can be solved using a low resolution. An advantage of the ANM method is its ability to capture the detailed response of structures with geometric complexities without having to incorporate the complexity (via higher-order elements or high mesh densities) in the modeling of the large global problem. [Work sponsored by ONR.]

8:45

4aSA4. Vibration power transmission coefficients for the coupling of circular cylindrical shells to flat plates. Benjamin F. Willis and Courtney B. Burroughs (Grad. Prog. in Acoust., The Pennsylvania State Univ., P.O. Box 30, State College, PA 16804)

Bending and in-plane waves are coupled by curvature of structures and at junctions between structures. To estimate the effects of wave coupling on the transmission of vibrational power through junctions between curved and flat elastic structures, an analytical model of circular cylindrical shells coupled to flat plates along a circumference of the shell was developed. Donnell–Mushtari shell equations are used to model the infinite and semi-infinite cylindrical shells. Classical theory is used to model both bending waves and in-plane longitudinal and shear waves in the plates. Models include plates inside and outside the shell as well as continuous through the shell. All plates are perpendicular to the shell surface. Plates coupled to both semi-infinite shells, where the shell terminates at the junction with the plate, and to infinite shells are modeled. Power transmission coefficients are presented as a function of circumferential mode number and frequency for different types of incident waves either in the shell or in the plate, and for different shell/plate configurations.

9:00

4aSA5. A new algorithm for modal identification using mode isolation. Michael V. Drexel and Jerry H. Ginsberg (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30322-0405)

Multiple degree of freedom (MDOF) algorithms are the dominant methods for identifying modal properties from measured response data. Although there are numerous MDOF methods, all are based on the notion that because the response of a linear dynamic system is the sum of many modal contributions, the extraction technique must deal with all of the modal parameters in a simultaneous fashion. Thus, these methods do not exploit the fact that each mode has unique characteristics that can be used to isolate that mode from the contribution of the other remaining modes. The mode isolation method proposes an algorithm that extracts the modal parameters of each mode in a recursive search, and then refines the estimation of each mode by isolating its effect from the other modal contributions. The mode isolation method has an advantage over MDOF methods in that MDOF algorithms require an *a priori* guess of the number of significant modes, or degrees of freedom, of the system. An error in that guess can have serious consequences with respect to the accuracy of the extraction process. Several MDOF methods attempt to remedy this situation by utilizing peak counting to find the number of modes contained in the system. In contrast, the mode isolation method offers a simple and consistent modal extraction methodology that is inherently automatic in nature. A numerical example of the mode isolation method is presented for a system that was previously studied with the eigensystem realization algorithm (ERA) and enhanced ERA. Eigenvalues and mode shapes are compared for each algorithm. The results suggest that the mode isolation method is more robust in the treatment of noisy data.

9:15

4aSA6. Causal recovery of the nonminimum phase from the measured magnitude of one-dimensional acoustic reflections. Cory L. Clarke and J. Gregory McDaniel (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

A new approach is presented for obtaining a reflection coefficient phase from magnitude using frequency-domain causality relations. This is useful because in many situations the reflection coefficient phase is more difficult to accurately measure than the magnitude; however, a knowledge of the phase is often vital to understanding the dynamics of the reflecting object. If a transfer function is minimum phase, then causality requires that its magnitude and phase are related by the Hilbert transform. Such relations do not exist for nonminimum phase-transfer functions. Previous work found that reflection coefficients are not necessarily minimum phase [McDaniel, J. Acoust. Soc. Am. **105**, 2710–2716 (1999)]; however, the drive-point impedance of a passive reflecting object is. For one-dimensional systems, the drive-point impedance of the reflecting object is algebraically related to the reflection coefficient. The approach combines this relationship with causality relations for the impedance to find the complex reflection coefficient given only its magnitude. It is applied to analytic data for a one-dimensional structural acoustic system so that the actual phase may be compared to that predicted by the method. Extensions of the approach to multidimensional scattering will be discussed. [Work supported by ONR.]

9:30

4aSA7. Determining impact source location in structural networks. Joseph Dickey (Ctr. for Nondestruct. Eval., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218) and Gideon Maidanik (Taylor Model Basin, Carderock, MD 20817)

An impact, or similar event, generates a transient in a structural network consisting of connected one-dimensional systems. A sensor, attached at an arbitrary point in the network, will ''hear'' a complicated time series of pulses. A few of these pulses are recorded and used to generate a time-reversed series which is injected at the sensor location. If the network is stationary in time, the impulse response function (IRF) can be used to show that the introduced series will constructively interfere at the original impact point and this location can thereby be determined. The procedure is demonstrated computationally using a time-domain IRF. The relationship between the above and ''reciprocity'' and ''inverse filtering'' will be discussed.

9:45

4aSA8. The influence of substructure modeling on the predicted vibroacoustic response of a fluid-loaded plate system. W. Steve Shepard, Jr. (Mech. Eng. Dept., The Univ. of Alabama, Box 870276, Tuscaloosa, AL 35487, sshepard@coe.eng.ua.edu) and Kenneth A. Cunefare (Georgia Inst. of Technol., Atlanta, GA 30332)

When predicting the vibroacoustic behavior of complex systems, one must decide how much effort to spend on modeling the features with dimensions smaller than those of the primary structure. To investigate these issues, this work examines changes in the vibroacoustic response of a semi-infinite fluid-loaded plate due to variations in modeling details associated with an attached substructure. The substructure consists of a smaller plate supported by springs along each edge. To examine the impact of spring scales, the manner in which the elasticity is distributed over the main plate is varied. Substructure modeling issues are examined by varying the number of degrees-of-freedom allowed in the substructure model. Both the combined system response and acoustic radiation from the main plate are computed using the Acoustic Surface Variational Principle and Hamilton's Principle. Furthermore, sensitivity relationships that express changes in the system response to changes in the scale of the spring elements are presented. For the cases considered, it is shown that details associated with the scale of the spring element are only important for frequencies near or below the resonances of the isolated subsystem. Furthermore, only the dynamics of the substructure that includes rigid-body-type motions is important. [Work supported by Georgia Tech.]

10:00–10:15 Break

10:15

4aSA9. Improved power input methods in SEA modeling. Robert C. Haberman (BBN Systems and Technologies, Union Station, New London, CT 06320, rhaberman@bbn.com)

The traditional method for determining SEA input power to structures subjected to point-force excitation is to consider the directly driven structural element as infinite. This allows for a relatively simple calculation of input power. For example, the power input to three-dimensional frame structures is to treat the driven beam as infinite and use the corresponding real part of the infinite beam mobility along with the mean-square force. Similarly, the power input to a structure consisting of interconnected plates is to likewise consider the driven plate as infinite and use the corresponding real part of the infinite plate mobility and force. Errors are introduced by this method when the drive point approaches the boundaries (in terms of flexural wavelengths) of interconnected structural elements. At these locations, the power is influenced by near-field effects and the transmission of power into the connected structure. To reduce these errors, a method based on modeling the driven member and directly connected structural elements is investigated. All the members are modeled as semi-infinite. Examples are given of interconnected beam and plate structures, and guidelines are introduced for improved SEA power input calculations.

10:30

4aSA10. Sound transmission through finite laminated composite panels using statistical energy analysis. Avinash R. Patil and Malcolm J. Crocker (Mech. Eng. Dept., 201 Ross Hall, Auburn Univ., Auburn, AL 36849)

Laminated composite panels are widely used in aerospace structures and other applications where weight reduction of the structure is desired without compromising the strength. They can be made up of a number of layers of unidirectional laminas. Sound radiation and sound transmission properties of these panels are different than those for isotropic panels. This is because of the variation of the material properties along the direction of bending wave propagation. During this work, analysis of those finite laminated composite panels which exhibit uncoupling of bending/twisting phenomenon is done. A bending wave number diagram is drawn to study the variation of bending wave number with respect to the angle of bending wave propagation for a given frequency. Instead of a single critical frequency as in the case of an isotropic panel, a band of critical frequencies is observed. In this band of critical frequencies, coincidence occurs only at a certain angle of bending wave propagation for a particular frequency. Frequency average radiation efficiency of such a panel is predicted by considering the variation of wave number along different directions of bending wave propagation. Modal density is predicted from the bending wave number diagram. Transmission loss of the panel is predicted using the statistical energy method.

10:45

4aSA11. Partitioning, substructuring, and imposition of impedance: Are they the same? John J. McCoy (The Catholic Univ. of America, Washington, DC 20064)

Formulation partitioning, substructuring, and the imposition of an impedance operator, or DtN map, all refer to solution methods with the same purpose. This is to distinguish those coordinates necessary for describing the response of a structure, which are not directly forced by an external agent, and to eliminate these as primary variables in a reduced formulation. It is increasingly common to encounter use of the three terms interchangeably. This raises the question. Are the solution methods identical? Fundamental differences are identified and the significance of these differences is discussed.

11:00

4aSA12. Probabilistic and possibilistic models of uncertainty in structural dynamics. Dexter F. Johnson and Robin S. Langley (Dept. of Eng., Univ. of Cambridge, Cambridge CB2 1PZ, UK)

The prediction of structural reliability under static loading has received much attention in the field of civil and structural engineering, and well-established methods such as FORM and SORM are available to estimate the failure probability. If the quality and quantity of the available data do not justify a probabilistic analysis, then the reliability can alternatively be assessed by using less precise ''possibilistic'' methods such as interval analysis or convex modeling. None of these probabilistic or possibilistic methods has as yet been applied extensively to noise and vibration problems, and the present work considers this issue. Initially it is shown that the probabilistic and possibilistic algorithms can all be expressed in the form of a constrained optimization problem, and the methods are then applied to a vibration problem involving a beam with an uncertain mass distribution. The beam is modeled both experimentally and theoretically to highlight various types of uncertainty that can arise: in addition to ''controlled'' uncertainties involving adjustable lumped mass positions, there are ''uncontrolled'' uncertainties arising from items such as boundary conditions and the support stiffnesses. The relative merits of the various reliability assessment techniques are discussed in light of this example.

11:15

4aSA13. The vibration of uncertain structures: Random matrix theory and non-Poisson statistical models of the system's natural frequencies. Robin S. Langley (Dept. of Eng., Univ. of Cambridge, Cambridge CB2 1PZ, UK)

At medium to high frequencies, the noise and vibration levels in an engineering structure can be sensitive to small changes in the properties of the system. This means that the response of a set of nominally identical units (for example, automobiles) from a production line can be radically different, and knowledge of the response statistics would significantly aid quality assurance and control. Unfortunately, the prediction of the noise and vibration response statistics for a complex built-up random structure is an extremely difficult task, as is the associated problem of determining the statistics of the natural frequencies and mode shapes. In recent years it has been suggested that progress might be made by employing the techniques of random matrix theory previously developed for applications in nuclear physics. In particular, the Gaussian Orthogonal Ensemble (GOE) has been shown to yield good agreement with the statistics of the natural frequencies of various metal blocks. In this paper the applicability or otherwise of random matrix theory to realistic engineering structures is considered, and a comparison is made with other approaches, such as a non-Poisson point process model of the natural frequencies. Several examples are considered to illustrate the theoretical developments.

11:30

4aSA14. Le Chatelier's Principle in noise control. G. Maidanik (Carderock Div., Naval Surface Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817-5700)

Le Chatelier's Principle is perhaps better known to those who studied chemistry at the university. Therefore, I find that few in the noise control community are familiar with this principle. This paper features some examples of noise control problems in which ''the Principle'' plays a role. For noise control purposes and for the record, Le Chatelier's Principle states that ''when a dynamic system is modified with the intention of achieving a high degree of change in the noise issued by this dynamic system, the achievement is mitigated by the implementation of the modification.''

11:45

4aSA15. When is a ''fuzzy'' not a fuzzy (continued)? M. Strasberg (Code 702, David Taylor Model Basin, NSWCCD, West Bethesda, MD 20817-5700)

The term ''fuzzy substructures'' as originally coined by C. Soize includes four categories of substructures whose influence on the vibratory behavior of the master structure to which they are attached may be significantly different. Either (1) the substructures are sufficiently numerous and their modal antiresonant frequencies sufficiently close to each other to result in modal overlap over the frequency range of interest; or (2) they do not have modal overlap; also, either (a) successive modal frequencies are separated by a constant frequency increment, or (b) the frequency increments are irregular. The effect of category (1) substructures on the master structure is independent of the detailed spacing between modal frequencies, provided modal overlap exists, for both steady-state and transient excitation. Contrariwise, the effect of category (2) substructures on the transient behavior of the master structure does depend on whether or not the modal frequencies are equally spaced. Although the statistical average behavior of an ensemble of such (2b) substructures may be similar to that of category (1), individual members of the ensemble may behave quite differently from their average. The differences will be exhibited by examples. To avoid confusion, it is suggested that the term ''fuzzy structures'' be limited to substructures with modal overlap.

Session 4aSC

Speech Communication: Hearing Impairment and Audio-Visual Cues in Speech (Poster Session)

Lynne E. Bernstein, Chair
House Ear Institute, 2100 West Third Street, Los Angeles, California 90057

Contributed Papers

All posters will be on display from 9:00 a.m. to 11:30 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:15 a.m. and contributors of even-numbered papers will be at their posters from 10:15 a.m. to 11:30 a.m.

4aSC1. Combining acoustic and electric hearing: Simulations and real-patient results. Christopher W. Turner, Bruce J. Gantz, Sue A. Karsten, and Bom Jun Kwon (Univ. of Iowa, Iowa City, IA 52242)

For patients with severe high-frequency sensorineural hearing loss, providing audible high-frequency, acoustic speech information often results in no increase, or in some cases a decrease, in speech recognition performance. One approach to this problem is to transmit the high-frequency speech information via a multichannel cochlear implant located in the base of the cochlea, while allowing the patient to continue to use the remaining low-frequency acoustic hearing. Data from simulation experiments are presented to show the amount of speech information that can be transmitted via this combination. The simulation of electric hearing was accomplished by using multiple channels of narrow-band noise modulated by the high-frequency speech signal. In general, speech recognition scores have the potential to be quite high if low-frequency (acoustic) information (below 500 or 1000 Hz) is combined with electrical stimulation for frequencies of 3000 Hz and above. An adverse effect of a mismatch between the speech frequencies and carrier band frequencies was observed, similar to the effect of assigning ''wrong'' frequencies to the cochlear implant. Some promising results from several human patients who have been implanted with this device are also shown. [Work supported by NIDCD.]

4aSC2. Bone conduction transmission and head-shadow effects for unilateral hearing losses fit with transcranial CIC hearing aids. Marc A. Fagelson (Dept. of Commun. Disord., East Tennessee State Univ., Johnson City, TN 37614), Colleen Noe, Jennifer Blevins, and Owen Murnane (James H. Quillen VAMC, Mountain Home, TN 37694)

Bone conduction transmission and head-shadow effects were determined with transcranial completely-in-the-canal (TCCIC) CROS hearing aids. Five subjects with documented profound unilateral hearing loss and experience with traditional CROS/BICROS fittings (TCROS) were tested with a CIC hearing aid placed in their poorer ear. Peak SPL was measured at the tympanic membrane and ranged from 105–115 dB SPL at 2000 Hz. Pure-tone crossover thresholds and functional gain tested at frequencies from 250–8000 Hz varied considerably more than the SPL measures. The pure-tone results indicated that sensitivity in the better ear was moderately associated with functional gain across frequency. Speech recognition was then tested in the sound field in two conditions: direct (noise in the poorer ear, speech in the better ear) and indirect (noise in the better ear, speech in the poorer ear) at S/Ns of -6, 0, $+6$, $+12$, and quiet. The TCCIC fittings were more effective than TCROS aids across S/Ns, particularly in the direct condition. In the indirect condition, the two fittings performed similarly. When data were pooled across conditions, the TCCIC aids provided better word recognition than the TCROS aids, particularly for those subjects with greater sensitivity in the better ear.

4aSC3. Phoneme recognition before and after hearing aid fitting. Amy T. Neel and Ayaskanta Rout (Dept. of Audiol. and Speech Sci., Purdue Univ., 1353 Heavilon Hall, West Lafayette, IN 47907-1353, atneel@purdue.edu)

Many listeners with sensorineural hearing loss experience only modest increases in speech recognition after receiving their hearing aids and very small increases in hearing aid benefit up to one year after hearing aid fitting. In this study, phoneme recognition patterns before and after hearing aid fitting were examined to determine which phonemes improved most with amplification and which phonemes continued to improve with hearing aid experience. Preliminary results using a nonsense syllable recognition test revealed that voiceless stops and fricatives improved the most immediately after hearing aid fitting although accuracy remained below 50% for these phonemes. Aided recognition for these phonemes, however, improved very little from one month to one year postfitting. Variability in phoneme recognition before and after hearing aid fitting across subjects was large. Phoneme recognition results from an ongoing study of first-time hearing aid users with comparable levels of signal processing will be presented. Differences in phoneme recognition before and after hearing aid fitting can help explain differences in hearing aid benefit across hearing aid users. In addition, phoneme recognition profiles can be used to design individualized, efficient auditory training programs for new hearing aid users.

4aSC4. The vowel spaces of normal-hearing listeners and patients with cochlear implants. James D. Harnsberger, David B. Pisoni (Speech Res. Lab., Dept. of Psych., Indiana Univ., Bloomington, IN 47405), Mario A. Svirsky, Adam R. Kaiser (Indiana Univ. School of Medicine, Indianapolis, IN), and Richard Wright (Univ. of Washington, Seattle, WA)

Cochlear implant (CI) users show substantial individual differences in their ability to understand speech in general, and vowels in particular. These differences may result from widely different abilities in identifying formant frequencies or in adapting to the more basal than normal spectral information presented by the implant. In this study, we administered a vowel perception test, using a method-of-adjustment (MOA) paradigm, to 8 CI users and 43 normal-hearing listeners. The MOA vowel test consisted of 330 steady-state synthetic vowel stimuli, varying in $F1$ and $F2$, arranged in a visual two-dimensional grid. Subjects were asked to label and rate on a 7-point scale those stimuli that matched the vowels contained in ten visually-presented words, ''heed,'' ''hid,'' ''aid,'' ''head,'' ''had,'' ''hut,'' ''odd,'' ''whod,'' ''hood,'' ''owed,'' and ''odd.'' Plots of subjects' responses for all ten words constituted the vowel spaces of the subjects. With one exception, no systematic shift was observed across all vowel categories of CI users, suggesting that these subjects were able to

adapt completely to the spectral shift introduced by the implant. However, the CI users' spaces differed substantially from normal vowel spaces in terms of the relative size of the vowel categories and their location in perceptual space.

4aSC5. Perception of prosodic phrasing by hard-of-hearing listeners.
Dragana Barac-Cikoja, Sally Revoile, and Michelle Waters (Gallaudet Univ., 800 Florida Ave. NE, Washington, DC 20002, Dragana.Barac-Cikoja@gallaudet.edu)

Severely to profoundly hard-of-hearing (HOH) listeners ($n=12$) were tested for their perception of prosodic phrase continuity in clear and conversational speech. In two experiments, fluently spoken sentences were compared with utterances artificially assembled from isolated words (experiment 1) and prosodic phrases (experiment 2) that replicated the sentences' lexical composition. The difference in the overall duration of the assembled and fluent utterance was eliminated via signal processing, but the amplitude envelope and the intonational contour differences were left intact. Randomized repetitions of each utterance pair (72 pairs, 12 sentences $\times$ 2 speaking styles $\times$ 3 speakers, in experiment 1, and 40 pairs, 10 sentences $\times$ 2 speaking styles $\times$ 2 speakers, in experiment 2) were presented to each listener over several experimental sessions. In a 2IFC procedure, the listener identified the utterance that sounded more like fluent speech. Four normal-hearing (NH) controls were tested in white noise (S/N ratio$=-20$ dB). NH listeners identified the fluent sentences with accuracy greater than 90%. Across HOH listeners, identification accuracy varied from completely random to less than 75% correct, and was dependent on a listener's sensation level and preferred communication mode (oral versus manual). [Work supported by NIDCD.]

4aSC6. Auditory–visual context effects in a speech and nonspeech condition.
Linda W. Norrix, Iris Oved (Dept. of Psych., Univ. of Arizona, Tucson, AZ 85721), and Lawrence D. Rosenblum (Univ. of California, Riverside, CA 92521)

Perception of an auditory continuum can be changed by a preceding visual context [Norrix and Green, in Proc. AVSP'99, Santa Cruz, CA, August, 1999]. This study asked whether this perceptual shift requires that listeners experience the stimuli as speech. A sine-wave continuum (/ara–ala/) was presented to three groups of trained participants for identification (auditory-only condition). Tokens from the continuum were also paired with a point-light display of a talker saying /aba/ (auditory–visual condition) and presented to the same listeners for identification. Observers in group one (speech) identified the sounds as containing an /r/ or /l/ and reported after the experiment that they heard the sounds as speech. Group two, also instructed to identify the speech sounds as containing /r/ or /l/, reported they did not hear the sounds as speech. Group three (nonspeech) was instructed to identify the environmental sounds as most similar to the ''first'' or ''last'' sound of the continuum. Results indicated a reliable shift for the auditory–visual compared to the auditory-only identification function only in the speech group, suggesting that auditory–visual context effects might depend on observers interpreting the stimuli as speech. [Work supported by NSF Grant #SBR9809013 awarded to Kerry P. Green (deceased).]

4aSC7. Effects of auditory and visual stimuli on motor facilitation of speech muscles.
Aravind, N. K. (Dept. of Speech Pathol., Univ. of Toronto, Tanz Neurosci. Bldg., 6 Queens Park Crescent W., Toronto, ON M5S 3H2, Canada, a.namasivayam@utoronto.ca), Megha Sundara (McGill Univ., Montreal, QC H3G 1A8, Canada), and Robert Chen (Toronto Western Hospital, Toronto, ON M5T 2S8, Canada)

Transcranial magnetic stimulation (TMS) was applied to the left motor cortex during presentation of video and audio speech stimuli. Motor evoked potentials (MEPs) were recorded from the right orbicularis oris muscle which is activated during pronunciation of the consonant /ba/ but not the consonant /ta/. Subjects were instructed to pay attention to six types of stimuli video presentation of the consonant /ba/, audio presentation of /ba/, audio and video presentation of /ba/, video presentation of /ta/, audio presentation of /ba/ with time-locked video presentation of video /ta/, and the face of the speaker when he was silent. The MEPs were enhanced only when subjects were visually presented with the consonant /ba/. Auditory presentations of the same consonant failed to elicit similar enhancement. The results from visual observation of the consonant /ba/ support other studies of motor facilitation in limb control research in monkeys and humans. In these studies observation of a motor action present in the repertoire of the animal results in enhanced MEPs representing automatic, covert retrieval of the matching action in the premotor cortex. This retrieval is, however, modality dependent. Auditorily presented signals do not trigger action retrieval from the premotor cortex in an identical way.

4aSC8. Development of a facility for simultaneous recordings of acoustic, optical (3-D motion and video), and physiological speech data.
Lynne E. Bernstein, Edward T. Auer, Jr., Brian Chaney (Dept. of Commun. Neurosci., House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057, lbernstein@hei.org), Abeer Alwan, and Patricia A. Keating (Univ. of California—Los Angeles, Los Angeles, CA 90095)

A multidisciplinary, multilaboratory project is underway whose focus is optical and acoustic phonetic signals and their relationships to each other in speech production and perception. The goals are to quantitatively characterize optical speech signals, examine how optical phonetic characteristics relate to acoustic and to physiologic speech production characteristics, study what affects the intelligibility of optical speech signals, and apply obtained knowledge to optical speech synthesis and automatic speech recognition. We describe the data acquisition facility, which includes recording of video, acoustic, electromagnetic midsaggital articulography, and 3-D motion capture data simultaneously and in synchrony. We also outline the design of the database of recordings being obtained and show examples from the database. [Work supported by NSF 9996088.]

4aSC9. Confidence ratings in auditory–visual speech perception.
Bart R. Clement, Sarah K. Erickson, Su-Hyun Jin, and Arlene E. Carney (Dept. of Commun. Disord., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455)

In an earlier report, Carney *et al.* [J. Acoust. Soc. Am. **106**, 2270 (1999)] showed that the perception of the McGurk effect, the result of a visual syllable biasing the perception of a simultaneously presented auditory syllable, was affected by the use of multiple talkers. In addition, individual listeners could be strongly auditory in their perception, even when a visually biasing stimulus was present. During the original experiment, listeners were asked to rate each of their perceptions in the auditory-only (AO), visual-only (VO), and auditory–visual (AV) modes for syllabic stimuli. Listeners were very confident of their perceptions for bilabial stimuli in the VO mode but not for alveolar or velar stimuli. Confidence ratings were also high for all AO stimuli and for AV stimuli for matching visual–auditory tokens. However, confidence ratings varied for tokens with mismatched A and V stimuli. Some individual listeners were always confident in their perception regardless of speaker or condition. Other listeners varied in confidence with speaker and condition.

Across listeners, certain talkers evoked different ratings of confidence as well. Results support the notion of a graded perception of bimodal stimuli. [Work supported by NIDCD.]

4aSC10. Recognizing words from optical phonetic signals. Sven Mattys, Lynne E. Bernstein, and Edward T. Auer, Jr. (Dept. of Commun. Neurosci., House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057, smattys@hei.org)

Lipreading is thought to be extremely difficult, and the difficulty is attributed to lack of perceptual distinctiveness among most visible spoken words. Therefore, it is thought that accurate lipreading requires reliance on postlexical processes, such as guessing. However, we [E. T. Auer and L. E. Bernstein, J. Acoust. Soc. Am. **102**, 3704–3710 (1997)] predicted, based on computational modeling of the lexicon, that many words maintain their visual distinctiveness. In this study, we took only visual phoneme confusability and word frequency into account to predict the relative intelligibility of lipread words. We predicted that words without visual lexical competitors can be perceived as readily as heard words, and a systematic change in intelligibility would be observed as a function of word frequency and number of competitors. Predictions of near-100% accuracy for many words were confirmed in a perceptual word identification experiment with skilled deaf American lipreaders. Accuracy was not influenced by word length. When phoneme confusability in words was statistically controlled, the number of lexical competitors correlated significantly with word score, as did word frequency. These results will be compared with those of hearing participants, and the implications discussed with attention to bottom-up processes versus lexical competition in spoken word recognition. [Work supported by NIH/NIDCD 02107.]

4aSC11. On the visual distinctiveness of English words. Margaret MacEachern (Dept. of Linguist., Univ. of Pittsburgh, Pittsburgh, PA 15260)

Lipreading is facilitated when words are visually distinct. It has been established [Auer and Bernstein, J. Acoust. Soc. Am. **102**, 3704–3710 (1997)] that words in the English lexicon manifest a low degree of visual similarity: most words remain visually unique, even when distinctions are collapsed across phoneme classes, as is characteristic of visually perceived speech. But, is the high degree of visual uniqueness a distinctive (nonaccidental) property of the English lexicon? Perhaps not: the lexical space of English is large, both in crosslinguistic terms and in absolute terms. In other words, many lexical slots exist; a fairly low percentage of them are filled by actual words [Hockett, 288–290 (1958)]. Therefore, the high degree of visual distinctiveness might be an ''accidental'' property of the lexicon, deriving from the segment inventory and phonotactics of the language. This study will explore this question by comparing the visual similarity properties of words in simulated (hypothetical) lexicons to those of words in the actual English lexicon. Each novel lexicon will be generated on the basis of the phonotactic patterns of the actual lexicon. The modeled lexicon is PhLex, a 30 000-word phonologically transformable lexicon of American English [Seitz, Bernstein, and Auer (1995)].

4aSC12. Phonetic structure is similar across auditory and vibrotactile speech perception. Christopher T. Kello and Lynne E. Bernstein (Dept. of Commun. Neurosci., House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057, ckello@hei.org)

Vibrotactile vocoders have long been viewed as a possible method for conveying speech information to deaf perceivers. However, modest levels of speech transmission have led to numerous alternative notions concerning limitations of vibrotactile devices. This study investigated whether perceived acoustic phonetic structure is preserved in the transformation of the $F2$ speech region via a linear vibrotactile vocoder array. In experiment 1, naive hearing subjects discriminated pairs of vibrotactile word stimuli immediately before and after 2 h of practice. At both times, relative vibrotactile discrimination levels were well predicted by identification of phonemes presented by an analogous $F2$ acoustic vocoder, suggesting that structure is preserved. In experiment 2, subjects separately rated auditory and vibrotactile consonant similarity of stimuli presented by auditory and vibrotactile vocoders, respectively. Analysis of phonological feature information showed similarities and differences in perceptual structure across modalities. Vibrotactile stimuli were rated to be more similar to each other than were auditory stimuli, suggesting that phonetic information was degraded relative to auditory information. Based on these results, current investigations are directed at methods for inducing perceptual learning of vibrotactile phonetic information. [Work supported by NIH/NIDCD 00695.]

4aSC13. Sources of variability for talker and listener effects in auditory–visual speech perception. Su-Hyun Jin and Arlene E. Carney (Dept. of Commun. Disord., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455)

Carney *et al.* [J. Acoust. Soc. Am. **106**, 2270 (1999)] reported on talker and listener effects for auditory–visual speech perception for syllabic stimuli that were disparate [a visual /gi/ paired with an auditory /bi/ or a visual /bi/ paired with an auditory /gi/]. Listeners varied in the ratios of fused and combination responses (the McGurk effect) to auditory responses, depending upon the talker and their own most frequently observed mode of perception. The purpose of this experiment was to determine if listener and talker variability could be manipulated by recombining talkers' voices with different talkers' faces. Green *et al.* [Percept. Psychophys. **50**, 524–536] had demonstrated that the McGurk effect was observed even when voices and faces did not match in gender. The same 11 talkers (5 male and 6 female) were used from the Carney *et al.* experiment. Twenty-four listeners heard the combination of all male talkers with each other and all female talkers with each other. Similar patterns of listener and talker variability were observed even when voices and faces did not match, supporting the earlier conclusion that the McGurk effect is a graded phenomenon, depending upon listener and talker. [Work supported by NIDCD.]

4aSC14. The effects of audiovisual speech on sentential phoneme detection. Ethan A. Cox (Dept. of Psych., Univ. of Arizona, Tucson, AZ 85721, ecox@u.arizona.edu)

Many previous studies in audiovisual speech perception concentrate on situations where auditory information is degraded, or stimuli with conflicting auditory and visual information. However, relatively little work has addressed the issue of how the availability of visual information contributes to postphonetic levels of processing. Presented here are results from a series of studies targeting audiovisual processing in lexical and sentential structures. Subjects in a phoneme-monitoring task for target words in sentences showed significantly faster RTs for audiovisual conditions as compared to auditory-only conditions. Additionally, a lexical frequency effect was observed in the auditory-only, but not the audiovisual condition. These initial findings focused on the bilabial viseme group and were compatible with an interpretation based on the earlier temporal availability of the visual information as compared to the auditory. Subsequent experimentation with the same general procedures incorporated catch trials with nontarget bilabials, and indicates a more complex account. Comparison of the original findings with other viseme groups also supports the view that the availability of visual information contributes more than a simple temporally earlier trigger for target identification. Monitoring performance cannot be well accounted for by treating the auditory and visual information as separate channels for controlling the response.

Session 4aUW

Underwater Acoustics, Signal Processing in Acoustics and Acoustical Oceanography: Model-Based Processing of Sources in Motion I

Peter G. Cable, Cochair
BBN Technologies, Union Station, New London, Connecticut 06320

William M. Carey, Cochair
Aerospace and Mechanical Engineering Department, Boston University, 110 Cummington Street, Boston, Massachusetts 02215

Chair's Introduction—8:25

Invited Papers

8:30

4aUW1. Inference of source–receiver motion from phase measurements in the Modal Mapping Experiment. George V. Frisk (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

In March 1997, the Modal Mapping Experiment (MOMAX) was conducted aboard the R/V Endeavor in about 70 m of water off the New Jersey coast. Three drifting MOMAX buoys, each containing a hydrophone, GPS navigation, and radio telemetry received signals up to ranges of 10 km from a NUWC J15-3 source suspended from the moving ship and transmitting pure tones at 50, 75, 125, and 175 Hz. A striking feature of the data is the remarkable stability and regularity of the phase, even though the magnitude displays a complex multimodal interference pattern. This phase behavior occurs even at the higher frequencies and higher source/receiver speeds (up to 3 kts) measured in the experiment. A phase model is developed which indicates that the leading-order behavior of the time rate-of-change of the phase is simply equal to the product of a typical wave number in the water column and the source-receiver speed. This model accurately predicts the source-receiver speed from the phase measurements obtained in MOMAX. The implications of this model for other scenarios, such as long-range, deep-water situations are also discussed. [Work supported by ONR.]

8:50

4aUW2. Synthetic aperture processing to extract ocean bottom acoustic properties. Subramaniam D. Rajan (Science Solutions, Inc., 845 106th Ave. NE, Ste. 200, Bellevue, WA 98004)

A review of an approach to determine ocean bottom acoustic properties in a shallow-water environment from data acquired on a synthetic aperture horizontal array will be presented. The performance of this method will be demonstrated using data from a number of field experiments. The feasibility of using this approach for rapid environment assessment will also be discussed.

9:10

4aUW3. Model-based and data-based processing of moving sources in a shallow water channel. Lisa M. Zurk and James Ward (MIT Lincoln Lab., 77 Masachusetts Ave., Cambridge, MA 02139)

Adaptive MFP algorithms are applied to passively detect and localize a moving source while mitigating interference from surface shipping. A model-based motion compensation algorithm is applied to prevent smearing loss from target motion and extend the observation interval for adaptation. Moving interference is removed by applying a time-varying spatial filter at each data snapshot. The time-variation of the filter decreases the size of the instantaneous interference subspace and allows for a larger signal subspace. The interference subspace is formed by one of three methods. In the first method, knowledge of the ship position is combined with a propagation model to predict the time-varying acoustic interference. A sector null is formed to protect against mismatch degradation. In the second method, a data decomposition is used to determine the strong interferers. The third method combines model-based and data-based approaches. For each method, interference rejection is quantified by presenting SINR values for the candidate algorithm and comparing to conventional approaches. The results use data obtained from the Santa Barbara Channel Experiment. [This work was sponsored by DARPA under Air Force Contract F19628-95-C00022E. Opinions, interpretations, conclusions, and recommendations are those of the author and are not necessarily endorsed by the United States Air Force.]

9:30

4aUW4. The effects of source motion on the performance of matched field processors. Ahmad T. Abawi, Newell O. Booth, Phil Schey (SPAWAR Systems Ctr., 53560 Hull St., San Diego, CA 92152-5001, abawi@nosc.mil), and W. S. Hodgkiss (Scripps Inst. of Oceanogr.)

To obtain a reliable and accurate estimate of the cross-spectral matrix, it is necessary to average individual matrices over a number of snapshots. However, an upper limit in the number of snapshots that can be used to average the cross-spectral matrix is imposed by source and interferer motion. Averaging while the target moves through resolution cells introduces signal gain degradation, spreading

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the beam-former peak response through the resolutions cells occupied by the source. Averaging while interferences move smears sidelobes, increasing the number of eigenvalues in the cross-spectral matrix and reducing the effectiveness of adaptive noise cancellation processes. During the May 1996 Shallow Water evaluation cell Experiment (SwellEx-96), which was performed in shallow water over a relatively flat bottom off the coast of San Diego, data was obtained with towed sources and interfering surface ships moving at many different rates. Data were recorded on several arrays including two 118-m 64-phone vertical array, one tilted by 45 degrees. This paper presents an analysis of source and interferer motion effects using SwellEx-96 data. Signal gain degradation, target and side-lobe smearing, and adaptive noise cancellation effectiveness are examined as a function of integration time, and number of elements used in the processing.

9:50

4aUW5. A hybrid plane-wave beamforming/matched-field processing, track-before-detect technique for detection of broadband moving sources with horizontal line arrays. Paul A. Baxley and Randall Brannan (SPAWAR Systems Ctr. San Diego, 53560 Hull St., San Diego, CA 92152-5001, baxley@spawar.navy.mil)

A hybrid plane-wave beamforming/matched-field processing (MFP), track-before-detect (TBD) technique for the automated detection and tracking of broadband moving sources in shallow water is presented. The approach, applicable to bottom-mounted horizontal line arrays (HLAs), provides estimates of the full track parameters (range, depth, course, and speed) with a processing load substantially lower than that required for a fully three-dimensional MFP search. The method begins with the production of a bearing-time record (BTR) for the HLA. By applying a TBD algorithm to the BTR, a detection is made and track bearing as a function of time is estimated with some bias and uncertainty. The MFP range-depth ambiguity surfaces are then produced for an envelope of bearings bracketing the bearing estimate uncertainty at each time. The TBD processing of these surfaces, involving a search over tracks defined by all combinations of bearings in the envelopes at the beginning and end of candidate tracks, allows for an estimation of the full track parameters as well as a reduction of bearing uncertainty and bias. Results of the application of this technique to broadband source-tow data in the Black Sea are presented. [Work supported by the ONR 321US.]

10:10–10:30 Break

10:30

4aUW6. Broadband shallow-water sea-test results with a towed vertical aperture array. John P. Ianniello and John M. Tattersall (Naval Underwater Systems Ctr., 1176 Howell St., Newport, RI 02841)

We review matched-field processing (MFP) sea test results obtained using a towed vertical aperture array. The array consisted of five horizontal lines towed in a vertical plane, nominally one over the other, with an approximately 5-m separation between the lines. We review results from two tests, one conducted in the Gulf of Mexico in about 120 m of water, and the other conducted southwest of Key West, FL in about 600 m of water. The vertical aperture array was towed on straight-line courses, at various ranges, parallel to a towed sound source. The radiated signal was a Gaussian pseudorandom waveform in the 100–600-Hz band. Results using replica correlation techniques are first used to study the channel transfer function; next, MFP results obtained by treating the signal as a random process are shown. These results demonstrate accurate range and depth localizations from 5 to 20 km as well as the ability to resolve a loud surface interferer from a quieter source at depth which is on the same bearing. These test results demonstrate the inherent robustness of broadband MFP. [Work supported by ONR 321.]

10:50

4aUW7. Model-based towed array processor: Experimental results. Edmund J. Sullivan (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841), Dieter Brecht (Federal Armed Forces Underwater Acoust. and Marine Geophys. Res. Inst., Germany), and Leif Persson (Natl. Defence Res. Establishment, Stockholm, Sweden)

Results of a towed array bearing estimation experiment using a model-based processor are presented. The experiment took place in the southern Baltic using an Atlas towed array. The towship was the Swedish corvette GAVLE. Results for the case of a 121-Hz source are presented. The towship ran in a straight line, past the source, where the distance of closest approach was 500 m. A short overview of the model-based approach, which is based on a state space formulation using a Kalman estimator, will be presented. By formulating the problem in state space form, it is possible to include sophisticated models in the processor. This, in effect, is a self-consistent means of including *a priori* information into the processing. It will be shown that the inclusion of the array's forward motion provides significant improvement in performance over the conventional processor in terms of the variance of the estimate. The improvement in performance arises from the bearing information contained in the forward motion of the array. This information obtains, from the time scale, change in the signal or, in the narrrow-band case, the Doppler. The experimental results clearly show the improvement over the conventional processor.

11:10

4aUW8. Imaging a moving target using forward-look synthetic aperture sonar. Geoffrey S. Edelson (Sanders, A Lockheed Martin Co., Adv. Systems & Technol., P.O. Box 868, Nashua, NH 03061-0868, geoffrey.s.edelson@lmco.com), Charles J. Gedney, Philip A. Abbot, Ira Dyer (OASIS, Inc., Lexington, MA 02173), and Kenneth D. Rolt (Sonetech Corp., Bedford, NH 03110)

Mid-frequency active, surface ship sonar data from a recent shallow-water sea trial have been processed using synthetic aperture sonar (SAS) algorithms. The target of opportunity closed head-on directly toward the surface ship at a relative speed of 13 knots through an acoustic environment that was characterized by a downward refracting sound speed profile in 100-m-depth water over a sand/mud bottom during calm seas. This situation represents an endfire (90-degree squint angle) scenario for which the resolution of any line array (synthetic or real) is considerably degraded relative to the broadside case. The test was not designed for SAS application so the aperture is significantly undersampled. The processing approach utilizes signal adaptive, ping-to-ping correlation autofocusing

to correct for the effects of platform and target motion, and for ocean instabilities. The processing is tuned to the specific environment by setting the cross-correlation window size based on the two-way group arrival structure of the channel. The aliasing that results from the undersampled aperture is reduced to acceptable levels through the appropriate use of cross-range matched field processing. Images formed by processing the data from a forward-look synthetic aperture measuring 2200 wavelengths with a mean target distance of 9.5 km are presented.

Contributed Paper

11:30

4aUW9. Array measurements and characterizations with sources in motion. William M. Carey (Dept. of Aerosp. and Mech. Eng., College of Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, wcarey@bu.edu)

Traditional processing has treated space and time as independent random variables with assumed ergodicity; however, dynamic-noise fields are produced by moving sources. In these cases, can velocity and range information be used to increase gain, cancel noise, and improve resolution? High-resolution arrays have measured the dynamic ambient noise fields at low frequencies, over long time intervals and at basin/margin spatial scales, and have resolved/tracked shipping. This paper summarizes experiments with arrays which used source-relative velocity as a processing variable, that is, space–time processing. Results from a closed-basin shipping dominated noise field showed the relative velocity acted as a filter that improved the signal-to-noise ratio for sources with known relative velocity and reduced it for sources having different relative velocities. Results from a shallow water experiment with a vertical array, moving source, and the Bessel transform were shown to determine the wave number spectrum and were comparable to the modeled spectrum in both amplitude and phase. Thus the contention that improved array ship-noise measurement and cancellation is possible if the velocity is known may have merit, especially if modal variations can be adaptively modeled.

FRIDAY AFTERNOON, 2 JUNE 2000 — AMERICAN ROOM, 2:30 TO 4:50 P.M.

Session 4pAA

Architectural Acoustics: Computer Modeling and Measurement Techniques for Large Room Acoustics

Paul T. Calamia, Chair
Kirkegaard & Associates, 801 West Adams Street, 8th Floor, Chicago, Illinois 60607

Chair's Introduction—2:30

Contributed Papers

2:35

4pAA1. An echoic room model for acoustic simulations by non-reflection boundaries using the Bergeron method. Hidemaro Shimoda (Inst. of Technol., Shimizu Corp., 3-4-17 Etchujima Koto-ku, Tokyo 135, Japan, shimoda@sit.shimz.co.jp)

An analytical model of anechoic rooms by distributed equivalent circuits is constructed in the computer using the Bergeron method [Shimoda *et al.*, ''Analysis of sound fields in rooms using Bergeron's method,'' Trans. I.E.C.E. Japan, 72 A, 1, 1–11 (1989); English transl., 1989 Electronics and Communications in Japan, Part 3, Vol. 72(12), Scripta Technica, Inc. (Wiley, New York, 1990)]. In this analytical method, the non-reflection boundaries are modeled by gradually increasing sound transmission losses in specific regions near boundaries just like a sand beach exposing sea waves. Also, the sound transmission characteristics or the sound pressure distributions in the room are investigated. The model can be applied to the sound scattering analysis by an arbitrary object.

2:50

4pAA2. From auditorium ray acoustics to wave acoustics: The next giant step. Herman Medwin (OCEANAC Assoc., 4021 Sunridge Rd., Pebble Beach, CA 93953)

For at least five decades, architectural acousticians have used straight-line, infinite-frequency rays to determine which regions of an auditorium are insonified by a sound source and which are shadowed. In the next decade, leading acoustical designers of auditoriums will turn to predictions of the timing and amplitude of the different frequencies which partially reflect and bend around real auditorium components such as proscenium arches and boxes. The frequency dependence of wave reflection from rigid, finite surfaces has been developed by Clay *et al.* [J. Acoust. Soc. Am. **94**, 2279 (1993)]. The calculation of the frequency dependence of the diffraction component, which some have called the BTM technique, is a digitally formulated, finite-wedge, frequency-spectral interpretation of the classic Biot–Tolstoy (1957) theory for impulse scatter from an infinite wedge. The BTM technique, which we described first at an ASA meeting in 1978, has been applied to shadowing by highway noise barriers [Medwin, J. Acoust. Soc. Am. **69**, 1060–1064 (1981)] and to multifrequency reverberation from randomly rough, ocean surfaces [Keiffer and Novarini, in *Computational Acoustics*, edited by D. Lee *et al.* (1990), Vol. 1, pp. 67–81]. The procedure is explained in detail in Medwin and Clay [*Fundamentals of Acoustical Oceanography* (Academic, New York, 1998)]. The justification and analytical extension of this method have recently been formulated by Svensson [J. Acoust. Soc. Am. **106**, 2331–2344 (1999)], who has made the program available on his web site and has suggested applications to auditoriums. The physical interpretation of the BTM technique will be described in terms of the spectral and temporal behavior of finite plates and wedges as modules of more complex components of auditorium surfaces.

3:05

4pAA3. Subjective relevance of objective measures for spatial impression. Lily M. Wang and Anders C. Gade (Dept. of Acoust. Technol., Tech. Univ. of Denmark, Bldg. 352, DK-2800, Lyngby, Denmark)

Several objective measures have been proposed to describe the feeling of spatial impression in concert halls, including Lateral Energy Fraction (LF) and Interaural Cross-Correlation Coefficient (IACC). However, previous studies have shown that LF and IACC values did not highly correlate with each other at individual seat positions in real halls [J. S. Bradley, J. Acoust. Soc. Am. **96**, 3525–3535 (1994)]. To investigate the listener

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