Session 2aAB

Animal Bioacoustics, Speech Communication and Signal Processing in Acoustics: Signal Processing Techniques for Animal Bioacoustics

Larry L. Pater, Chair
USACERL, 2902 Farber Drive, Champaign, Illinois 61821

Chair's Introduction—7:55

Invited Papers

8:00

2aAB1. Applications of a high-quality sound manipulation algorithm STRAIGHT for animal voices. Hideki Kawahara (Dept. of Design Information Sci. Faculty of Systems Eng., Wakayama Univ./ATR/CREST, 930 Sakaedani, Wakayama, 640-8510 Japan)

Applications of a high-quality speech analysis/modification/synthesis algorithm [Kawahara *et al.*, Speech Commun. **27**, 187–207] for animal voices are discussed. The proposed algorithm consists of (1) $F0$ adaptive spectral smoothing to eliminate periodic interferences on a time-frequency representation, (2) $F0$ and source information extractor based on a fixed point analysis of mapping from frequency to instantaneous frequency, and (3) group delay manipulation to control temporal structure of an excitation source. Each component has to be tuned to specifications of animal voices to take full advantage of the method. There are two types of applications: The first method is converting animal voices to sound more like human speech: This modification tries to enhance the researcher's auditory inspection of animal voices by mapping their voice physical parameters onto a region which is ecologically valid for humans. The second method is manipulating specific parameters of an animal voice to test function/contribution of each physical parameter: This modification tries to keep the naturalness of the stimuli while keeping a high degree of control accuracy of specific physical parameters of stimuli. This strategy would enable measurements of behavior responses in the vicinity of natural stimulus conditions.

8:30

2aAB2. Wavelet transforms for bioacoustic signal processing. Leon H. Sibul (Penn State Univ., Appl. Res. Lab. and Acoust. Prog., University Park, PA 16802, lhs2@psu.edu)

Wavelet transforms (WT) are natural tools for analysis of bioacoustic signals. In analysis of transient signals and nonstationary stochastic processes it is important to know not only what are the frequency content transients but when did these transient signals occur. The importance of representation of signals on the time-frequency plane, not only on frequency or time axes, is now widely accepted. Linear time-frequency analysis of transient signals (signals with time-varying frequency content) can be based on short-time Fourier transforms (STFT) and generalized Gabor transforms (GT). Wavelet transform analysis is analogous to STFT and GT analysis, except WTs represent signals on time and scale plane. In time-frequency analysis there is a tradeoff between time and frequency resolution. WTs have the advantage that they have constant time-frequency resolution over a wide frequency range. WT represent constant fractional or constant Q frequency analysis, thus efficiently analyzing a wideband of frequencies. Continuous wavelet transforms are linear transforms that can be inverted. In this tutorial paper basic properties of continuous and discrete wavelet transforms are reviewed, basic issues between Fourier transform techniques are discussed and disadvantages and advantages of WT are pointed out. [Work supported by ONR, Code 333.]

9:00

2aAB3. Time-frequency analysis: A tutorial review. Patrick J. Loughlin (Dept. of Elec. Eng., Univ. of Pittsburgh, Pittsburgh, PA 15261)

Many signals, such as speech and other animal sounds, FM radio waves, machine vibrations, and sonar and radar echoes, exhibit frequency characteristics that change over time. Standard spectral analysis of such signals provides an incomplete description of the process, because the spectral density reveals only what frequencies existed in the signal, but not when they occurred. Time-frequency analysis, however, shows how the frequencies change over time and hence is a more complete characterization of these signals. This approach to time-varying spectral analysis has become standard and has revealed new physical properties of signals. Perhaps the most common method of time-frequency analysis is the spectrogram, developed over 50 years ago. Many other methods have since been developed, driven largely by a desire to overcome limitations of the spectrogram. Most of these methods were developed by effective use of Cohen's 1966 formulation for generating time-frequency distributions (TFDs). A review of the spectrogram and these TFDs, including the Wigner, Choi–Williams, and Zhao–Atlas–Marks distributions, among others, is presented. A variety of applications are shown that illustrate the basic ideas and the different methods. [Work supported by ONR grant N00014-98-1-0680.]

9:30

2aAB4. Speech recognition meets bird song: A comparison of statistics-based and template-based techniques. Sven E. Anderson (Dept. of Computer Sci., Univ. of North Dakota, Grand Forks, ND 58202-9015, anderson@cs.und.edu)

Pattern recognition technology that has been developed for recognizing units of human speech can often be adapted for both recognition and analysis of animal vocalizations. This paper discusses two types of speech recognition algorithms, template based and statistics based, with respect to their ease of deployment and potential application to the objective, quantitative analysis of animal vocalizations. Implementations of the two types of algorithms have been compared using a large database of song units recorded from two song bird species. The algorithms exhibit different strengths and weaknesses. The template-based dynamic time-warping algorithm provides quantitative sound comparisons that are directly useful to a researcher, but selection of training materials depends on expert knowledge. The statistics-based hidden Markov model algorithm requires more training data, but usually performs better in noisy environments and with more variable vocalizations. While both algorithms are accurate in restricted domains, recognition performance could be improved if it were based on species-specific features extracted from the acoustic input. [Work supported by NIH 1-F32-MH10525 and ARO DACA88-95-C-0016.]

10:00

2aAB5. Application of current speech recognition technology to nonstandard domains. Diane Kewley-Port (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405)

The typical input to automatic speech recognition (ASR) algorithms is a word-length or longer acoustic waveform. The typical output consists of names of items identified from stored vocabularies. Most algorithms also use scoring procedures to find the top matches between the input and the stored information. These evaluation scores, probabilities or distance metrics, may also be output. There are many nonstandard applications of ASR technology that combine identification and evaluation scores. Examples to be discussed include speech training to improve intelligibility, spoken language proficiency of non-native speakers, and language training for adults with developmental disabilities. Most current ASR algorithms are fine tuned for species-specific properties of human speech and language. In addition, many incorporate psychophysical properties of the human auditory system in the initial signal processing. However, the underlying pattern recognition algorithms are also applicable to a wide range of animal vocalizations. Some commercial and laboratory systems will be discussed in relation to nonstandard applications of ASR. [Work supported by NIH, RO1 DC-02229, R44 DC-02213, and R43 HD-35425.]

10:30–10:45 Break

Contributed Papers

10:45

2aAB6. Technique for the generation and frequency compensation of bandlimited white noise and its application in studies of masked hearing thresholds. James J. Finneran, Donald A. Carder, Sam H. Ridgway (SPAWAR Systems Ctr. San Diego PL-BS, Div. D35, 49620 Beluga Rd., San Diego, CA 92152-6266, finneran@spawar.navy.mil), and Carolyn E. Schlundt (Sci. Applications Intl. Corp., San Diego, CA 92110)

Masking noise is often used in hearing tests to create a floor effect in the presence of ambient noise or to examine specific features of the auditory system (e.g., the critical bandwidth). One of the chief requirements of the masking noise is that it possess a flat frequency spectrum within some user-defined bandwidth. Generation of suitable masking noise is complicated by the frequency response of the sound projector, which may possess a frequency-dependent transmitting sensitivity and/or exhibit resonances within the desired frequency range. At low frequencies acoustic standing waves may also alter the noise frequency spectrum. To overcome these limitations, a technique has been developed to generate bandlimited noise whose frequency content is compensated in order to flatten peaks or valleys in the measured frequency spectrum. Compensation is performed by passing white noise through a digital filter whose coefficients are determined from previous measurements of the acoustic system frequency response. The system has been implemented using a personal computer with commercial hardware and custom software. The method has been used to quickly generate bandlimited Gaussian and uniform white noise for studies of masked underwater hearing thresholds in marine mammals. [Work supported by ONR and the NRC Research Associateship Program.]

11:00

2aAB7. Improved signal processing techniques for measurement of the inter-pulse interval (IPI) of sperm whale clicks. Thomas J. Hayward (Naval Res. Lab., Washington, DC 20375) and G. Pavan (Università degli Studi di Pavia, Via Taramelli 24, 27100 Pavia, Italy)

Improved signal processing techniques are described for estimating the inter-pulse interval (IPI) associated with the multiple pulses in a sperm whale click. These pulses are believed to result from repeated reflections of the original pulse within the spermaceti organ in the whale's head. Previous estimates of the IPI have been based on time series correlation or on computation of the cepstrum of the received waveform [J. Goold, J. Acoust. Soc. Am. **100**, 3431–3441 (1997)]. The present work describes real-time computation and display of the spectrogram and cepstrogram (time-cepstral distribution) [G. Pavan, Proc. IoA **19** (1997)] and examines the benefits of the simultaneous display of this information for field analysis and for browsing long recordings to identify sperm whale clicks. In addition, a detailed model of the spectral modulation associated with the multiple pulses is developed and leads to improved estimation of the IPI. Examples are presented that indicate the method may be more robust with

respect to additive noise and the effects of surface reflection than the cepstrum-based approaches. [Work of the first author supported by ONR. Work of the second author supported by the University of Pavia, Italy, the Italian Ministry of the Environment, and by the Italian Navy.]

11:15

2aAB8. Testing the acoustic adaptation hypothesis for eastern and spotted towhee song. Hitesh Khanna (Dept. of EEOB, Ohio State Univ., Columbus, OH 43210)

The acoustic adaptation hypothesis predicts that signals adapted for long-distance communication in forested habitats should possess low rates of amplitude modulation (slow trills), whereas signals adapted to open habitats should possess high rates of amplitude modulation (fast trills). The songs of two songbirds, the eastern and spotted towhees, were used to test this hypothesis. These two species have only recently been recognized as distinct species. Differences in song are one of the reasons given for splitting the species. Eastern towhees occupy edges of deciduous forests and sing slow trills. Spotted towhees occupying open chaparral habitat sing fast trills. Songs of both species were broadcast in their native habitats, and recorded at several distances from the sound source. Spectrogram cross-correlation was used to quantify sound degradation with distance. The results support the predictions of the acoustic adaptation hypothesis. The slow eastern towhee trill degrades less than the fast spotted towhee trill in a forested habitat, whereas the fast spotted towhee trill degrades less than the slow eastern towhee trill in an open habitat.

11:30

2aAB9. Non-Gaussian randomness outside and inside the bat's brain. Rolf Müller and Roman Kuc (Intelligent Sensors Lab, Elec. Eng. Dept., Yale Univ., New Haven, CT 06520-8284, rolf.mueller@yale.edu)

Reflector placement in the habitats of bats is pronouncedly random. Furthermore, the impulsive nature of reflection sequences generated by facets found, e.g., in foliage can endow the probability density function of the received echo amplitudes with heavy tails. Two useful features for classification of foliages, which may be utilized by bats as ubiquitous landmarks, are the extent to which the amplitude distribution deviates from Gaussian (its ''peakiness'') and an eventual nonstationary gain of the propagation channel. The signal processing in the bat's auditory system does not have direct access to the target's reflection sequence, however. Even the shortest echolocation pulses seen in bats are chirps with durations, which are nonnegligible compared to the duration of the reflection sequence and the temporal spacing of salient peaks inside of it. On reception, echoes are subjected to bandpass filtering and demodulation. The latter step prohibits recovery of the reflection sequence by means of pulse compression using a matched filter. Consequently, it is investigated how the demodulated filter bank output may be used directly as a substrate for target classification. Specifically, the possible role of encoding the waveforms in sparse spike trains and comparing binaural information is evaluated.

TUESDAY MORNING, 2 NOVEMBER 1999 MCKINLEY ROOM, 7:55 A.M. TO 12:00 NOON

Session 2aAO

Acoustical Oceanography: Geoacoustic Propagation and Inversion

Kevin D. Heaney, Chair
Science Applications International Corporation, 888 Prospect Street, Suite 201, La Jolla, California 92037

Chair's Introduction—7:55

Contributed Papers

8:00

2aAO1. The perfectly matched layer for elastic waves in poroelastic media. Yanqing Zeng (Dept. of Civil and Environ. Eng., Duke Univ., Durham, NC 27708, yz3@acpub.duke.edu), Jiangqi He, and Qinghuo Liu (Duke Univ., Durham, NC 27708)

The perfectly matched layer (PML) as a material absorbing boundary condition (ABC) was first introduced by Berenger for electromagnetic waves, and later developed by Chew and Liu for elastic waves. In the continuous limit, an interface between a regular medium and a fictitious, lossy PML medium can be made perfectly matched so that there is no reflection from the PML to the regular medium. This property is independent of the incidence angle and the frequency of the incoming waves. Consequently, the PML provides an ideal ABC for the truncation of the computational domain in numerical methods such as the finite-difference, finite-element, and pseudospectral time-domain methods. Numerical experiments show that this ABC can reduce the reflection to several orders of magnitude below the level of the previous ABCs. In this work, the PML is further extended to elastic waves in poroelastic media through the approach of complex coordinates for Biot's equations. This nonphysical material is used as an ABC at the computational edge of a finite-difference algorithm to truncate unbounded media. Numerical results show that the PML ABC attenuates the outgoing waves effectively.

8:15

2aAO2. Frame bulk and shear moduli of air and water saturated glass beads. Masao Kimura (Dept. of Ocean Eng., Tokai Univ., 3-20-1 Orido, Shimizu, Shizuoka, 424-8610 Japan)

Are the values of the frame bulk and shear moduli of air saturated sediments, and those of water saturated sediments, the same or not the same? This problem is currently under debate. In this study, the frame bulk and shear moduli of air and water saturated glass beads with five different grain sizes are derived from the measured values of the longitudinal and shear wave velocities in these media. The grain sizes are 0.05, 0.1, 0.2, 0.4 and 0.8 mm. The longitudinal wave velocities are measured using piezoelectirc transducers with the operating frequency of 11.8 kHz for dry sample, and 500 kHz for wet sample. The shear wave velocities are measured using bimorph type piezoelectric transducers with the operating

frequency of 3.5 kHz for both samples. The relationship between the values of these moduli in air and water saturated glass beads and the effect of grain size are investigated.

8:30

2aAO3. Geoacoustic and signal gain modeling in a range-dependent, shallow-water environment. Ilya Rozenfeld (Rensselaer Polytechnic Inst., Troy, NY 12180, rozeni@rpi.edu), William M. Carey (79 Whippoorwill Rd., Old Lyme, CT 06371), Peter G. Cable (BBN Systems and Technologies, New London, CT 06320), and William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180)

The Acoustic Characterization Test III (ACTIII) was conducted in the Strait of Korea in 1995. One of the goals of the experiment was to determine limitations on spatial signal processing due to environmental variabilities. Measurements of transmission loss and array signal gain (ASG) provide quantitative estimates of fluctuations in the acoustic field. Analysis of observations is supported by using grab samples, historical cores, and seismic measurements to develop a geoacoustic model. The geoacoustic model along with sampled bathymetry and sound-speed profiles is used to generate simulations of an effective attenuation coefficient, which measures the rate of change of mean transmission loss with range. The introduction of nonlinear frequency dependence into sediment attenuation profiles produces good agreement between the simulations and measured data. The ASG is directly related to the spatial coherence function. By employing procedures we have developed previously for approximating the coherence function, we use the environmental model to compute ASG. This allows further comparisons with experimental data as well as a determination of validity limits for the procedures. [Work supported by ONR.]

8:45

2aAO4. Newly observed seismoacoustic wave characteristics in disturbed water-covered sand. Jacques R. Chamuel (Sonoquest Adv. Ultrason. Res., P.O. Box 81153, Wellesley Hills, MA 02481-0001)

Surprising new experimental results are presented revealing the characteristics of broadband transient seismoacoustic waves in water-covered naturally deposited sand and disturbed sand. In the naturally deposited sand, one compressional wave ($\sim$1650 m/s) and one shear wave ($\sim$140 m/s) were detected as expected. In the disturbed sand, the high-frequency ($\sim$100 kHz) fast compressional wave was highly attenuated, and a slow low-frequency ($\sim$20 kHz) compressional wave component was observed. In certain models, the slow low-frequency compressional wave component co-existed with the fast high-frequency compressional wave. The slow compressional wave velocity varied between 285 m/s and 210 m/s. The shear wave velocity and amplitude remained practically unchanged. Results from coarse and fine sand are compared. The findings are important for understanding the penetration and conversion of high-frequency acoustic waves in sandy seafloor critical for the detection of buried objects in littoral regions. [Work supported by ONR.]

9:00

2aAO5. The dispersion and attenuation of body waves in a transversely isotropic porous medium including the effect of pore shape. Yinbin Liu (Avadh Bhatia Phys. Lab., Dept. of Phys., Univ. of Alberta, Edmonton, AB T6G 2J1, Canada)

Based on Biot's theory of a general anisotropic porous medium [J. Appl. Phys. **33**, 1482–1498 (1962)] and the dynamic permeability formula given by Johnson *et al.* for an isotropic medium [J. Fluid Mech. **176**, 379–402 (1987)], we derived analytic expressions for the velocities and attenuations as functions of frequency for the quasi-P1 wave, quasi-P2 wave, quasi-SV wave, and SH wave in a transversely isotropic porous medium. Numerical results are given for one set of material parameters and compared with Schmitt's results [J. Acoust. Soc. Am. **86**, 2397–2421 (1989)] based on an alternative theoretical approach. The calculated results show that the phase velocities for quasi-P1 wave, quasi-SV wave, and SH wave are not sensitive to the pore shape, and the attenuations for four kinds of waves and the phase velocity for quasi-P2 wave are sensitive to the pore shape in the region of near and above threshold frequency.

9:15

2aAO6. Wave number sampling at short range. Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

The self-starter is an efficient approach for solving geoacoustic inverse problems involving a vertical array of receivers located on the order of ten wavelengths from a source [R. J. Cederberg and M. D. Collins, J. Acoust. Soc. Am. **22**, 102–109 (1997)]. With this approach, accurate solutions can be obtained by sampling the wave number spectrum at approximately ten points. There have been unsubstantiated claims that the spectral solution provides similar efficiency when implemented with an approach that involves perturbing the integration contour off the real line [F. B. Jensen *et al.*, *Computational Ocean Acoustics* (American Institute of Physics, New York, 1994), pp. 231–240]. It is demonstrated that this quadrature scheme breaks down at short ranges and that the self-starter solution is about an order of magnitude faster. These conclusions are based on comparisons of the rational approximations associated with the self-starter and the quadrature scheme, comparisons of acoustic fields for particular problems, and a simple analysis of the spectral integral. These conclusions are not surprising since the rational approximation associated with the self-starter is based on the analytic evaluation of the spectral integral while the other rational approximation is based on numerical integration. [Work supported by ONR.]

9:30

2aAO7. Geoacoustic inversion techniques applied to field data. Joseph F. Lingevitch, Michael D. Collins, Dalcio K. Dacol, Michael Nicholas, and John S. Perkins (Naval Res. Lab., Washington, DC 20375)

Reliable and efficient techniques for solving geoacoustic inverse problems have been developed. A coordinate rotation technique can be used to identify the best resolved combinations of parameters for a given frequency and configuration of hardware [M. D. Collins and L. Fishman, J. Acoust. Soc. Am. **98**, 1637–1644 (1995)]. Problems involving a vertical array of receivers at short range can be solved in a few seconds with an implementation of the self starter that is based on the method of undetermined coefficients [D. K. Dacol and M. D. Collins, J. Acoust. Soc. Am. (accepted for publication)]. These techniques are applied to data that were obtained in the Straits of Florida during the KWIX '98 experiment. [Work supported by ONR.]

9:45

2aAO8. Estimation of anisotropic sediment parameters. Andrew J. Fredricks, William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180), Michael D. Collins, and Joseph F. Lingevitch (Naval Res. Lab., Washington, DC 20375)

Parabolic equation techniques have recently been developed for solving geoacoustic inverse problems [R. J. Cederberg and M. D. Collins, IEEE J. Ocean Eng. **22**, 102–109 (1997)] and for handling propagating problems involving anisotropic elastic layers [A. J. Fredricks, J. Acoust. Soc. Am. **101**, 3182 (1997)]. These techniques are used to investigate inverse problems involving anisotropic elastic layers. Complications arise from the direction dependence of the two sediment sound speeds. For the particular case of transverse isotropy, both sound speeds need to be estimated in two different directions. A coordinate rotation technique [M. D. Collins and L. Fishman, J. Acoust. Soc. Am. **98**, 1637–1644 (1995)] will be employed to estimate the resolvability of the direction-dependent wave speeds for different experimental configurations. [Work supported by ONR.]

10:00–10:15 Break

10:15

2aAO9. A cross-relation matched field inversion for geoacoustic parameter estimation in shallow water. Reza M. Dizaji (Dept. of Elec. and Computer Eng., P.O. Box 3055, Univ. of Victoria, Victoria, BC V8W 3P6, Canada), N. Ross Chapman, and R. Lynn Kirlin (Univ. of Victoria, Victoria, BC V8W 3P6, Canada)

In this paper the application of a matched field (MF) processor for geoacoustic parameter estimation from a signal of opportunity such as the broadband random noise radiated by passing ships is considered. A novel cross-relation (CR) based matched field processor is introduced for the purpose of ocean tomography based on broadband random source. This class of MF processors is based on the cross-relation property of sensor outputs at an array and their corresponding transfer functions from the true source location to the array. The processors are developed for nonstationary (NS), and wide sense stationary (WSS) random signals. For each formulation, two processors are proposed, a self-CR and a cross-CR. The performance of the proposed MF processors for environmental parameter estimation is demonstrated for real ocean environments using data collected by a 16-element vertical line array during one of the experimental tracks from the Pacific Shelf experiment that was carried out in shallow water off the west coast of Vancouver Island in the Northeast Pacific Ocean. The high-resolution property makes the cross-CR processor an excellent candidate for inversion for model parameters of the ocean waveguide. The replica or modeled fields are calculated using the normal mode model, ORCA.

10:30

2aAO10. Geoacoustic inversions in a very shallow water environment. K. M. McArthur and W. S. Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA 92093-0701)

The genetic algorithm is used to invert the geoacoustic parameters of a near shore, very shallow water region (from 10-m to 20-m water depth). The experimental data are source tows in the frequency range from 70 Hz to 700 Hz recorded on a horizontal line array of a 120-m aperture. The source tows were conducted offshore of Camp Pendleton, north of Oceanside, CA, along range-independent and range-dependent bathymetry tracks. Generally, a thin layer of sediment overlies a harder subbottom in this region. An echosounding survey, a sediment coring survey, and regional geologic information are used to develop background geoacoustic models. Inversion results from transmission loss versus range data agree well with the results obtained from inverting complex-pressure data across the array for a single range.

10:45

2aAO11. Results of environment inversion using modes extracted from vertical line array data. Tracianne B. Neilsen and Evan K. Westwood (Appl. Res. Lab., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029, neilsen@arlut.utexas.edu)

The results are presented for geoacoustic inversion using the depth-dependent normal modes extracted from vertical line array (VLA) data. The normal modes for multiple frequencies are obtained by performing a singular value decomposition (SVD) on VLA data recorded as a source moves outward in range. The data-extracted mode functions are then used to invert for the environmental parameters. The Levenberg–Marquardt nonlinear optimization method is used to adjust the environmental parameters to find the best fit between the data-extracted modes and modes modeled by the ORCA normal mode model. The method is useful for determining the sound speed profile and the characteristics of the uppermost bottom layers [T. B. Neilsen and E. K. Westwood, J. Acoust. Soc. Am. **104**, 1741 (1998)]. Mode extraction and geoacoustic inversion results for a range-independent track during the ACTII experiment will be presented. Extension of the theory for mode extraction (and preliminary results for inversion using simulated data) in a range-dependent environment using the adiabatic approximation will also be presented. [Work supported by ONR.]

11:00

2aAO12. Sediment tomography in the Middle Atlantic Bight. Gopu R. Potty and James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882)

Sediment properties such as compressional speed and attenuation have been extracted using the modal dispersion of explosive signals in the Middle Atlantic Bight. Data collected during the Shelf Break PRIMER experiment are used to generate sediment sound speed profiles along various propagation paths. Compressional speeds are inverted using traveltime of modes 1 to 9 from 10 Hz to 225 Hz. Compressional attenuation is inverted using modal amplitude ratios. Gravity cores taken at the experimental site give direct mesurements of compressional speed and attenuation for comparison. *A posteriori* estimates of resolution are evaluated. Nonlinear inversion techniques based on a Genetic Algorithm and Levenberg–Marquardt methods are used. Methods for increasing the efficiency of the inversion algorithm are also investigated. [Work supported by ONR.]

11:15

2aAO13. Geoacoustic parameter estimation using a back-wave propagation technique. Reza M. Dizaji (Dept. of Electron. and Computer Eng., P.O. Box 3055, Univ. of Victoria, Victoria, BC V8W 3P6, Canada), N. Ross Chapman, and R. Lynn Kirlin (Univ. of Victoria, Victoria, BC V8W 3P6, Canada)

The concept of back-wave propagation is developed as an inversion method to estimate ocean geoacoustic parameters where the source location is known. A phase-regulated technique is introduced to increase the sensitivity of the method for low-sensitive geoacoustic model parameters. In this procedure, a sensitivity factor is varied to enhance the phase changes due to model and environmental mismatch. It is shown that the spatial resolution of signal-energy concentrated at the true source location is increased when the sensitivity factor increases. This leads us to define a criterion based on spatial variance of signal energy around the true source location. This technique is applied to the real data from the Pacific Shelf experiment that was carried out in the shallow water off the West Coast of Vancouver Island in the Northeast Pacific Ocean. The inversion for estimating three ocean parameters including water depth, compressional speed of the first sediment layer, and sediment density is demonstrated. A way to reduce the three-dimensional search to three one-dimensional searches is proposed by exploiting the fact that the pressure field has different sensitivity with respect to these parameters. The replica or modeled fields used in this section are calculated using ORCA.

11:30

2aAO14. Use of surfseisms for determining near-surf-zone properties. Gerald L. D'Spain, K. Megan McArthur, Grant B. Deane, and W. Kendall Melville (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA 92093-0704)

Geophone array recordings of ''surfseisms,'' i.e., seismic interface waves created by breaking waves in the surf zone, show that at times, a breaking wave creates a pair of arrivals. The first arrival has peak spectral levels between 10 and 15 Hz and rolls off sharply above this peak, and the second surfseism arrival occurs 5–15 s later with broader-band energy concentrated above 20 Hz. The first arrival is created offshore when wave breaking has a plunging character, rather than spilling, whereas the second is present during both breaking types and apparently signifies the final dissipation of the turbulent wave bore on the beach face. The time interval between arrivals measures the distance offshore that plunging occurs, their relative amplitudes is an indicator of the type of wave breaking, and their absolute amplitudes measure the breaking wave forces interacting with the ocean bottom. In addition, the frequency of the first arrival's spectral peak

is a sensitive function of the thickness of the unconsolidated sand layer overlying wave-cut terrace material, indicating that surfseisms are naturally occurring probes of sand layer thickness. [Work supported by ONR, Code 32.]

11:45

2aAO15. Estimation of bottom parameters in shallow water using interference pattern in frequency domain. Boris Katsnelson, Valery Grigoryev (Phys. Dept., Voronezh Univ., Universitetskaya sq.1, Voronezh, 394693 Russia, katz@mph.vsu.ru), and Valery Petnikov (General Phys. Inst., Vavilova str.38, Moscow, 117333 Russia, petniko@az.gpi.ru)

A method of data inversion for geoacoustic parameters of bottom using FM acoustic signals is presented. The experiment was conducted at the stationary track between two ships. The source radiated linearly modulated signals within the band 25–95 Hz. Frequency spectra of received signals are taken as input data for application of the Matched Field Tomography (MFT) method. For estimation of bottom parameters we compare the experimental and model interference patterns in frequency domain. A new element in our approach (for MFT-method) is modeling of the bottom using a two-components model: water plus mineral particles. The main parameter (used as one of the matching parameters), i.e., the characterizing sediment is porosity, sound speed and density of bottom can be expressed through porosity. Two mechanisms of bottom losses are considered: literally absorption-transformation of sound energy into heat (linearly depending on frequency), and volume scattering within bottom (depending on frequency as power function with the degree determined by dominating types of scatterers). As a result of MFT bottom parameters are estimated. It is shown that on the low-frequency (less than 40 Hz) bottom absorption dominates on the other side, for more the high-frequency loss in sediment is conditioned by volume scattering. [Work supported by RFBR, Grant 97-05-64878.]

TUESDAY MORNING, 2 NOVEMBER 1999 KNOX ROOM, 8:30 A.M. TO 12:05 P.M.

Session 2aBB

Biomedical Ultrasound/Bioresponse to Vibration and Signal Processing in Acoustics: Image and Signal Processing in Biomedical Ultrasound

Shira L. Broschat, Cochair
School of Electrical Engineering and Computer Science, Washington State University, Pullman, Washington 99164-2752

Hua Lee, Cochair
Department of Electrical Engineering, University of California, Santa Barbara, California 93106-2991

Invited Papers

8:30

2aBB1. A two-dimensional array system for studies of ultrasonic imaging with aberration correction. Robert C. Waag, Daniel B. Phillips, James C. Lacefield, Carsten G. Draeger, Feng Lin, and Makoto Tabei (Ultrasound Res. Lab., P.O. Box 648, Univ. of Rochester Med. Ctr., 601 Elmwood Ave., Rochester, NY 14642, waag@ece.rochester.edu)

A two-dimensional array system is described for pulse-echo studies of aberration correction. The transducer array is an 80×80 array with a center frequency of 3.0 MHz and a -6-dB bandwidth of 56%. At the center frequency, each element has a physical size of 1.04 wavelength and spacing of 1.2 wavelength. A multiplexer accesses any contiguous 128 elements for transmission and any contiguous 16 elements for simultaneous reception. Transmit electronics have independently programmable waveforms. Each receive channel includes a 20-MHz, 12-bit A/D converter, and a time varied gain programmable over 40 dB. Transmit and receive apertures up to the size of the array are formed synthetically. A method that iteratively predistorts transmit waveforms to produce a transmit focus compensated for aberration has been implemented. Point-spread functions have been measured for propagation through a water path and through a tissue-mimicking aberration path. Pulse-echo images have been formed through a water path, through a tissue-mimicking aberrator, and through the aberrator using aberration correction that consists of time-shift compensation in the transmit–receive aperture or backpropagation followed by time-shift compensation.The system is useful for pulse-echo measurements of aberration, development of adaptive focusing techniques, and formation of high-resolution ultrasonic images using aberration correction.

8:55

2aBB2. Novel breast imaging techniques using transmission ultrasound. Michael P Andre, Helmar S. Janee, Linda K. Olson (SDVA Dept. of Radiol. 114, Univ. of California, San Diego, San Diego, CA 92161, mandre@ucsd.edu), Constance D. Lehman (Univ. of Washington, Seattle, WA), and Barbara A. Fecht (Adv. Diagnostics, Inc., Richland, WA 99352)

Despite several limitations, ultrasound is an important adjunctive modality for detection and management of breast cancer, but it is not recommended for screening. We have explored novel approaches to breast imaging using transmission ultrasound to provide very large image fields, differentiation of breast masses, and improved detection of microcalcifications without the use of ionizing radiation or compression. One method uses a circular array and an iterative Born technique to reconstruct scatter properties in a series of coronal slices through the breast. The second method is based on acoustical holography in which an interference pattern is formed by combining a transmitted field with a reference beam. The signal is converted to the optical domain using a coherent laser detector. This latter method provides real-time images in a large fluoroscopy-like format but with pronounced soft tissue edge definition. A set of large acoustic lenses provides zoom and focal plane control for magnified imaging with enhanced resolution and depth. Both systems have undergone preliminary investigation in large groups of patients with known abnormalities to assess their suitability for detection and characterization of breast disease and to provide accurate guidance for biopsy or tumor ablation. Results of laboratory and clinical findings will be presented.

9:20

2aBB3. Recent progress on nonuniform fast Fourier transform algorithms and their applications. Q. H. Liu, B. Tian, X. Xu, and Z. Q. Zhang (Dept. of Elec. Eng., Duke Univ., Durham, NC 27708-0291)

Recently, nonuniform fast Fourier transform (NUFFT) algorithms have received significant attention [Dutt and Rokhlin, SIAM J. Sci. Stat. Comput. **14**, 1368–1393 (1993)]. Unlike the regular fast Fourier transform (FFT) algorithms, the NUFFT algorithms allow the data to be sampled nonuniformly. The leading order of the number of arithmetic operations for these NUFFT algorithms is $O(N \log_2 N)$. Here, we review the recent progress of the NUFFT algorithms using the regular Fourier matrices and conjugate-gradient method for the forward and inverse NUFFT algorithms [Liu and Nguyen, IEEE Microwave Guid. Wave Lett. **8**, 18–20 (1998); Liu and Tang, Electron. Lett. **34**, 1913–1914 (1998)]. Because of their least-square errors, these NUFFT algorithms are about one order of magnitude more accurate than the previous algorithms. These NUFFT algorithms have been applied to develop the nonuniform fast Hankel transform (NUFHT) and nonuniform fast cosine transform (NUFCT) algorithms. Both NUFFT and NUFHT algorithms have been used to solve integral equations in computational electromagnetics and acoustics; The NUFCT has been used to solve time-dependent wave equations. Numerical examples will demonstrate the efficiency of the fast transform algorithms, and the applications in computational electromagnetics and computational acoustics.

9:45

2aBB4. Noncontact ultrasonic imaging for the evaluation of thermal injury. Joie P. Jones (Dept. of Radiological Sci., Univ. of California—Irvine, Irvine, CA 92697-5000)

Although conventional wisdom suggests that ultrasonic imaging of the body cannot be accomplished without direct contact (or at least via water coupling), we have shown that noncontact imaging through air is possible, certainly for superficial body regions, provided judicious choices of piezoelectric materials and matching layers are made. In preliminary experiments and clinical studies reported here, noncontact imaging is demonstrated for the evaluation of thermal injury (including the quantitative measurement of burn-depth), for the assessment of wound healing, and for the examination of assorted skin lesions. Specifically, in the case of thermal injury, reflections from the dermal/fat interface in human skin is clearly seen using a noncontact 5-MHz transducer. Such measurements are sufficient to determine burn-depth which, in turn, are sufficient to provide, for the first time, a quantitative and noninvasive method for burn evaluation and treatment specification. Evaluating over 500 burn sites in some 100 patients, noncontact ultrasound showed a much greater accuracy and sensitivity than standard clinical assessment. Our method is applicable to a conventional clinical environment as well as a battlefield situation and should prove particularly effective for large-scale medical triage.

10:10–10:20 Break

Contributed Papers

10:20

2aBB5. Focus-directed processing of acoustic holography images. Ruming Yin, Shira L. Broschat (School of Elec. Eng. and Computer Sci., Washington State Univ., P.O. Box 642752, Pullman, WA 99164-2752), and Patrick J. Flynn (The Ohio State Univ., Columbus, OH 43210-1272)

Acoustic holography is a transmission mode imaging technique which was first proposed in the 1970s. As with optical holography, an image is obtained using coherent interference of the transmitted acoustic signal with a reference signal. The interference pattern is illuminated with a laser, and the resulting image is digitized. However, since image reconstruction is performed optically, acoustic holography introduces a focusing problem characteristic of optical systems. Ideally, an image is focused on tissues at a given depth along the optical axis—that is, in one planar slice of the object. In practice, the image is focused over a range of depths so that objects at different depths are blurred but still visible. In this paper, we consider several postprocessing algorithms developed to improve images obtained using an acoustic holography system. First, a focus measure technique is used to determine when the object of interest is best focused. Second, a technique called depth from focus is used to determine the depth of an object. Third, a technique is developed to increase the ''in focus'' interval, or focusing range. These techniques will be discussed and imaging results will be presented. [Work supported by the National Science Foundation and the Carl M. Hansen Foundation.]

10:35

2aBB6. A new *k*-space method for simulation of ultrasonic propagation in tissue. T. Douglas Mast (Appl. Res. Lab., Penn State Univ., University Park, PA 16802, mast@sabine.acs.psu.edu), D.-L. Donald Liu (Siemens Medical Systems, Issaquah, WA 98027), Laurent P. Souriau, Adrian I. Nachman, and Robert C. Waag (Univ. of Rochester, Rochester, NY 14642)

A new *k*-space method for large-scale computations of ultrasonic propagation is presented. In the new method, spatial derivatives from the second-order acoustic wave equation for inhomogeneous media are evaluated by Fourier transformation. Solutions are advanced in time using a $k-t$ space Green's function. Computational results indicate that the new method shares advantages of both past *k*-space and pseudospectral methods. For scatterers with properties similar to soft tissue, the *k*-space method provides much higher accuracy and lower computational cost than a 2–4 finite-difference time domain method. The *k*-space method also allows high accuracy to be obtained for time steps much larger than those required by a leapfrog pseudospectral method. The low dispersion inherent to the *k*-space method is illustrated by large-scale quasi-one-dimensional computations, in which pulse waveforms incur negligible shape change for propagation distances as large as 1000 wavelengths. Example applications of the *k*-space method are demonstrated, including simulation of propagation through a large-scale tissue cross-sectional model and incorporation of a *k*-space solver into a nonlinear inverse scattering method employing eigenfunctions of the far-field scattering operator.

10:50

2aBB7. Calibration of a two-dimensional array system for ultrasonic aberration correction. James C. Lacefield, Daniel B. Phillips, and Robert C. Waag (Dept. of Elec. and Computer Eng., Univ. of Rochester, Rochester, NY 14627-0126)

Two procedures for electronic compensation of array spatial and temporal nonidealities are compared. The motivations for array calibration are to account for signal variations due to differences in the impulse responses of the elements and to reduce beam degradation caused by nonideal element directivities. Both methods calculate inverse filters to equalize the signal at each element. One approach equalizes the array response to a planar reflector, while the other equalizes the response at a specified focal point. The point calibration method reduces the standard deviation of arrival time fluctuations in the measured wave front from a point reflector from 21 to 2 ns and reduces the standard deviation of energy level fluctuations from 2.5 to 1.7 dB. However, the point method also diverts more energy outside the main peak of the focused beam, which causes the -10-dB peripheral energy ratio to increase from 0.30 to 0.32. Point calibration of the receive aperture is nevertheless desirable for aberration correction using backpropagation followed by time-shift compensation because the fidelity of the correlation algorithm is dependent upon unbiased measurement of the echo wave front at the receive aperture.

11:05

2aBB8. Comparison of high-frame rate and delay-and-sum imaging methods. Jian-yu Lu and Anjun Liu (Ultrasound Lab., Dept. of Bioengineering, The Univ. of Toledo, Toledo, OH 43606, jilu@eng.utoledo.edu)

Recently, a high-frame rate imaging method has been developed with limited diffraction beams to construct either two-dimensional (2-D) or three-dimensional (3-D) images (up to 3750 frames or volumes/s for biological soft tissues at a depth of about 200 mm). In this talk, the new method is compared with the conventional delay-and-sum (dynamic focusing) method. Both computer simulation and experiment results show that the quality of images constructed with the two methods are virtually identical when the maximum Axicon angle of X waves in the high-frame rate method is approaching to 90 degrees. Theoretical analysis is carried out to confirm the results. This is significant because the high-frame rate method requires thousands of times less computations while achieving the same high-imaging quality as the conventional delay-and-sum method. [This work was supported in part by grant HL 60301 from the National Institutes of Health.]

11:20

2aBB9. High-frequency dependence of the backscatter coefficient on selected bovine tissues from 10–30 MHz. Subha Maruvada, Kirk K. Shung, and Shyh-Hau Wang (Grad. Prog. in Acoust. and The Bioengineering Prog., University Park, PA 16802)

Very high-frequency diagnostic ultrasonic imaging operates at frequencies of 20 MHz and higher. Thus it is critical to obtain data on ultrasonic attenuation and scattering in this frequency range. At high frequencies, it is not feasible to make scattering measurements with unfocused transducers due to their decreased sensitivity, therefore focused transducers are needed. Using the standard substitution method to calculate the backscatter coefficient, as is used with unfocused transducers, yields erroneous results for focused transducers. The assumption that the reflected echo from a perfect reflector in the far field can be calculated as though the transducer acted like a point source is not valid for focused transducers. A method is presented for focused transducers where the flat reflector is substituted by a particulate reference medium whose backscatter coefficient is well known and documented, in this case, a red cell suspension. Results between focused and unfocused transducers match closely between 10 and 20 MHz. The backscatter coefficient for bovine tissues has been well documented between 1 and 10 MHz. These measurements have been extended to 30 MHz. The frequency dependence of backscatter on bovine tissues will be presented in the range 10–30 MHz and compared to previous results.

11:35

2aBB10. Assessing arterial stenoses by tracking turbulence with Doppler ultrasound. Megan M. Miller, Christy K. Holland (Dept. of Radiol., M.L. 0742, Univ. of Cincinnati, Cincinnati, OH 45219-2316, Christy.Holland@uc.edu), and Peter J. Disimile (Univ. of Cincinnati, Cincinnati, OH 45267-0700)

When a stenosis, or narrowing, causes a significant area reduction of a blood vessel, turbulence in the post-stenotic jet can be detected with Doppler ultrasound distal to the stenosis. A technique is investigated for assessing the severity of a stenosis, defined as the pressure drop across the lesion, by extracting the streamwise turbulence intensity (or the normalized square root of the velocity variance) from the Doppler ultrasound in an arterial flow model. The model consists of an optically and acoustically transparent polyurethane tube that mimics femoral artery compliance, a pump capable of continuous and pulsatile flow, 10-μm glass spheres as an ultrasound and laser scatterer, and both blunt and rounded inlet stenoses. The flow field through three axisymmetric, Plexiglas stenoses with diameter reductions from 60% to 95% mH were investigated with an ATL HDI 3000 using the L7-4 linear array in Doppler mode (1.0-mm spatial pulse length) and a Dantec two-color 55X laser Doppler anemometer (0.7-mm major axis). To validate the ultrasound technique, correlation of Doppler ultrasound and laser Doppler anemometry flow measurements was examined. The correlation of the peak velocity, the maximum turbulence intensity, and the pressure drop across each stenosis was also investigated. This Doppler ultrasound technique could be sensitive to more subtle alterations in hemodynamics and therefore could aid early detection of atherosclerosis.

11:50

2aBB11. Backward propagation algorithms for image reconstruction: Signal processing, algorithm architecture, and applications. Hua Lee (Dept. of Elec. and Computer Eng., Univ. of California, Santa Barbara, CA 93106)

Traditionally, image reconstruction algorithms were developed, following the design and configurations of the data-acquisition systems. So, the algorithms are typically special-purpose and system-specific. Consequently, there is a lack of consistency in terms of algorithm architecture, organization, and performance. In this paper, we present a unified framework for algorithm design and development. This allows us to implement image reconstruction for various data-acquisition configurations including active or passive sensing, linear or circular receiving apertures, CW or wideband illumination, and monostatic or bistatic formats, based on a single theoretical framework in an organized manner. The computation schemes for both linear and circular apertures will be discussed in detail. The layered backward propagation technique, as the main processing modality, provides the flexibility for dynamic updating for changes of propagation parameters. In addition, we illustrate parallel processing and recognition as integrated components for the algorithm structure. The presentation includes the theoretical background on signal processing for image formation, and overview of algorithm architecture for various configurations, a discussion on computation complexity and commonality, and several applications.

Session 2aEA

Engineering Acoustics: Compatibility of Hearing Aids and Cellular Telephones: Standards Progress

Stephen C. Thompson, Chair
Knowles Electronics, Inc., 1151 Maplewood Drive, Itasca, Illinois 60143

Chair's Introduction—8:30

Invited Papers

8:35

2aEA1. Suitcase lab for measuring digital cellphone interference in hearing aids. Mead C. Killion (Etymotic Res., Inc., 61 Martin Ln., Elk Grove Village, IL 60007) and Harry Teder (Consultant to Hearing Industries Assoc., Excelsior, MN 55331)

A low-cost, ''real-life'' method for measuring the interference caused by cellular telephones in hearing aids is proposed. Data would be valid for specific phone and hearing aid models. Estimated equipment cost is about $500. Real−ear recordings will demonstrate the audible interference.

9:05

2aEA2. Real-ear measurement of hearing-aid interference from digital wireless telephones. Harry Levitt (Ctr. for Res. in Speech & Hearing Sci., City Univ. of New York, 33 W. 42nd St., New York, NY 10036), Judy Harkins, Linda Kozma-Spytek (Gallaudet Univ., Washington, DC 20002), and Eddy Yeung (City Univ. of New York, New York, NY 10036)

Two studies were performed. The first measured hearing-aid interference under field conditions. Signal levels in the ear canal were measured using an in-the-canal probe system, the output of which was stored digitally in a laptop computer. Both bystander and user interference were measured as well as the user's ratings of intelligibility, annoyance, and usability (of the digital wireless telephone). The levels of user-interference were found to be unacceptably high for the large majority of hearing-aid wearers. In contrast, bystander interference was at a relatively low level under conditions typical of telephone use. The second experiment controlled the level of interference by means of the test mode in the digital wireless telephone. Speech-to-interference ratios were measured in the ear canal for various levels of intelligibility and usability, as rated by the subjects. Data were obtained for 37 subjects on each of three digital wireless technologies (PC1900, TDMA, CDMA). [Research supported by NIDRR.]

9:35

2aEA3. Wireless hearing aid compatibility—handset testing requirements. H. Stephen Berger (Siemens Information and Communication Products LLC, 2205 Grand Ave. Pkwy., Austin, TX 78728, stephen.berger@icp.siemens.com)

The Hearing Industry and Wireless Industry has been working together since 1996 to develop a standard to resolve this issue. This paper reports on the technical challenges and the testing requirements for wireless handsets. It concludes with a brief discussion of the need for consumer education and involvement in order to achieve the optimal solution to this issue. Hearing aids operate in two modes. In their primary mode a microphone is used to sense acoustic signals. The secondary mode uses an inductive coil, known as a T-Coil, to receive a magnetic signal which has been modulated by the audio signal. For the wireless handset, which may be used with a hearing aid, there are test requirements arising from both modes. The field strength of both the E and H field, in the area intended for use by a hearing aid must be measured and assured to be within specified parameters. For the T-Coil mode a magnetic signal, modulated by the audio, must be assured. However, in T-Coil mode there is also the issue of assuring that the signal has sufficient quality and does not suffer from magnetic interference from secondary sources.

10:05

2aEA4. Hearing aid measurement and consumer counseling. Thomas A. Victorian (Starkey Labs., Inc., 6700 Washington Ave. S., Minneapolis, MN 55344)

The relationship between digital cellular telephones and hearing aid devices requires specific measurement techniques to determine system compatibility between the two technologies. The ANSI C63.19 working group has developed such measurement techniques related to the actual usage of a digital phone while wearing a hearing aid—known as ''near field'' or ''user'' condition. The hearing aid rf near-field immunity test requirements involve hearing aid orientation and bias as well as rf near-field simulated test conditions. The testing is performed in both the audio and telephone-coil modes. The measurement figure of merit is defined as IRIL (input referred interference level) which normalizes the hearing aid acoustic gain. It can be shown that hearing aids will be required to withstand digital phone E-field emission levels greater than 50 dB V/m and H-field emission levels greater than −2 dB A/m. By characterizing the simulated phone antenna properly this test enables hearing aid immunity testing within a controlled rf laboratory. The outcome of these measurements will be used to classify hearing aids for compatibility with categorized digital phones. This classification process will assist the hearing aid dispenser in patient counseling.

10:35–10:45 Break

10:45

2aEA5. Development of hearing-aid near-field EMI testing. Marco Candiago (Unitron Industries, Ltd., 20 Beasley Dr., Kitchener, ON N2G 4X1, Canada)

With the introduction of digital cellular telephone technology, cellular telephone interference to hearing aids is now a noticeable problem for both hearing-impaired digital cellular phone users and bystanders. Previously, bystander hearing-aid immunity levels (far field) were defined by the IEC committee and are contained in the current IEC 118-13 hearing-aid electromagnetic immunity standard. With the realization that an up-scaled far-field test would be insufficient to provide an accurate correlation of digital cellular telephone usability, the ANSI C63.19 working group developed a near-field test procedure using calibrated dipole antennas. This paper discusses the development of near-field testing, broken down in the following topics: (1) digital cellular phone interference and spectral properties; (2) rf interference coupling into the hearing aid; (3) far-field testing and rationalization behind the use of 1-kHz 80% AM modulation to quantitatively determine hearing-aid immunity characteristics; (4) near-field simulation using dipoles and antenna calibration; (5) near-field repeatability testing; and (6) preliminary results of near-field hearing-aid interference measurements. The result of these developments has led directly to the formulation of the near-field test procedure described in Sec. 5 of the currently developing ANSI C63.19 standard.

11:15

2aEA6. Validation of the C63.19 standard: Acoustic measurements and subjective evaluations. Robert E. Schlegel, Hank Grant, and Tamy L. Fry (Wireless EMC Ctr., School of Industrial Eng., Univ. of Oklahoma, 100 E. Boyd, Ste. R-208, Norman, OK 73019, schlegel@ou.edu)

The ANSI C63.19 measurement and performance standard addresses the electromagnetic compatibility of hearing aids with wireless communication devices. Validation of the measurement procedures and performance criteria comprised a pilot study, objective validation, and subjective validation. Objective validation involved acoustic measurements of 80 combinations of hearing aids and wireless phones. The E-field and H-field immunity levels of each hearing aid and the E-field and H-field emissions of each wireless phone were measured according to the standard. These data were used to theoretically predict the overall input referenced interference level. Across a broad range of hearing aid immunity levels, the measured acoustic interference was accurately predicted from the individual C63.19 measurements. Subjective validation involved the testing of twenty hearing aid wearers with six phone technologies. Hearing aids offering a variety of immunity levels were custom manufactured for the participants and tested according to the C63.19 measurement procedures. While using each wireless phone, participants provided ratings of the speech intelligibility, usability, and annoyance caused by any rf interference. In addition to confirming the existence of compatible hearing aids and phones, the results of this study contributed to the setting of the immunity and emissions criteria for a usable hearing aid wireless phone system.

Contributed Paper

11:45

2aEA7. Clinical studies of digital wireless phone interference in hearing aids. Robert E. Schlegel, Randa L. Shehab, Tamy L. Fry, and Hank Grant (Wireless EMC Ctr., School of Industrial Eng., Univ. of Oklahoma, 100 E. Boyd, Ste. R-208, Norman, OK 73019, schlegel@ou.edu)

Two clinical studies were conducted to evaluate the impact of electromagnetic interference from three digital wireless phone technologies on hearing aid wearers. Sixty-eight participants were tested with their own hearing aids to determine the interference detection threshold and annoyance, along with the impact on speech intelligibility. Bystander interference varied as a function of the hearing aid type and the phone technology. The interference was significant only when the digital wireless phone was within 2 ft of the hearing aid user. Significant declines in speech intelligibility were observed as a function of the specific phones used. In the second study, interference signals for three phone technologies were generated at five sound pressure levels from 35 to 75 dB and mixed with speech at 65-dB SPL to test the speech intelligibility of 24 hearing-impaired participants. Scores for the TDMA-217 Hz phone signal at low speech-to-noise ratios (less than 10 dB) were significantly lower than those for CDMA and TDMA-50 Hz at the same level. Speech intelligibility scores at speech-to-noise ratios of 20 and 30 dB were similar to those for a ''no noise'' condition. The articulation index accurately represented the impact of the wireless phone interference on speech intelligibility.

Session 2aED

Education in Acoustics: Undergraduate Laboratory Experiments

Daniel A. Russell, Chair
Science and Mathematics Department, Kettering University, 1700 West Third Avenue, Flint, Michigan 48504-4898

Chair's Introduction—7:55

Invited Papers

8:00

2aED1. Classical measurements of sound. Elmer L. Hixson (Univ. of Texas, Austin, TX 78712)

From the 1500s the study of sound involved acoustics and vibrations. Early theoreticians tried to experimentally justify their analytical models. Many vibrating devices were available to generate acoustic waves but few means were available to quantify these sources and sound propagation. The human ear was the only means of amplitude measurement and spectral analysis. The development of electrical and electromechanical technology allowed great advancements in the measurement of sound. Bell's telephone transmitter/receiver and Flemming's valve brought great strides in quantifying sound. The capacitor microphone brought high accuracy. The advent of the oscilloscope and electrical filters provided analysis of complex sounds. This level of technology served the field well until the advent of the computer age.

8:30

2aED2. Acoustics laboratory experiments for all. Uwe J. Hansen (Dept. of Phys., Indiana State Univ., Terre Haute, IN 47809) and Thomas D. Rossing (Northern Illinois Univ., DeKalb, IL 60115)

For most students, understanding scientific principles requires hands-on experience, and that is the reason for having undergraduate laboratories. This is especially true of acoustics, in which the laboratory experience is important not only to beginning and advanced students in science and engineering but also to students taking a general education course. We will describe some favorite experiments on the introductory, intermediate, and advanced levels. Most of them are based on experiments in *Acoustics Laboratory Experiments*, a collection of 52 experiments on all levels. Some of them are appropriate for introductory courses in musical acoustics or speech and hearing, others are suitable for physics and engineering students in more advanced courses. Ways to incorporate acoustics experiments into courses not having a scheduled laboratory period will also be discussed.

9:00

2aED3. A senior engineering design elective with industry and community-sponsored noise control projects. Robert Celmer (Acoust. Prog. and Lab., College of Eng., Univ. of Hartford, 200 Bloomfield Ave., W. Hartford, CT 06117, celmer@mail.hartford.edu)

One of the transitions that engineering students must make as they enter the work force is the progression from theoretical concepts, or *book learning* to applied or *real world* applications. At the University of Hartford, engineering design courses serve as culminating experiences devised to bridge this passage. One such course, *Noise Control Design*, challenges the student to apply the past three years' conceptual base to a problem solving opportunity replete with actual scenarios encountered in industry or the community. Each year local firms and organizations approach the laboratory for assistance with a variety of sound or vibration problems. After an initial training period, students make use of the laboratory's dual channel FFT and real time analyzers, anechoic chamber, sound intensity and modal analysis software, acoustic modeling software, vibration tables and transducers, portable sound level meters and digital tape recorders. Using a consultant-client model, students work collaboratively in teams of two defining the problem, developing a method of approach, making appropriate measurements, devising alternate solutions, and ultimately delivering a written and oral presentation at the end of the semester. The paper discusses specific projects and some experiences students had with their first industrial assignment.

9:30–9:40 Break

Contributed Papers

9:40

2aED4. Laboratory experiments as demos and projects in the underwater acoustics and sonar course. Elizabeth L. Simmons and Murray S. Korman (Dept. of Phys., U.S. Naval Acad., Annapolis, MD 21402, korman@nadn.navy.mil)

Underwater Acoustics and Sonar (SP411) is a 3-h course that is typically offered to Midshipmen in their senior year. General science majors take the course in the fall while the oceanography majors enroll in the spring. A sprinkling of physics, electrical engineers, ocean engineers, and systems majors also populate the course (totaling ~110 students/yr). Since this course is offered without a lab, the ''in-class'' experience has been enhanced with the development (over many years) of our demo carts which surround the classroom. Although Friday is our major ''demo day,'' demos are performed throughout the week. They motivate the students' ''out-of-class'' experimental projects. Demos include: (a) waves on slinkies; (b) Fourier analysis of tones in noise, homemade musical instruments; (c) harmonic synthesis; (d) receiver operating characteristics from processed signals in noise; (e) two-element and loudspeaker beam patterns; (f) sound speed versus temperature in water; (g) target strength versus angle of a model sub; (h) Ref. coef. from an Al/water interface; (i) PC-IMAT (interactive multisensor analysis training) simulations of array steering, ray tracing, active sonar, propagation loss; and (j) FM detection and Doppler effects. Students get involved with the measurements, have fun, and their understanding of underwater sound is greatly enhanced.

9:55

2aED5. The acoustic interferometer as a teaching laboratory experiment. Thomas B. Gabrielson (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804, tbg3@psu.edu)

Interferometry is a popular subject for teaching-laboratory experiments in optics. Furthermore, some important classes of acoustic and vibration sensors rely on optical interferometry. Consequently, the fundamentals of interferometry and the interpretation of the resulting signals are important concepts even outside the field of optics. Unfortunately, procurement, alignment, and maintenance of optical interferometers can be prohibitively expensive for a program that does not specialize in optics. For institutions lacking a substantial infrastructure in free-air or fiber optics, the acoustical interferometer is an attractive alternative. Most of the issues associated with optical systems can be incorporated into straightforward laboratory exercises. The components for a 40-kHz acoustical system are inexpensive and readily available. The alignment is greatly simplified by the much longer wavelength and the function of the components is more intuitive. Moreover, absolute displacement calibration can be done with conventional micrometer translation stages; the effects of different bias points can be investigated; and the characteristics of large-signal behavior in interferometers can be studied. For advanced experiments, a heterodyne interferometer can be constructed along with a simple demodulator. [Work supported by the Graduate Program in Acoustics at Penn State.]

10:10

2aED6. Vortex-driven whistling tube undergraduate experiment. Timothy W. Lancey and Tracy J. Haeggstrom (Dept. of Mech. Eng., California State Univ. Fullerton, Fullerton, CA 92834)

An experiment to investigate the tones produced by the generation and shedding of periodic vortices in a tube was designed, constructed, and implemented as a student project in a senior laboratory course. A 0.102-m-diam by 0.762-m-long lucite tube was installed in a subsonic wind tunnel; the longitudinal axis of the tube was located streamwise with the mean flow, laterally centered in the test section, 0.178 m above the floor. Two lucite rings, with minimal clearance at the tube inside diameter and each having inside diameters of 0.0762 m, were connected inside the tube. One of the rings was at the tube inlet plane, while the other was slightly downstream, at a location which could be varied. A microphone was attached to the tube near the exit plane and connected to a PC which incorporated a data acquisition system. Airflow through the wind tunnel was varied until tones were heard, and the air velocities and frequencies of the tones were recorded. Results of the data analysis indicated that the frequency of vortex shedding, at the measured air velocity and ring separation, was in agreement with the natural frequency of the tube, for each of the audible tones.

10:25

2aED7. Demonstration of Doppler effects and sound-speed measurement using an analog frequency-to-voltage (FM) detection circuit. Angela K. Huegel, Michael S. Naff, and Murray S. Korman (Dept. of Phys., U.S. Naval Acad., Annapolis, MD 21402, korman@nadn.navy.mil)

An audio frequency FM detector has been designed and built to measure very small Doppler shifts for classroom demonstrations. This detector uses a limiter, a series $L-C$ phase-shift network, two sine-to-square wave circuits, a multiplier and low-pass filter. One can easily measure a deviation of 1/20 Hz in a 4000-Hz carrier. The apparatus consists of a stationary piezo-tweeter radiating a tone near a microphone that translates with periodic motion, using a motorized disk and linkage mechanism, which drives a translation stage. The output of the mic goes to the FM detector which outputs to a digital scope or meter for display. Calibration is extremely linear and is performed using the digital frequency readout on the generator. From a measure of the peak to peak Doppler shift Δf (from the periodic Doppler shift signal versus time), the source frequency f, the radius R of the disk, and the period T, one can predict the sound speed to be $C=(f/\Delta f)(4\pi R/T)$. Here, an approximation involving the linear velocity has been made. Results for C are typically within 1% of the expected value for air at room temperature.

10:40

2aED8. New acoustics experiments on an old NeXT computer. Daniel A. Russell (Sci. and Math Dept., Kettering Univ., Flint, MI 48504, drussell@kettering.edu)

The manufacture of NeXT computer hardware was discontinued in 1993. However, the operating system (now known as OPENSTEP and Macintosh OS-X) is still alive and kicking, and refurbished NeXT systems are still available at greatly reduced cost. Coupled with an A/D converter, the NeXT computer is a powerful, and relatively inexpensive, tool for an undergraduate acoustics laboratory. There is a fairly wide selection of free acoustics software available over the internet, including applications for digital sound recording, mixing, and editing; oscilloscopes; FFT analysis; spectrum analysis; MIDI sequencing; sound file conversion; and signal generation. In addition, there are some excellent new commercial software packages which allow the NeXT to act as a professional quality digital function generator and audiometer. This paper will summarize how NeXT computers are used at Kettering University for undergraduate acoustics laboratory experiments involving digital sound recording and manipulation, signal analysis, Fourier synthesis, principles of digital audio, noise, and human hearing.

Session 2aMUa

Musical Acoustics: Musical Instruments: Experiments and Modeling

Thomas D. Rossing, Chair
Physics Department, Northern Illinois University, DeKalb, Illinois 60115

Invited Paper

8:30

2aMUa1. Acoustics of Baltic psaltery. Andres Peekna (Innovative Mech., Inc., 265 Coe Rd., Clarendon Hills, IL 60514-1029) and Thomas D. Rossing (Northern Illinois Univ., DeKalb, IL 60115)

The Baltic psaltery, a plucked-string instrument, is known in Finland as ''kantele,'' in Estonia as ''kannel,'' in Latvia as ''kokle,'' in Lithuania as ''kankles,'' and in northwestern Russia as the ''wing-shaped gusli.'' In its archaic form, it had a limited range, $\sim 5-13$ strings, usually tuned diatonically, though the lower strings were sometimes tuned as bourdons, several tones lower than the other strings. Results of experimental investigations on several instruments are presented. Methods used for calculating the lowest resonance of the air cavity are described. Investigations include determining the modes of vibration using electronic TV holography and recording sound spectra throughout the tonal range. Results indicate that the more successful instruments have many resonances close together in frequency, within their limited tonal range. This may account for why the archaic Baltic psalteries had several small soundholes rather than the single soundhole found on guitars and mandolins, which have much wider tonal ranges. Several instruments will be demonstrated.

Contributed Papers

9:00

2aMUa2. Asymmetrical properties of a vibrating wire and their effects. Roger J. Hanson, H. Kent Macomber, and Andrew Morrison (Dept. of Phys., Univ. of Northern Iowa, Cedar Falls, IA 50614, hansonr@uni.edu)

Some effects of the asymmetries causing a splitting of the fundamental natural vibrating frequency of a wire have previously been reported [Hanson *et al.*, J. Acoust Soc. Am. **103**, 2873 (1998)]. It has been demonstrated in this work on brass harpsichord wire that the splitting of the frequency is due to intrinsic properties of the wire itself and not of asymmetries in the end clamps. The two frequencies are associated with two definite orientations with respect to the wire. These two vibrational directions have been determined to be orthogonal within one degree experimental uncertainty. This orthogonality is in agreement with predictions of a simple model which assumes that, for small-amplitude free vibrations, the observed portion of the wire moves under the action of a linear anisotropic conservative restoring force. Measured splittings for several samples of harpsichord wire have ranged from 0.12 to 0.30 Hz for a frequency of about 70 Hz. The nonlinear effects of generation of motion perpendicular to the driving direction [Hanson *et al.*, J. Acoust. Soc. Am. **96**, 1549–1556 (1994)] and generation of higher harmonics are profoundly influenced by the orientation of the driving direction with respect to these vibrational orientations for low-driving forces. Related effects for plucked strings will be discussed.

9:15

2aMUa3. Physical modeling of the piano: First results for a simple monochord. N. Giordano (Dept. of Phys., Purdue Univ., West Lafayette, IN 47907)

Initial results are presented for a project aimed at constructing a physical model of the piano. Newton's laws are used to calculate the motion of the hammer, strings, soundboard, and room air, and thereby compute the sound produced by the instrument from ''first principles.'' In this talk we focus on a prototype instrument consisting of a monochord with a small soundboard. Modeling results for this geometry are described, and compared with experimental results for a laboratory version of the instrument. If possible, calculated tones will be presented. [Work supported by NSF Grant PHY-9722031.]

9:30

2aMUa4. Sound production by a vibrating piano soundboard: Theory. Minghui Jiang and N. Giordano (Dept. of Phys., Purdue Univ., West Lafayette, IN 47907)

We describe a theoretical study of the sound generated by a piano soundboard. The soundboard model, which is similar to that developed recently by our group, is driven harmonically at a point on one of the bridges, and the resulting vibrational motion obtained by numerical solution. This soundboard is situated in a numerical room, and the resulting room pressure is calculated with two different methods. In one approach the pressure throughout the room is obtained by solution of the appropriate coupled equations for the air pressure and velocity. In the other treatment every point on the soundboard is treated as a simple source, each of which contributes to the pressure at a specified listening location. The results of these two approaches are compared to each other, and to recent experiments by our group. [Work supported by NSF Grant PHY-9722031.]

9:45

2aMUa5. Spectral characteristics and efficient critical-band-associated group synthesis of piano tones. Hua Zheng and James W. Beauchamp (Dept. of Elec. and Computer Eng., Univ. of Illinois at Urbana-Champaign, 5237 Beckman Inst., 405 N. Mathews St., Urbana, IL 61801)

The physics of piano sound production has been studied extensively, while little has been done to explore its spectral characteristics for efficient and perceptually accurate digital synthesis. Piano tones with a large variety of explicit performance parameters were recorded from a MIDI-controlled acoustic piano, and spectral modeling methods were used for the analysis. A number of features have been concluded: (1) relationship between harmonic structure and pitch for six levels of dynamic; (2) relationship between spectral envelope and dynamic for all pitches; (3) typical amplitude envelopes; (4) effects of the sustain and soft pedals; (5) inharmonicity characteristics; (6) noise characteristics. A multiple-wavetable synthesis algorithm implementing a group synthesis model was developed. Each harmonic group corresponds to one or more consecutive critical bands. Amplitude versus time envelopes of all harmonics within one group share a weighted-average shape with individual weight applied to each harmonic. With appropriate numbers of groups and characteristic noise, convincing results have been obtained for the entire range of pitches and dynamics. The synthesis algorithm has been implemented as a subroutine of an analysis/synthesis software package, as a Music 4C instrument, and as a real-time plug-in for a software sound server.

10:00

2aMUa6. Acoustics of flute sound in laboratory experimental conditions. Isabelle Cossette (Meakins-Christie Labs., McGill Univ., 3626 St-Urbain St., Montreal, QC H2X 2P2, Canada and NVC, Univ. of Sydney, NSW 2006, Australia) and C. William Thorpe (NVC, Univ. of Sydney, NSW 2006, Australia)

Experimental studies of musical performance are often performed in laboratories with sometimes invasive apparatus. A concern often raised by musicians is that recordings obtained in such laboratory conditions may not reflect what would occur in a real performance. This study aims to provide an assessment of the extent to which data acquired in an invasive experiment, involving the physiology of flute performance, represents what would occur in actual performance. Five professional flutists, playing on the same flute, performed four well-known pieces of the flute repertoire in invasive and noninvasive conditions. The invasive condition consisted of apparatus on the flute head joint to measure the jet velocity and lip aperture, and trans-nasal catheters to measure respiratory pressures. The noninvasive condition occurred before the apparatus had been attached, but with other conditions kept as consistent as possible. The acoustic output of the flute was recorded and characterized by measuring the fundamental frequency and relative magnitudes of the harmonics. The results did not show any significant change in the acoustic parameters overall, although several subjects exhibited some differences (<6 dB) in harmonic amplitudes between the conditions. We therefore conclude that, in professional players at least, performance is not unduely compromised by experimental constraints.

10:15

2aMUa7. A hybrid waveguide model of the transverse flute. Mark A. Bartsch (Univ. of Dayton, Dept. of Elec. Eng., 300 College Park, Dayton, OH 45469, bartscma@flyernet.udayton.edu)

Recent years have seen substantial improvements in the modeling and synthesis of jet-reed instruments such as flutes and organ pipes. Developments in the modeling of woodwinds using so-called digital waveguides have allowed the efficient acoustical modeling of the main body and toneholes of the instrument. Further, a new and more complete model of the jet's behavior in the presence of the edge and the resonator has been formulated for recorderlike instruments [Verge *et al.,* J. Acoust. Soc. Am. **101**, 2925–2939 (1997)]. A new simulation model of the transverse flute is presented which combines the contributions of digital waveguide modeling and the new source model. This new model is defined almost entirely by physical parameters (such as dimensions of the instrument) rather than by the arbitrary adjustment parameters often employed. The model is evaluated by comparing the effects of certain parameters on the model's operation with their effects on the performance of an actual flute. The model is further evaluated for its tuning characteristics by comparing its frequencies of oscillation with the sounding frequencies of a simple flute with matched physical parameters. [Work supported by the University of Dayton Honors Program.]

10:30

2aMUa8. Analysis of cymbal vibrations: Lyapunov spectrum and route to chaos. Cyril Touzé and Antoine Chaigne (ENST, Dept. TSI, CNRS URA 820, 46 rue Barrault, 75634 Paris Cedex 13, France)

A new step in the analysis of cymbal vibrations is presented. The goal of this analysis is to gain a better understanding of the underlying mechanisms that govern the sound of these instruments. In a previous work, geometrical invariants (correlation dimension and percentage of false nearest neighbors) were extracted from the reconstruction of the cymbal vibration in phase space [Touzé *et al.*, Proceedings of ISMA'98, Leavenworth (1998)]. Here, the Lyapunov exponents of the system are computed in order to assess the presence of chaos and to quantify it. The method used consists of approximating the tangent flow of the trajectory in the reconstructed phase space. An algorithm which uses a higher-order polynomial approximation has been developed. This algorithm is validated on a number of well-known theoretical time series (Henon map, Duffing equation, etc.). The Lyapunov spectrum of the cymbal shows, in particular, the convergence of the greatest exponent, which is a strong indication for the presence of chaos. A closer analysis reveals a kind of Ruelle–Takens scenario for the transition from linear to chaotic motion: Power spectra and Lyapunov exponents show that mode-locking and quasi-periodicity are present in the vibrations just before the chaotic regime.

Session 2aMUb

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

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

Chair's Introduction—11:00

Invited Paper

11:05

2aMUb1. Musical acoustics in the Twentieth Century. Gabriel Weinreich (Randall Lab. of Phys., Univ. of Michigan, Ann Arbor, MI 48109-1120)

Of humankind's material creations that developed without benefit of logical understanding, musical instruments may well be the most complex and sophisticated. Although some of the greatest theoretical minds of the nineteenth century (such as Helmholtz and Rayleigh) applied notable efforts to the study of musical instruments, that period lacked the experimental tools to match the virtuosity of its theoreticians; so that it was not until the twentieth century, and more particularly, the period that overlaps the life of the Acoustical Society of America, that the field of Musical Acoustics was truly able to blossom forth. Specifically, ASA's first quarter century saw the development of sophisticated vacuum-tube technology (powerfully stimulated by World War II); the second produced solid-state electronics; and the third brought personal computers to every scientist's desk and made digital methods part of everyday life. But, unlike the situation in other fields, modern electronic methods not only made the physics of musical instruments more accessible, they themselves invaded their subject matter in the form of electronic music, giving rise to a totally new kind of symbiotic relationship. This talk will draw upon the annals of JASA to summarize that fascinating period of history.

TUESDAY MORNING, 2 NOVEMBER 1999 MADISON ROOM, 8:45 A.M. TO 12:00 NOON

Session 2aPA

Physical Acoustics: Earth and Air-Borne Sound

Craig J. Hickey, Chair
National Center for Physical Acoustics, University of Mississippi, University, Mississippi 38677

Contributed Papers

8:45

2aPA1. Simplified model of the seismic reverberation observed due to the high-resolution transient acoustic signal. Yevgeniy Y. Dorfman, Peter A. Krumhansl, and Michael Goldsmith (BBN Technologies, Operating Unit of GTE, 70 Fawcett St., Cambridge, MA 02138)

Unexploded ordinance (UXO) presents a lethal hazard to individuals worldwide. To predict and/or simulate the performance of a seismic sonar for detecting UXO, one needs to develop a model of propagation and reverberation in the inhomogeneous seismic media. An underground seismic detection experiment was performed during fall 1998 as a part of the BBN Innovative Seismic System for Buried Unexploded Ordinance Detection and Classification program. A physics-based model of monostatic reverberation due to the low-directivity shear wave source pointed down was postulated. This model provides a way to (a) simulate the time series for different source-receiver geometries and source signals; and (b) to invert the field measurements for ground parameters. Exercising the model, it was found that the observed signal due to reverberation is a function of shear wave speed (usually known), exponential attenuation, and assumed reverberation strength (often not well-known). It was further found that under realistic assumptions for UXO detection scenario frequency-dependent attenuation must be accounted for in the modeling. A numerical code was developed based on the same physical assumptions in order to account for nonmonostatic source-receiver configuration. It was found that collected data are matched well by the numerical model predictions. [Work supported by SERDP.]

9:00

2aPA2. Acoustic landmine detection using laser scanning vibrometer. James Sabatier and Ning Xiang (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

Airborne acoustic waves that couple into the surface of the ground excite Biot compressional and shear waves. If a minelike target is present below the surface, the ground surface vibrational velocity will show distinct changes due to reflection and scattering of these waves. Taking advantage of a noncontact measurement technique, the surface vibrational velocity is detected with a laser Doppler vibrometer. Sound waves with a wavelength comparable to the object size are suitable for recognizing geometrical shapes of targets, while true wavelike acoustic scattering phenomena can be observed with a shorter acoustic wavelength. In the present contribution, both shape/size recognition and scattering phenomena of buried mines detected by the laser Doppler based acoustic detection will be visualized in terms of a number of 2-D and 3-D animations. [The work is supported by U. S. Army Communications-Electronics Command.]

9:15

2aPA3. Notes on nonlinear elasticity of rocks. Lev A. Ostrovsky (Univ. of Colorado/NOAA Environ. Technol. Lab., 325 Broadway, Boulder, CO 80303)

Recent experiments with samples of Earth materials such as sandstone have shown that they possess a strong nonlinearity, and that water saturation, even in small amounts, may strongly affect nonlinear properties of the material. Here, a theoretical model is considered which represents the rock as a conglomerate of ''large'' grains (of order 100 μm) with two types of contacts: hard ones which determine the basic elastic matrix, and soft contacts, due to much smaller contacting grains which can create strong nonlinearity. Also, the effect of capillary and Van der Waals forces in thin liquid layers between grains is considered. The theoretical estimates are compared with available experimental data.

9:30

2aPA4. Resonance inversion for elastic moduli of anisotropic rocks. Seiji Nakagawa, Kurt T. Nihei, and Larry R. Myer (Earth Sci. Div., E. O. Lawrence Berkeley Natl. Lab., 1 Cyclotron Rd., Berkeley, CA 94720, seiji@friction.lbl.gov)

In this research, acoustic resonance spectroscopy was applied to determine the dynamic elastic constants of isotropic glass and transversely isotropic rock cubes. This technique consists of resonating the specimen over a broad range of frequencies, measuring the resonance frequencies, and computing the elastic constants by nonlinear inversion of the measured resonance frequencies. Specimens were tested under unconfined, traction free conditions. Resulting surface vibrations were measured using a miniature accelerometer and their spectral characteristics were analyzed. The inversion was performed using a numerical algorithm based on the Rayleigh–Ritz method that minimized the difference between measured and computed resonance frequencies iteratively. Mode shapes of the anisotropic specimens were also measured using a laser Doppler vibrometer and compared with the prediction of the numerical model. Comparison between the elastic moduli of rock specimens determined by static loading tests, resonance inversion, and ultrasonic transmission tests showed good agreement between the ultrasonic and resonance results but the moduli determined from ultrasonic measurements were consistently higher than the resonance inversion. Such results may be due to the frequency-dependence of the wave velocity in microscopically heterogeneous rock and nonelastic (frictional) deformation of the rock specimen during the static loading tests.

9:45

2aPA5. Acoustic signatures of a fracture during air injection. Kurt T. Nihei, John E. Peterson, Jr., Larry R. Myer, and Ernest L. Majer (Earth Sci. Div., Lawrence Berkeley Natl. Lab., 1 Cyclotron Rd., M.S. 90-1116, Berkeley, CA 94720)

This study examines the acoustic signatures of transmitted, reflected, and guided waves during air injection into a single, natural, water-saturated fracture in limestone. The presence and location of the fracture were established in a series of geologic, hydrologic, and seismic studies (Queen and Rizer, 1990; Datta-Gupta *et al.*, 1994; Majer *et al.*, 1997) that ultimately led to its verification in core obtained from a slanted well. This work describes the results of a follow-up high frequency (1 to 10 kHz) crosswell survey that was designed to illuminate the fracture by air injection into the fracture. Zero-offset P-wave crosswell transmission and reflection measurements conducted during air injection showed a large decrease in the amplitude of the transmitted wave (approximately 10 times reduction at 3 kHz), and a smaller increase in the amplitude of the reflected wave (approximately 1.5 to 5 times increase at 3 kHz). Measurements of the P-wave and an interface wave propagating along the fracture also show a small increase in amplitude during air injection. Analyses of these measurements using numerical boundary element and finite-difference simulations and the importance of including fracture stiffness heterogeneity arising from irregular distributions of air inside the fracture will be presented.

10:00

2aPA6. Wind generated sound in standing corn. David G. Browning (Dept. of Phys., East Hall, Univ. of Rhode Island, Kingston, RI 02881)

The ''sound of corn growing'' in farming folklore appears to be due to wind puffs, during otherwise calm conditions (especially at night), causing isolated audible popping sounds due to leaf striking leaf [D. G. Browning, Am. J. Botany **84**, 38(A) (1997)]. When the wind becomes steady, the number of events greatly increases, resulting in a rustling sound with a smoothed, broad spectrum between 1 and 5 kHz, and a peak at 2.5 kHz. For a windspeed of 20 mph the peak level is 20 dB above ambient. As the corn matures, the increased weight of the corn ears causes greater stalk sway at a given windspeed; also the leaves tend to become more brittle due to reduced moisture content. Both of these changes appear to alter the measured sound spectra, offering a means for evaluation.

10:15

2aPA7. Seasonal variability in the atmosphere and its effect on infrasonic propagation. David E. Norris and Robert Gibson (BBN Technologies, 1300 N. 17th St., Arlington, VA 22209)

Infrasonic waves can propagate thousands of kilometers in range and sample regions of the atmosphere from the ground up to and including the thermosphere. In this study, seasonal changes in the atmosphere and their effect on infrasonic propagation are characterized. The NASA/NRL empirically based models HWM-93 (for winds) and MSIS-90 (for temperature) are used. Three-dimensional ray traces are computed through the modeled atmosphere for several representative scenarios. Seasonal trends in both ray arrival times and ray azimuth bias are computed, and limited comparisons with data are made where possible. [Sponsored by Defense Threat Reduction Agency, Contract No. DSWA01-97-C-0160.]

10:30–10:45 Break

10:45

2aPA8. Spectral broadening of sound scattered by atmospheric turbulence. George H. Goedecke, Roy C. Wood (Dept. of Phys., New Mexico State Univ., P.O. Box 30001, Las Cruces, NM 88003-8001), Harry J. Auvermann (U.S. Army Res. Lab., Adelphi, MD 20783-1155), and Vladimir E. Ostashev (Environ. Technol. Lab., Boulder, CO 80303)

Scattering of a monochromatic sound wave by atmospheric turbulent eddies that are moving with the mean wind is described. The source and detector have wide radiation patterns and are at rest in a ground fixed frame. For eddies that make the dominant contribution to the detector signal, scattering angles change substantially with time, so the signal displays a time-dependent frequency which may include the full longitudinal Doppler width. A computer code is developed that calculates the time-dependent detector response and its Fourier spectrum due to one or many eddies, including a steady-state collection of eddies of many different scale lengths that models homogeneous and isotropic atmospheric turbulence. Several numerical results from this code are presented, including one for a simulation of a recent experiment. The predicted spectral characteristics are in very good agreement with the experimental ones. Some possible extensions of the model for describing anisotropic and intermittent atmospheric turbulence are discussed. [Work supported in part by the U.S. Army Research Office under Contract Nos. DAAG55-98-1-0463 and DAAG55-97-1-0178, and an NRC-ETL Research Associateship.]

11:00

2aPA9. Numerical solution of a second-moment parabolic equation for sound propagation in a random medium. D. Keith Wilson (U. S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783) and Vladimir E. Ostashev (Environ. Technol. Lab., Boulder, CO 80303)

A parabolic equation for the second-moment of the acoustic pressure field in a refractive medium with random fluctuations is derived using the Markov approxiation. The PE is then discretized and written as a matrix equation for the purpose of numerical solution. The matrix equation involves the product of an $N_z^2 \times N_z^2$ tridiagonal matrix with ''fringes'' times the $N_z^2 \times 1$ second-moment vector, where N_z is the number of points in the vertical grid. (The numerical formulation of the usual PE for the acoustic pressure involves an $N_z \times N_z$ pure tridiagonal matrix.) Direct solution of this extremely large linear system is prohibitive. A pair of alternative strategies are therefore devised. One involves Cholesky factorization of the second-moment matrix. The other is based on iterative solution of the second-moment equation, with the coherent field used to start the iterations. Both of these methods are applied successfully to propagation in an upwardly refracting atmosphere, although solution of the second-moment PE is still very computationally intensive. [Work supported in part by the U.S. Army Research Office under Contract No. DAAG55-98-1-0463, and NRC-ETL Research Associateship.]

11:15

2aPA10. Daytime sound levels and coherence as a function of height: Experiment and theory. Xiao Di (Appl. Res. Lab., Pennsylvania State Univ., State College, PA 16804), Kenneth E. Gilbert (Univ. of Mississippi, University, MS 38677), and Richard D. Clark (Millersville Univ., Millersville, PA 17551-0302)

Daytime ground-to-ground acoustic propagation has been measured by a number of researchers over the years. Measurements of daytime sound levels as a function of height, however, have been made only in a limited number of near-ground experiments and apparently never at a considerable height. This paper reports measurements using a tethered balloon to measure the mean sound level and coherence from the ground up to 210 m at three separate ranges (300 m, 600 m, and 900 m) and at three frequencies (210 Hz, 380 Hz, and 600 Hz). A meteorological instrument package on the balloon measured the mean temperature, wind and humidity as a function of height. Fluctuation spectra were estimated using a stationary sonic anemometer at 5-m height which sampled the temperature and vector wind at 10 Hz. Acoustical inputs computed from the meteorological data were used with a three-dimensional parabolic equation model to predict the mean sound level and coherence as a function of height and range. The predictions and measurements agree reasonably well, both showing a clear shadow-zone boundary that is a function of the meteorological conditions. Descending across the shadow boundary, the predicted and measured mean sound levels and coherence decrease dramatically.

11:30

2aPA11. Propagation of a monochromatic sound wave in a turbulent atmosphere near the ground: Theory and laboratory experiment. Vladimir E. Ostashev[a)] (Environ. Technol. Lab., 325 Broadway, Boulder, CO 80303), Julie Wasier, Philippe Blanc-Benon, and Danile Juvé (Ecole Centrale de Lyon, BP 163, 69131 Ecully, France)

The theory of sound propagation in a turbulent atmosphere near the impedance ground is presented. It is assumed that the monochromatic source and receiver are located above the ground, and there is no temperature or wind velocity stratification in the atmosphere. A formula for the mean square sound pressure at the receiver is obtained and studied analytically and numerically as a function of different parameters of a problem: the variance and integral length scale of atmospheric inhomogeneities, the horizontal distance between the source and receiver, the value of impedance of the ground, etc. Theoretical results obtained are also compared with data from laboratory experiment modeling sound propagation in an atmosphere with temperature fluctuations near the ground. The experiment was carried out at Ecole Centrale de Lyon. The agreement between theoretical and experimental results is generally good. [This material is based upon work supported in part by the U.S. Army Research Office under Contract No. DAAG55-98-1-0463, and NRC-ETL Research Associateship.] [a)] On leave from New Mexico State Univ.

11:45

2aPA12. The effects of the wind velocity fluctuations on the sound backscattering cross section in the stratified moving atmosphere. Vladimir E. Ostashev (Environ. Technol. Lab., 325 Broadway, Boulder, CO 80303) and D. Keith Wilson (U.S. Army Res. Lab., Adelphi, MD 20783-1197)

Recent studies have shown that, in the case of strong diffraction, the relative contributions from wind velocity and temperature fluctuations to many statistical moments of a sound field propagating in a turbulent atmosphere are determined by the ratio of the normalized variances of wind velocity and temperature fluctuations, multiplied by factor 4. On the other hand, for weak diffraction, these relative contributions are determined by the ratio of the normalized structure–function parameters of wind velocity and temperature fluctuations, multiplied by factor 22/3. In this paper, we calculate the vertical profiles of both ratios for different turbulent regimes in unstable atmospheric boundary layers. Furthermore, vertical profiles of the ratio for weak diffraction are used to study the effects of the wind velocity fluctuations on the sound backscattering cross section in a stratified moving atmosphere. This study is important for acoustic remote sensing of the atmosphere by monostatic sodars which measure a signal proportional to the backscattering cross section. [Work supported in part by the U.S. Army Research Office under Contract No. DAAG55-98-1-0463, and an NRC-ETL Research Associateship.]

Session 2aPP

Psychological and Physiological Acoustics: Detection and Resolution

Nandini Iyer, Chair
Speech and Hearing Science, The Ohio State University, 110 Pressey Hall, 1070 Carmack Road, Columbus, Ohio 43210

Contributed Papers

9:00

2aPP1. Hearing discrete elements of an auditory temporal sequence: Evidence for a central mechanism. Dennis T. Ries, Jeffrey J. DiGiovanni, and Robert S. Schlauch (Dept. of Commun. Disord., Univ. of Minnesota, Minneapolis, MN 55455)

The ability to hear discrete elements of a temporal sequence was assessed. In one experiment, just-noticeable differences (jnds) in level between adjacent pulses of a 3- or 4-pulse temporal sequence that alternated in level was measured. The stimuli were pure tones (0.5-, 1.0- and 4.0-kHz tones), frozen noise, and white noise. All of the stimuli produced identical results. The jnd was about 5 dB for pulse rates less than 15 Hz and increased dramatically for higher rates. In a second experiment, listeners judged the initial direction change (high or low) of a frequency-modulated pure-tone sweep and a sinusoidal amplitude-modulated broadband noise. As in the first experiment, subjects performed poorly for rates higher than about 15 Hz. The similarity of the results for white noise, frozen noise, tones of different frequencies, AM noise, and FM sweeps is consistent with the notion that the factor-limiting performance is not in the auditory periphery. It is interesting to note that the approximately 15-Hz upper limit of performance on these tasks is consistent with the highest rates observed for phase locking in the auditory cortex. [Work supported by NIH-NIDCD R29 DC01542.]

9:15

2aPP2. Effects of masker harmonicity on masked threshold. William C. Treurniet and Darcy R. Boucher (Commun. Res. Ctr., P.O. Box 11490, Station ''H,'' Ottawa, ON K2H 8S2, Canada, bill.treurniet@crc.ca)

The role of masker harmonicity in auditory masking is not yet fully understood. This paper compares the effect of harmonic and inharmonic maskers on the simultaneous masked thresholds of narrow bandwidth noise probes using a three-alternative, forced-choice method. The harmonic masker had an 88-Hz fundamental and 45 consecutive partials (random starting phases). The inharmonic masker differed in that the partial frequencies were adjusted to nearby prime numbers. Maskers were presented at several levels, and the probe at several frequencies. Thresholds were consistently lower for the harmonic masker (4–8 dB). The masking release persisted when masker and probe bandwidths were equal, so it is not due to comodulated auditory filter outputs. The difference between maskers disappeared when the harmonic partials were separated by more than 176 Hz, suggesting that the effect may be related to critical bandwidth. Further, the threshold of noise probes located between two partials was always lower for the harmonic masker. However, the threshold for a tonal probe at the frequency of an omitted partial was higher with the harmonic masker than with the inharmonic masker. The results are consistent with a harmonic template model that proposes inhibition at the template frequencies and disinhibition between those frequencies.

9:30

2aPP3. Masking by harmonic complexes with different phase spectra in hearing-impaired listeners. Jennifer J. Lentz and Marjorie R. Leek (Army Audiol. and Speech Ctr., Walter Reed Army Medical Ctr., Washington, DC 20307)

At the last meeting, we showed that in normal-hearing listeners, masking effectiveness of harmonic complexes depends on the maskers' phase spectra. Such masking differences are thought to partially reflect an interaction between the input waveform and phase characteristics of the normal auditory system. We have extended that work to explore masking by harmonic complexes in listeners with hearing impairment. The masker was a harmonic complex with equal-amplitude components, with phases selected according to scaled modifications of the Schroeder algorithm. Masking was measured for several different signal frequencies and levels. In contrast to normal-hearing listeners, most hearing-impaired listeners showed little change in masking for different phase selections. Some listeners did demonstrate masking differences, but the greatest reduction in masking occurred when the components had nearly equal phases (i.e., a highly modulated masker waveform). Apparently, in those listeners, the within-channel phase characteristic did not substantially alter the phase structure of the input stimulus, as is thought to occur in normal ears. In aggregate, these data suggest that sensorineural hearing loss is accompanied by deficits in temporal resolution, possibly involving changes in the phase characteristics of auditory channels. [Work supported by NIH DC00626.]

9:45

2aPP4. Factors accounting for performance in a frequency-sample-discrimination task with distracters. Donna L. Neff, Eric C. Odgaard, Walt Jesteadt, and Huanping Dai (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, neff@boystown.org)

This paper presents results of fitting several linear models, including the CoRE model of Lutfi [R. A. Lutfi and K. A. Doherty, J. Acoust. Soc. Am. **96**, 3443–3450 (1994)] to data from a study of the interaction of target and distracter tones in a sample-discrimination task. In this 2IFC task, ten listeners judged which of two pairs of target tones were drawn from the higher of two overlapping frequency distributions. After training with targets alone, two distracter tones were presented simultaneously with the target tones, with each distracter placed in a frequency region remote from the target region. The important variables were the frequency regions and degree of frequency variability of targets versus distracters. Main features of the data included a large range of individual differences, a dominance of the effects of lower-frequency context tones over the higher-frequency context tones in many conditions, and differences in performance in the baseline, no-context conditions. When stimulus variability, perceptual weights across frequency, and performance in the no-context conditions are included as factors in the analysis, both mean and individual performance can be described reasonably well. [Work supported by NIDCD.]

10:00

2aPP5. Temporal factors in auditory peak detection of modulation of tones. Julius L. Goldstein (Dept. of Elec. Eng., Washington Univ., Campus Box 1127, St. Louis, MO 63130) and Joseph L. Hall (Bell Labs., Lucent Technologies, Murray Hill, NJ 07974)

Auditory detection of tone modulation has been modeled as envelope-peak detection of the cochlear filterbank responses [J. L. Goldstein, J. Acoust. Soc. Am. **41**, 458–479 (1967)]. To study the ability of this model to quantify temporal smoothing in detection, new psychophysical experiments using a 4IFC paradigm were conducted, with the authors as subjects. AM or quasi-FM sinusoidally modulated tones were discriminated from the carrier tone. Only one should was modulated among the four pushed sounds in the paradigm. The first series of exploratory experiments (1994) focused on low modulation frequencies (4–16 Hz). A second series of systematic experiments (1996) with a wide range of modulation frequencies allowed estimation of detector smoothing at carrier frequencies of 0.25, 1, and 4 kHz. Despite differences between subjects in estimated spectral filtering, modulation thresholds for both subjects gave similar estimates of smoothing at each carrier frequency. At modulation frequencies above 20 Hz, the smoothing function showed similar dependence on carrier frequencies as found from the early experiments (1967). Below 20 Hz, the early experiments, which used a method of adjustment paradigm, reveal a hyperacuity. This suggests a second slower detection mechanism that exploits repeated presentations of test sounds. [Work supported by NSF Grant IBN-9728383.]

10:15–10:30 **Break**

10:30

2aPP6. Statistical theory of peak detection for human hearing. Julius L. Goldstein (Dept. of Elec. Eng., Washington Univ., Campus Box 1127, St. Louis, MO 63130)

Envelope peaks and peak factors of narrow-band sounds have been found in simulation studies to explain a wide range of experiments on sound discrimination [e.g., J. L. Goldstein, J. Acoust. Soc. Am. **99**, 2541(A) (1996)]. To extend understanding of peak detection, mathematical approximations were developed for the first and second moments of log (dB) envelope statistics of a tone centered in noise with duration T and bandwidth W. Noises considered are: uniform periodic noise (UPN) having uniform amplitudes and random phases, Gaussian periodic noise (GPN) having random amplitudes and phases, and true Gaussian noise (TGN). Key properties quantified include: (1) Three nearly independent causes of waveform fluctuation underlie peak variance, viz., phase noise, energy noise, and interaction of tone with common frequency noise (energy cross term). (2) The energy variances in peak detection are similar as for energy detection. (3) Peak factor variance is primarily phase noise, which, for noise alone, is ~ 1 dB for $T \times W > 2$. (4) Peak and peak factor means differ by mean energy. (5) Mean peak growth for a small tone added to noise follows energy summation, while for a small noise added to a tone, the growth follows rms amplitude summation. [Work supported by NSF Grant IBN-9728383.]

10:45

2aPP7. Detectability of Gaussian noise by humans and simulated observers: A stimulus-level correlation analysis. Judi A. Lapsley Miller (Psychophys. Lab., Victoria Univ. of Wellington, New Zealand)

The correlation between ratings made by humans and simulated observers in detecting the same acoustic stimuli was used to estimate the detection (critical) bandwidth, and the form of rectification, temporal integration, and sampling strategy used by the human hearing system. By using small-$\mathcal{WT}$ Gaussian noise, bandwidth ($\mathcal{W}$) and duration ($\mathcal{T}$) were manipulated independently, resulting in 18 different combinations of bandwidth and duration for $\mathcal{WT}=1, 2$, and 4. The pattern of correlation, as properties of the simulated observer were systematically varied, indicated the best correlated observer. This tended to be the *full-linear* observer, consisting of a bandpass filter (which was generally wider than the bandwidth of the signal), full-wave rectifier, and a full (true) integrator. The full-linear observer was slightly, but consistently, better at describing human performance than the energy or envelope detector.

11:00

2aPP8. Effect of BM compression on masking period patterns (MPPs). Magdalena Wojtczak, Anna C. Schroder, Ying-Yee Kong, and David A. Nelson (Clinical Psychoacoust. Lab., Univ. of Minnesota, 516 Delaware St. S.E., Minneapolis, MN 55455)

MPPs were measured in listeners with normal and impaired hearing using tonal 4-Hz SAM maskers and short tonal probes with frequencies that were either identical to or higher than the carrier frequency of the masker. The probe frequencies were 500, 1200, and 3000 Hz for on-frequency masking, and 1200, 2400, and 6000 Hz for off-frequency masking. In normal-hearing listeners MPPs measured with off-frequency probes had valleys that were much longer and deeper than valleys observed with on-frequency probes. A similar result was observed in hearing-impaired listeners in the frequency region of mild hearing losses, where significant residual compression was presumably operating. However, in the frequency region with substantial hearing loss where compression is substantially reduced or absent, MPPs measured with the on- and off-frequency probes were very similar. A model consisting of peripheral filtering, compressive nonlinearity, and a sliding temporal window was used in an attempt to predict the data and to estimate the compression index. The results suggest that the similarity of on- and off-frequency temporal resolution in hearing-impaired listeners may be due in part to the lack of the compressive nonlinearity that is evident at the level of the basilar membrane in normal-hearing listeners. [Work supported by NIH-NIDCD DC00149.]

11:15

2aPP9. Estimation of psychometric functions using a hybrid up–down maximum likelihood procedure. He Yuan, Harry Levitt (Ctr. for Res. in Speech & Hearing Sci., City Univ. of New York, 33 W. 42nd St., New York, NY 10036), and James D. Miller (Central Inst. for the Deaf, St. Louis, MO 63110)

Rapid, efficient, and reliable estimation of psychometric functions is a much sought after goal in psychophysics. Adaptive techniques of the up–down type have many advantages in this regard, but they are not without their limitations. An alternative approach is the use of maximum likelihood estimation in placing observations. Whereas the maximum likelihood method is as efficient if not more efficient than any other method of estimation, this property only holds for very large samples. The use of maximum likelihood estimation in adaptive testing has significant limitations when the sample size is very small. In this study, a hybrid adaptive procedure was developed using an up–down technique initially and then converting to a maximum likelihood procedure once sufficient samples have been obtained for reliable estimation using the latter technique. The up–down technique that is used reduces the step size systematically employing a rule analogous to that used in analog-to-digial converters and is referred to as the AD procedure. Monte Carlo simulations of experiments using the hybrid adaptive technique showed promising results in comparison with other adaptive procedures. [Research supported by Grant 5P50DC00178 from NIDCD.]

Session 2aSA

Structural Acoustics and Vibration, Engineering Acoustics and Physical Acoustics: Acoustic Nondestructive Evaluation: New Directions and Techniques, Part I

Joseph W. Dickey, Chair
Center for Nondestructive Evaluation, Johns Hopkins University, Baltimore, Maryland 21218-2689

Chair's Introduction—9:25

Invited Papers

9:30

2aSA1. Advances in ultrasonic nondestructive evaluation. Robert E. Green, Jr. (Ctr. for Nondestruct. Eval., Johns Hopkins Univ., Baltimore, MD 21218)

The fundamental principles of linear elastic-wave propagation in isotropic and anisotropic materials emphasizing the modifications to ultrasonic wave propagation caused by anisotropy will be reviewed. Among the topics covered will be wave-propagation modes, energy-flux vector, diffraction-beam spreading, attenuation, and nonlinear effects. Verification of these principles by optical probing of ultrasonic-wave fields in transparent solids will be presented. Recent advances and practical applications of noncontact ultrasound, including pulsed lasers, optical interferometers, electromagnetic acoustic transducers (EMATs), and air-coupled systems will be described. Application of x-ray diffraction topography to verify the mechanical vibration modes of quartz crystals will be illustrated. Finally, suggestions will be made about possible future advances in ultrasonic nondestructive evaluation of materials and structures.

10:00

2aSA2. A nonlinear mesoscopic elastic class of materials. Paul A. Johnson (Nonlinear Mesocopic Elasticity, Los Alamos Seismic Res. Ctr., Los Alamos Natl. Lab., Los Alamos, NM 87545) and Robert Guyer (UMASS Amherst, Amherst, MA)

It is becoming clear that the elastic properties of rock are shared by numerous other materials (sand, soil, some ceramics, concrete, etc.). These materials have one or more of the following properties in common: strong nonlinearity, hysteresis in stress–strain relation, and discrete memory. Primarily, it is the material's compliance, the mesoscopic linkages between the rigid components, that give these materials their unusual elastic properties. It can be said that these materials have nonlinear mesoscopic elasticity and encompass a broad class of materials. Materials with nonlinear mesoscopic elasticity stand in contrast to liquids and crystalline solids whose elasticity is due to contributions of atomic level forces, i.e., materials with atomic elasticity. Atomic elastic materials are well described by the traditional (Landau) theory of elasticity; however, mesoscopic elastic materials are not. Mesoscopic materials are well described by the P-M (Preisach–Mayergoyz) model of nonlinear elasticity developed by Guyer and McCall. A sequence of experiments on numerous materials illustrate the evidence of nonlinear mesoscopic elastic behavior. In experimental analysis a surprising discovery was made: damaged atomic elastic materials behave as mesoscopic elastic materials. It is significant that the nonlinear mechanism(s) in mesoscopic elastic materials remains a mystery.

10:30–10:45 Break

10:45

2aSA3. Noncontact ultrasonics for materials nondestructive evaluation. Francesco Lanza di Scalea and Robert E. Green, Jr. (Ctr. for Nondestruct. Eval., Johns Hopkins Univ., Baltimore, MD 21218)

Recent advances in ultrasonic techniques for remote and noncontact nondestructive evaluation of structural materials will be presented. The principles and techniques of the laser generation and detection of ultrasonic waves in opaque solids will be reviewed. A laser-based beam-steering C-scan system will be described, along with a novel stabilization procedure for ultrasound detection by a confocal Fabry–Perot interferometer which appears to be more effective than previous approaches for high-sensitivity measurements in beam-steering scan applications. A hybrid system which is based on laser generation of narrow-band acoustic surface waves and their detection by gas (air) -coupled transduction will also be presented. This system uses fiberoptic light delivery which enhances its flexibility and remoteness from the test piece. The main issues governing the efficient coupling of high-power laser light into optical fibers will be discussed. An improved theory for the laser generation of narrow-band surface acoustic waves by the spatial modulation technique will be outlined and the effect of the number of illumination sources on the bandwidth of the generated wave will be assessed experimentally and theoretically. Finally, the suitability of the hybrid system for the intelligent control of the tow-placement process of advanced thermoplastic composites will be demonstrated.

Contributed Papers

11:15

2aSA4. Computational methods for an inverse problem in acoustics. Thomas DeLillo, Victor Isakov, Nicolas Valdivia, and Lianju Wang (Dept. of Mathematics and Statistics, Wichita State Univ., Wichita, KS 67260-0033)

The problem of computing normal velocities on the boundary of a region from pressure measurements on an interior surface is considered. The pressure satisfies the Helmholtz equation and is represented by a single layer potential. Once the density function is found, the normal velocities can be easily computed by applying a second kind of integral operator. The problem of solving for the density function from the interior pressure measurements is ill-posed. Regularization methods using the singular value decompostion and iterative methods, such as the conjugate gradient method for the normal equations, are compared for pressure data with noise. [Work supported by NSF and Cessna Aircraft Company.]

11:30

2aSA5. Remote ultrasonic classification of fluids in cylindrical containers by analyzing the response of circumferential guided waves. Gregory Kaduchak and Dipen N. Sinha (Los Alamos Natl. Lab., Electron. and Electrochemical Mater. and Devices Group, MS D429, Los Alamos, NM 87545)

A novel technique for classifying fluids in sealed, metal containers at large stand-off distances has been developed. It utilizes a recently constructed air-coupled acoustic array made from inexpensive, commercial-off-the-shelf components to excite the resonance vibrations of fluid-filled vessels. The sound field from the array is constructed by transmitting a high-frequency modulated carrier wave which utilizes the nonlinearity in the air medium to demodulate the carrier frequency along its propagation path in air. The array has a narrow beamwidth and an operating bandwidth of greater than 25 kHz. The vibrations are detected using a laser vibrometer in a monostatic configuration with the acoustic source. It is shown that the propagation characteristics of the ao Lamb wave are highly affected by different interior loading conditions on the interior wall of a cylindrical container. Classification of the interior fluid is obtained by analyzing the change of this response as a function of frequency. Experiments demonstrate that classification of the fluid-filler inside closed, steel vessels is possible with incident sound pressure levels of the demodulated wave as low as 80 dB at the container location. Preliminary experiments demonstrate that stand-off distances greater than 3 m are achievable for classification purposes.

11:45

2aSA6. Acoustical free oscillation NDT method of cracks for turbomachinery. Leonid M. Gelman, Olena V. Bezvesil'na, Sergey V. Gorpinich (Dept. of Nondestructive Testing, Natl. Tech. Univ. of Ukraine, 37, Peremogy pr., Kiev, 252056 Ukraine), and Aleksandr P. Strokov (GSKBD, Kharkov, Ukraine)

A new vibroacoustical nondestructive testing and evaluation method is proposed. The method uses phase spectral density harmonics of testing object free oscillations as features. New analytical expressions of phase spectral density of testing object free oscillations are received and considered.

TUESDAY MORNING, 2 NOVEMBER 1999 DELAWARE A & B ROOM, 9:00 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication: Cross-Linguistic Aspects of Speech Production and Perception (Poster Session)

James E. Flege, Chair

Department of Rehabilitation Sciences, University of Alabama, VH 503, Birmingham, Alabama 35294-0019

Contributed Papers

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

2aSC1. Searching for foreign accent. Z. S. Bond, Verna Stockmal, and Danny R. Moates (Ohio Univ., Athens, OH 45701-2979)

Native speakers of English can identify non-native speakers with relatively little difficulty. Further, they claim to be able to identify the native language of non-native speakers, at least as suggested by such descriptive terms as French accent or Arabic accent, implying that non-native English carries properties which are characteristic of native languages. In four experiments, we investigated whether English speakers could match an unknown foreign language with a foreign accent. In the first two experiments, listeners heard a sample of accented English and were asked to identify the native language of the speakers from a series of competitors. The target languages were Japanese and Latvian. Listener performance did not exceed chance. In the third experiment, listeners made ''yes–no'' responses to accented English matched with foreign languages, including the native language of the speaker. Although listeners thought some languages were more likely to be the source of the foreign accent than others, they did not identify the target language correctly. In the fourth experiment, listeners made ratings on a seven-point scale, judging the similarity of accented English and various foreign languages, with results very similar to those of experiment 3.

2aSC2. Effects of age and L1 use on native Italian speakers' perception of English vowels.
James E. Flege (Dept. of Rehabilitation Sci., VH503, Univ. of Alabama at Birmingham, Birmingham, AL 35294, jeflege@uab.edu) and Ian R. A. McKay (Univ. of Ottawa, Ottawa, ON K1N 6N5, Canada)

This study examined the categorial discrimination of pairs of Canadian English vowels (/i/–/ɪ/, /ɪ/–/ɛ/, /eɪ/–/ɛ/, /ɛ/–/æ/, /eɪ/–/ɪ/, /ɒ/–/ʌ/, /æ/–/ʌ/, /ɚ/–/ʌ/, /eɪ/–/i/) by groups of native Italian speakers who arrived in Canada at average ages of 7.6 and 19.5 years. Half of the 36 early and the 36 late bilinguals reported using Italian relatively seldom (*mean*=8%) or often (*mean*=48%). The subjects clicked ''1,'' ''2,'' or ''3'' to indicate the serial position of an odd item out, or ''no'' to indicate hearing three physically different instances of a single category. The discrimination (A′) scores were significantly higher for early rather than late bilinguals, and for bilinguals who used Italian seldom rather than often. The lack of a two-way interaction indicated that the L1 use effect was comparable for the early and late bilinguals. The ''Early-High Use'' group obtained lower scores than the native English comparison group for one constant (/ɒ/–/ʌ/) but the ''Early-Low Use'' group did not differ from the native English group for any contrast. The results suggest that both the state of development of the L1 phonetic system at the time L2 learning commences and language use influence the accuracy with which L2 vowels are perceived.

2aSC3. The effect of vocalic $F0$ on the identification of Korean stops.
Mi-Ryoung Kim (Prog. in Linguist., Univ. of Michigan, Ann Arbor, MI 48109-1285, kmrg@umich.edu)

This study investigates the contribution of vocalic $F0$ following the release of an initial stop to the identification of the three-way stop contrast (aspirated, lenis, and fortis) in Korean. Previous acoustic studies have shown an effect of initial stops on $F0$ of the following vowel. Using synthetic stimuli, perceptual studies [Han (1996); Cho (1996)] have shown that listeners use $F0$ in contrasting stops. However, the relative contribution of the consonantal and vocalic information remains unclear. To address this issue, a perceptual study was conducted using cross-spliced stimuli from three naturally produced syllables across all three places articulation in the /a/ context and for alveolars in the /i/ context. Stimuli were created by cross-splicing each consonantal portion (prior to voicing onset) with each vocalic portion (after voicing onset) to create all possible combinations. Twelve Korean listeners each responded to a total of 1444 tokens. The results showed that vocalic $F0$ is more important than the consonantal onset to stop identification, suggesting that Korean uses tonal variation to differentiate consonant types.

2aSC4. Using resynthesized speech in /r/–/l/ production and perception training.
Rieko Kubo, Reiko Akahane-Yamada, and Hideki Kawahara (ATR Human Information Processing Res. Labs., 2-2 Hikaridai, Seika-cho, Soraku-gun, Kyoto, 619-0288 Japan)

To find effective speech stimuli for second-language (L2) training, we generated resynthesized speech using the STRAIGHT algorithm developed by Kawahara (1997). First, Japanese speakers were trained to produce English /r/ and /l/ in a reproduction task. Training proceeded by first using model words which were artificially lengthened by STRAIGHT, and then gradually introducing words at a natural speed. Spectrograms of the model speech and the learner's speech were given as feedback. Subjects improved significantly from pretest to post-test in their production ability as evaluated by human judges. Second, resynthesized speech was used in perception training. Previous perceptual training studies using synthetic speech have not always been successful. However, training based on resynthesized speech from STRAIGHT led to the same amount of improvement as training based on the original natural tokens, suggesting the potential for using STRAIGHT in L2 perception training. We then manipulated certain acoustic parameters of the stimuli, namely, lengthening and/or enhancement of the third formant. The training effect using these stimuli did not differ significantly from the effect found with the unmodified synthetic stimuli. These results suggest that resynthesized speech from STRAIGHT is suitable for L2 training, but further effort is necessary to find more effective stimulus manipulations.

2aSC5. Listeners' representations of within-word structure of native and nonnative words by Japanese speakers.
Takashi Otake (Dokkyo Univ., 1-1, Gakuen-cho, Soka, Saitama, 340-0042 Japan) and Kiyoko Yoneyama (Ohio State Univ., Columbus, OH 43210-1298)

This study further explored how Japanese represented within-word structure of native and nonnative spoken words in terms of two phonological units, morae and syllables, using a new task. The main question addressed in this study was to investigate which phonological unit was preferred to represent nonnative and native words. Three experiments were conducted with Japanese college students, using a tapping task. In experiment 1, 42 syllable Japanese materials which included a CVN syllable in the first syllable and fillers were presented to 20 Japanese college students to identify the number of chunks. The result showed that they preferred morae to syllables. In experiments 2 and 3, 42 syllable English and Spanish materials which included exactly the same syllable structure and fillers were presented to the same number of students, respectively. The results showed that the preference was around 50% each, respectively. These results obviously contradicted to the previous findings in which morae were preferred to nonnative words. The results in this study may suggest that the previous findings using an explicit segmentation task may have been influenced by orthography and that Japanese may be able to suppress morae to represent nonnative words. [Work supported by IFC and Ministry of Education.]

2aSC6. The effect of duration on the perception of Cantonese level tones.
Patrick C. M. Wong and Randy L. Diehl (Dept. of Psych., Univ. of Texas, Austin, TX 78712)

An inverse relationship has been claimed to exist between the pitch of phonemic tones and the duration of the tone-bearing syllables [e.g., Blicher *et al.*, J. Phonetics **18**, 37–49 (1990)]. If this generalization is correct, syllable duration might be expected to serve as a perceptual cue to tone category. In this study, Cantonese-speaking listeners identified tones in sentence context. A naturally produced Cantonese sentence (/ha6 yat1 go3 zi3 hai6 si3/ ''The next word is to try'') was resynthesized in 25 versions such that the fundamental frequency ($F0$) contour of the context (the first five syllables) was shifted upwards, 1%, 3%, 6%, 9%, or 12% relative to the original, and the final, target syllable was resynthesized with durations that were 80%, 90%, 101%, 110%, or 120% of the original. We predicted that the higher context F0s and longer target syllable durations would yield more low-tone identification responses. The results of an analysis of variance showed a significant main effect of contextual F0 in the expected direction, but no significant effect of target syllable duration. This last result calls into question the generality of the assumed perceptual relationship between duration and tone category. [Work supported by NIDCD.]

2aSC7. Effect of linguistic background on accuracy of discrimination task and reaction time.
Aman Kumar (Prog. in Linguist., Univ. of Michigan, 1076 Frieze Bldg., Ann Arbor, MI 48109-1285)

This study examines how American English (AE) speakers categorize Hindi laryngeal and place of articulation contrasts. These results are interpreted with respect to the Perceptual Assimilation Model (PAM) [Best, ''A direct realist view of cross-language speech perception,'' in *Speech Perception and Linguistic Experience: Issues in Cross Language Research* (York Press, Inc., 1995), pp. 171–204]. Accuracy and reaction time data were obtained for discrimination and identification of Hindi bilabial and dental phonation contrasts and the retroflex-dental place contrast. Subjects included both AE speakers and a control Hindi speaker group. Discrimination accuracy averaged across the subjects for the phonation contrasts was nativelike while accuracy for the retroflex-dental contrast was significantly lower but above chance. Overall, the reaction time was inversely proportional to discrimination accuracy. For some subjects discrimination performance for the retroflex-dental contrast was nativelike. However, the reaction time for those subjects was significantly higher than that of other speakers. In general, the reaction time was higher for the retroflex-dental contrast than for the other contrasts.

2aSC8. Perception of English sentence stress by Yarmouk University English majors. Fares Mitleb, Fawwaz Al-Haq, and Rasheed Al-Jarrah (Speech and Hearing Ctr., Yarmouk Univ., Irbed 21163, Jordan)

Perception of English sentence stress at discourse level was the focus of the present paper. Twenty English majors, freshmen and seniors at Yarmouk University, voluntarily took part in the study. In the scope of this paper, it was assumed that the subject would give diverse responses on the test material which comprised 13 English utterances, each of which was produced with the correct stress pattern by native speakers of American English. The subject's dependence on their native language perceptual pattern (i.e., Arabic in this case) made them fail to take advantage of the auditory correlates of stress such as pitch, duration, and loudness in the most appropriate way. The subjects' task was to mark out the constituent made most prominent by the native speaker in each utterance. Their responses were then classified as normal (if stress was correctly perceived), contrastive (if it was perceived on a different constituent), or undecided (if the utterance was perceived unstressed altogether or if two constituents or more were stressed simultaneously). It turned out that the subjects sometimes perceived stress on constituents that are hardly stressed such as pronouns; determiners and prepositions made it clear that the subjects' mastery of the phenomenon under discussion was inadequate.

2aSC9. Intelligibility of Arabs' production of English grammatical words. Fares Mitleb (Speech and Hearing Ctr., Yarmouk Univ., Irbid, Jordan)

Features of connected speech such as the use of weak forms of grammatical words in English are said to have an important role in the communication process between native and nonnative speakers. The present study is intended to describe how intelligible is the production of certain English grammatical words for British English listeners by Arabs. Six Arabs learning English and four British speakers produced 15 sentences with 17 grammatical words for the purpose of this study. Four native British listeners were asked to judge the stimuli they heard of certain grammatical words as either having a ''strong'' or ''weak'' form. They reported that Arabs' production of strong forms was almost the same as that of native subjects. However, the Arab subjects failed to produce a high percentage of intelligible weak forms. Findings of this impressionistic study were then examined against the well-established approaches to the problem of foreign-accented speech. Our results seem to support claims about the interference of segmental knowledge onto features of connected speech since the first is generally given too much emphasis in the teaching of the target language to the extent that the latter is most neglected. [Research supported by Yarmouk University.]

2aSC10. Training Chinese speakers to perceive English /nl/. Anna M Schmidt, Amy Kaminski, and Hyunjoo Chung (School of Speech Pathol. and Audiol., Kent State Univ. P.O. Box 5190, Kent, OH 44242, aschmidt@kent.edu)

Native speakers of some southern Chinese dialects have difficulty perceptually differentiating English /n l /. In this study, five native Chinese subjects participated in a training experiment designed to increase perception of the /n/-/l/ contrast using a fading technique [D. Jamieson and D. Morosan, Percept. Psychophys. **40**, 205–215 (1986)]. Stimuli consisted of initial /n/ and /l/ words produced by four native English speakers in which the initial consonants were increased by varying amounts. Subjects were first trained on the longest exemplars, and progressed to the natural versions. Both discrimination and identification training sets were used along with immediate feedback. Pre- and post-training perception generalization and production data were obtained. Results and implications will be discussed.

2aSC11. The discrimination of Mandarin Chinese alveolo-palatal fricative and affricate contrasts by American English and Mandarin Chinese speakers. Feng-Ming Tsao, Huei-Mei Liu, and Patricia K. Kuhl (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA 98195)

This study examined how language experience affects the perceptual sensitivity of adult speakers to nonnative phonetic contrasts. Three of Mandarin contrasts (two alveolo-palatal fricative versus affricate contrasts and one alveolo-palatal affricate aspirated versus unaspirated contrast) were computer synthesized. For each of these three contrasts, three different variations were contrasted by varying either amplitude rise time or fricative noise duration. Both English and Mandarin speakers participated in a speech discrimination task using these stimuli. When compared to the performance of Mandarin speakers, English speakers showed poorer performance for each stimulus pair in terms of accuracy and sensitivity measures (d'). However, both English and Mandarin speakers demonstrated similar difficulty when tested with three stimulus pairs that were not typical exemplars of Mandarin phonetic categories. The results show the effect of language experience on the discrimination of phonetic contrasts. [Research supported by NIH grant to P. K. Kuhl.]

2aSC12. Cross-language speech intelligibility in noise: The comparison on the aspect of language dominance. Yasue Uchida (Dept. of Otorhinolaryngology, Nagoya Univ. School of Medicine, 65 Tsurumai-cho, Showa-ku, Nagoya, 466-8550 Japan, yasueu@nils.go.jp), David J. Lilly (Veteran Administration Natl. Ctr. for Rehabilitative Auditory Res., Portland, OR), and Mary B. Meikle (Oregon Hearing Res. Ctr., Portland, OR)

The purpose of this study is to obtain information about the influence of language characteristics on the results of speech intelligibility in two different languages: English and Japanese. This study investigates the speech intelligibility of both English and Japanese under quiet and noisy situation on 14 bilingual subjects aged 23 to 42 years with normal hearing. As test materials, the CID W-22 word lists which are meaningful monosyllables are used for English and the 57-S word lists which are nonsense monosyllables are used for Japanese. The subjects whose dominant language is English are 6. Results show that percentage-correct of speech intelligibility is higher in Japanese than in English at lower intensity and noisy situation. This advantage in Japanese is seen regardless of the language dominance. It is considered that the phonetic characteristics and the simple structure of Japanese can make it relatively resistant to the hard listening condition. [Work supported in part by Japan Foundation for Aging and Health.]

2aSC13. Effect of syllable structure on syllable counting in English by Japanese. Tsuneo Yamada, Donna Erickson (National Inst. of Multimedia Education, 2-12 Wakaba, Mihama, Chiba, 261-0014 Japan), and Keiichi Tajima (ATR Human Information Processing Res. Labs., 2-2 Hikaridai, Seika-cho, Soraku-gun, Kyoto, 619-0288 Japan)

A previous study [Erickson, Akahane-Yamada, Tajima, and Matsumoto, Proceedings of ICPhS (1999)] showed that Japanese listeners have difficulty correctly counting the number of syllables in spoken English words, presumably because Japanese syllable structure is more restricted than English. A complex syllable like ''stress'' is not allowed in Japanese, and tends to get mapped onto many syllables, such as ''su.to.re.su.'' To

test what factors contribute to Japanese listeners' perception of syllables, a set of nonsense words varying in number and voicing of initial and final consonants, vowel type, and number of syllables, was presented auditorily to Japanese and American listeners. The task was to count the syllables in the words. American speakers counted the syllables with nearly 100% accuracy; Japanese listeners performed at about 50% accuracy. The number of initial consonants seems to have a stronger negative effect on the syllable counting performance by Japanese listeners than the number of final consonants, or vowel type. These findings suggest that onsets may play a stronger role than codas in Japanese listeners' perception of the psychological weight of syllables, contrary to current notions of phonological weight which are sensitive to codas and ignore onsets. Implications for second-language speech perception/production learning will be discussed.

2aSC14. A computational analysis of uniqueness points for Japanese auditory word recognition. Kiyoko Yoneyama and Keith Johnson (Dept. of Linguist., Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, yoneyama@ling.ohio-state.edu)

This paper discusses uniqueness points for all words in a 65 000-word computerized lexicon in Japanese. The uniqueness point is the point in a left-to-right scan where the word is distinguished from all other words in the lexicon. This is an important concept in the Cohort Theory of auditory word recognition. Four different analyses were conducted based on two variables: the processing unit (segment or mora) and the phonetically transcribed representation (with or without pitch information). The probability of the uniqueness point being prior to the end of the word was highest when pitch information was included and/or when words were represented in segments rather than in moras. Still, in the best combination of variables (segments with pitch information) only 48% of the words reached a uniqueness point before the end of the word. This study predicts that if Japanese listeners use small processing units and attend to pitch, recognition performance will be maximized, but for most words this optimal processing strategy will still result in a Cohort size greater than one at word's end.

2aSC15. Correlation between vowel production and perception for a given speaker. Jean-Marie Hombert (Lab. Dynamique du Langage, CNRS and Universite Lyon 2, Institut des Sci. de l'Homme, 14 avenue Berthelot, 69363 Lyon cedex 07, France, hombert@univ-lyon2.fr) and Rene Carre (ENST-CNRS, 75634 Paris cedex 13, France)

The goal of this paper is to investigate the correlation between production and perception of 10 French oral vowels at the individual level. Acoustic data ($F1$, $F2$ and $F3$) from 10 French native speakers were collected. Measurements were obtained from the mid-point of the vowels using cepstral analysis. Each vowel was represented by 10 measurements. The same 10 speakers were subjected to perceptual experiments in which 53 synthetic vowel stimuli covering the entire vowel space were presented through headphones. The subject's task was to identify which French word from a list of 10 words (from the same list used in the production experiment) representing the 10 French oral vowels. Each synthetic stimulus was presented 10 times in randomized order. Subject responses were mapped in a $F1$-$F2$ space with groupings corresponding to 90% of identical labeling. These perceptual results were compared with the measurements obtained from the production data. Our results show that the target values in the perceptual domain are correlated for a given subject with this production data. This conclusion is extremely important for understanding how individual variations within the same Linguistic Community can be the source of sound change. [Work supported by CNRS-GDR Cognition and Linguistic Diversity.]

2aSC16. Evidence for the independence of speech categories in perception and production. Rachel Hemphill (Dept. of Linguist., Univ. of Chicago, 1010 E. 59th St., Chicago, IL 60637, rmhemphi@midway.uchicago.edu)

Models of speech processing, which attempt to account for speech input and output in phonological or categorical terms, generally rely on one single underlying representation for both perception and production. Previous research by Hemphill [J. Acoust. Soc. Am. **104**, 1757–1758 (1998)] has suggested that two separate representations are necessary to account for asymmetries in acquisition data. Sixteen native speakers of English were trained to categorically perceive a three way voicing contrast in Thai bilabial stop consonants. The results of perception and production tests following training indicate that subjects were unable to accurately produce the contrasts that they had learned from perceptual training, despite the fact that they could identify the perceptual stimuli correctly. Additional perception testing after a two-month interval found that subjects were still able to label the training stimuli accurately, but could not identify their own productions with greater accuracy than they identified the training stimuli or the productions of unfamiliar talkers. These results are explained in terms of separate perception and production representations in the mind and an hypothesis about the reorganization of perceptual categories is explored.

2aSC17. Branching and phrasing in Seoul Korean. Minkyung Lee (Dept. of Linguist., Indiana Univ., Bloomington, IN 47401)

This paper examines the relationship between the structural composition, the phrasing and duration and pitch range in Korean utterances. Previous linguistic models (Selkirk, 1986) predict that the syntactically left branching (LB) and right branching (RB) structures should correspond to different phrasings, which according to previous phonological models (Jun, 1993, 1995) should be indicated by the presence of rises in fundamental frequency. Speakers were instructed to produce syntactically ambiguous phrases in a frame which highlighted the ambiguity. The experiment indicates that LB and RB cases were collapsed together resulting in the same phrasing. Even though LB and RB structures usually exhibited the same phrasing, speakers did show differences in pitch range and duration which distinguished left from right branching structures. Initial words in RB cases were longer and had a relatively larger pitch range than RB cases. Such effects have also been found as a correlate of focus. Thus speakers tend to express structural properties of utterances by means of phonetic differences in relative peak $F0$ height and duration, apart from differences in phrasing.

2aSC18. Tonal alignment in Seoul Korean. Byung-Jin Lim and Kenneth de Jong (Dept. of Linguist., Indiana Univ., Bloomington, IN 47405)

Bruce found that a local $F0$ peak is aligned very precisely in time with the segmental material in Swedish [Swedish word accents in sentence perspective (1977)]. Alternatively, it is possible that such $F0$ peaks may function as edge markers, and hence not necessarily be aligned with any particular aspect of the word-internal structure. This study investigates how an initial high tone is aligned in time with the segmental material in Standard Korean. Recordings of two speakers of Seoul Korean producing three-syllable words with various syllable structure combinations in two prosodic conditions were digitized and analyzed. One speaker shows no apparent pattern of alignment with the segmental material; peaks are simply reached at a fixed duration from the beginning of the utterance regardless of the structure of the word. Another speaker, however, shows that initial high tones fall into two distinct groups, ones aligned with initial syllables and ones aligned with second syllables. This syllable association is statistically related to syllable composition. These results suggest that Korean tone alignment is currently in a state of fluctuation between durationally fixed edge tones and tones associated with internal syllables according to a stochastic rule.

2aSC19. Articulatory effects of contrastive emphasis on the Accentual Phrase in French. Helene Loevenbruck (Institut de la Commun. Parlee, INPG, Univ. Stendhal, UPRESA CNRS 5009, 46 av. F. Viallet, F38031 Grenoble, France)

Recent works (Beckman, 1996) show that prosody is itself a complex linguistic structure, and it is imperative to better describe its phonological and phonetic (acoustic and articulatory) characteristics. Articulatory studies of French prosody provide variable conclusions. The irregularities could come from the fact that prosodic structure is rarely considered and that different phenomena (''accents primaires,'' ''secondaires'') are examined together. Articulatory correlates of a prosodic entity, the Accentual Phrase (AP), are studied here, using a model of French prosody (Fougeron and Jun, 1998). The AP features an initial high tone Hi, also called ''accent secondaire,'' a final high tone H* (''primaire''), and two low L tones preceding them. Sentences containing four-syllable words (APs), were recorded for two French speakers (one male, one female), using EMA. The position of the AP in the sentence varied, several speaking conditions were elicited. Displacement, peak velocity, and movement duration are analyzed for the tongue-middle vertical position. The results suggest that LHi could be related to hyper-articulation of the first or second syllable, and LH* to even stronger hyper-articulation of the last syllable. With contrastive emphasis on the AP, the initial hyper-articulation can become as strong as, and even stronger than, the final one.

2aSC20. Acoustics of Spanish vowels as spoken in two regions of Colombia. Dolly Urueta Mazzilli, Ruth Bahr (Dept. of Commun. Sci. & Disord., 4202 E. Fowler Ave., BEH 255, Univ. of South Florida, Tampa, FL 33620, urueta-m@luna.cas.usf.edu), and Winifred Strange (City Univ. of New York, New York, NY)

It has long been believed that Spanish dialects do not differ with respect to vowels. However, acoustic differences between Spanish dialects have been reported anecdotally. A reasonable assumption then is that, like English, Spanish dialects would vary both within and across countries. In fact, dialect differences have been found within regions of Panama [M. C. McNair, ''An acoustical analysis of Panamanian vowels,'' unpublished Master's Thesis (1996), University of South Florida]. Therefore, the following analysis was conducted to determine if acoustic differences exist within Colombian Spanish. Nine male monolingual Spanish speakers from two cities of Colombia (Barranquilla and Santa Fe Bogota) produced three tokens of each of the five Spanish vowels in bilabial, alveolar, and velar environments within the frame, ''Escribe CVCV bien.'' Spectral and temporal measurements were calculated for each vowel in the first syllable of the word. A repeated measures ANOVA revealed a significant four way interaction suggesting that differences between dialects were dependent upon specific contexts, vowels, and formants. While vowel differences between dialects were subtle, perceptual differences in prosody and consonant production emerged.

2aSC21. Taiwanese final stops and following initial voiced stops and nasals. Ho-hsien Pan (Dept. of Foreign Lang. and Lit., Natl. Chiao Tung Univ., 1001 Ta-hsueh Rd., Hsinchu, Taiwan 30050)

The places of articulation of unreleased Taiwanese final stops, e.g., /p,t,k/ and glottal stop, have been known to assimilate into that of initial voiceless stops in following a syllable, depending on the speed of articulation (Peng, 1997). Besides assimilation in place of articulation, nasalization can be observed across syllables between voiced stops and nasals (Pan, 1999). However, little is known about the assimilation in place of articulation, voicing, and nasalization between Taiwanese final stops, and following voiced stops and nasals across different prosodic boundaries. This study uses EPG, airflow, and acoustical data to investigate the coarticulation between Taiwanese final stops, and initial nasals, initial voiced stops, across syllable boundary, morpheme boundary, phrase boundary of narrow focus, and intonation boundary. Preliminary results showed that Taiwanese final stops are lost in certain contexts. There is assimilation in place of articulation between unreleased final stops and initial voiced stops and nasals. No progressive nasalization is observed between final stops and initial nasals. Taiwanese final stops are easily coarticulated with following segments. Though the presence of final stops and short syllable duration are both vital cues to Taiwanese entering tones syllables, duration is a more invariant cue than the presence of final stops. [Project supported by NSC 87-2411-H-009-008.]

2aSC22. An acoustic analysis of Cantonese rising tones. Connie So (Dept. of Linguist., Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada)

The two Cantonese rising tones (the High Rising and the Low Rising) are traditionally described as Tone 35 and Tone 23. Recent studies, however, show that the High Rising tone has changed from Tone 35 to Tone 25. It starts at a pitch level similar to that of the Low Rising tone. The High Rising tone is also claimed to have a long and sharp rise from the midway of its fundamental frequency pattern. The present study attempts to look for these phenomena and to describe their characteristics in the two rising tones. Native Hong Kong Cantonese speakers were asked to read two sets of Cantonese rising-tone words formed of root-words /si/ and /fu/. Fundamental frequency patterns and duration of both rising tones in these two root-words are compared and discussed.

2aSC23. Overlap at the interface: High vowel devoicing in Japanese. J. Kevin Varden (Dept. of English, Meiji Gakuin Univ., Minato-ku, Tokyo, 108-8636 Japan)

High vowel devoicing (HVD) in Japanese has been characterized as both phonological [J. D. McCawley, *The Phonological Component of a Grammar of Japanese* (1968)], phonetic due to gestural overlap [S.-A. Jun and M. E. Beckman, LSA Paper (1993)], and as a result of overall vowel reduction [M. Kondo, Ph.D. thesis, University of Edinburgh (1997)]. Electromyographical and laryngeal spread data from one speaker [A. Tsuchida, Ph.D. dissertation, Cornell University (1997)] supports the existence of both phonological and phonetic devoicing in complementary environments, while data from ten speakers [J. K. Varden, Ph.D. dissertation, University of Washington (1998)] indicates the presence of both types of devoicing in overlapping environments. Results of Varden (1998) also indicate high-speaker variability in rate of devoicing, duration of voicing for voiced vowels, and context-dependency. A synthesis of these studies supports a characterization of HVD as the result of varying processes, each having the same acoustic end: the devoicing and overall reduction of high vowels in devoicing environments. The data in Varden (1998) also indicate a high incidence of devoicing of /u/ in the mora [tsu] before /n/ for some speakers. This indicates relaxation of the requirement of a following underlyingly voiceless segment.

2aSC24. Analysis of voice onset time in Georgian. Tamra M. Wysocki (Linguist. Dept., Univ. of Chicago, 1010 E. 59th St., Chicago, IL 60637, t-wysocki@uchicago.edu)

The Georgian stop system has a three-way opposition: voiced, voiceless aspirated, and voiceless ejective. This paper reports on results from an analysis of voice onset time (VOT) in word-initial stop consonants preceding /a/. Two native speakers of Georgian (one male, one female) were recorded reading two randomized token lists: one containing nonsense syllables (e.g., ''ba'') and one containing actual Georgian words (e.g., ''balaxi'' ''grass''). Tokens were repeated twice. VOT values were measured using spectrograms and waveforms. Results show that VOT clearly distinguishes two categories in Georgian: voiced and voiceless. Initial voiced stops had short positive VOTs, while the initial voiceless stops had relatively long positive VOT values. Within the voiceless category, however, mean VOT values for ejectives and aspirates overlapped. Articulatory and acoustic characteristics of ejectives and aspirates are discussed in relation to other possible cues for distinguishing between the two types of stops. In addition to stop categories, the results show a trend toward an-

2a TUE. AM

other distinction: place of articulation. Labials had the lowest VOT values, followed by coronals, and then velars. The results from these analyses are compared to studies of voice onset time in other languages.

2aSC25. A cross-gender examination of the breathy tone in Green Mong. Jean E. Andruski (Audiol. and Speech-Lang. Pathol., Wayne State Univ., Detroit, MI 48202) and Martha Ratliff (Wayne State Univ., Detroit, MI 48202)

Tone is generally thought of as a pitch difference that carries information regarding word meaning, but tone often correlates with phonation type as well as pitch. This paper presents the acoustic portion of a cross-gender study on acoustic and perceptual correlates of the breathy tone in Green Mong. Three male and three female native speakers of Green Mong residing in the Detroit area were asked to produce a series of target words containing the breathy tone and comparison tones in sentence context. Target words were minimal sets in which only tone varied. Each speaker produced a sequence of minimal sets containing each possible voiceless stop and voiced fricative onset of Green Mong. Vowel identity also varied across minimal sets. Four acoustic correlates of tone (fundamental frequency or $F0$, $F0$ contour, duration, and breathiness) were measured in each target word. Degree of breathiness was measured by subtracting the amplitude of $F0$ from the amplitude of the second harmonic at the durational midpoint of the vowel. Each of these four acoustic correlates will be examined in the breathy tone and the comparison tones, and a cross-gender comparison of the results will be presented.

2aSC26. Gender differences in fundamental frequency in focused words: A case from Japanese. Kyoko Nagao (Dept. of Linguist., Indiana Univ., Bloomington, IN 47401, knagao@indiana.edu)

When a word is focused in some context in Japanese, its fundamental frequency ($f0$) range is usually increased [Pierrehumbert and Beckman (1988)]. It is also well known that Japanese females tend to employ higher frequencies when they speak in Japanese than Caucasian females do [Yamazawa and Hollien (1990)]. However, previous studies have not examined the interactions between these two factors, focus and gender. The present study was done to determine whether the same strategy for realizing focus is employed by female and male speakers, in terms of the $f0$ and temporal patterns. From the analysis of 252 utterances produced by six Japanese speakers, almost the same amount of increase in absolute $f0$ values was observed in the speech of both sexes. Therefore, when viewed in proportion to the speakers' normal $f0$ range, females showed smaller increases in the proportional $f0$ values on the focused words than males did. The results suggest that Japanese females have less room for focus expansion because they normally use higher frequencies within their pitch range. Since temporal focus effects do not occur in female speech, it is considered that the perception of intended words as emphasized will be more ambiguous in female than in male speech.

2aSC27. Speaking fundamental frequency of young adult Arabic men during oral reading and spontaneous speaking in both Arabic and English languages. Ali Abu-Al-Makarem and Linda Petrosino (Dept. of Commun. Disord., Bowling Green State Univ., 200 Health Ctr., Bowling Green, OH 43403)

Currently, there is a paucity of normative data on speech and voice characteristics of different linguistic and ethnic groups. In particular, there is no available published data on the speaking fundamental frequency (SFF) characteristics of the voice of the Arabic population. The purpose of this study was to obtain preliminary data on the SFF of a group of normal speaking, young adult Arabic males. Fifteen, native Arabic, adult men served as subjects and received the identical experimental treatment. Four speech samples were collected from each subject (Arabic reading, Arabic spontaneous speech, English reading, and English spontaneous speech). Results showed the mean and SD of SFF are ([146,9, 15.4], [145.8, 13.8], [149.1, 12.6], and [145.5, 12.0]), respectively. No significant differences were found in the mean SFF between language and type of speech, nor between languages. A significant difference in the mean SFF was found between the type of speech ($F1{,}14=5.51$, $p=0.03$). Reading was significantly higher than speech. Also, Arabic men in this study had higher SFF values than previously reported for young adult males of other ethnic groups [H. Hollien and B. Jackson, J. Phonetics 117–120 (1973); A. Hudson and A. Holbrook, J. Speech Hear. Res. 197–201 (1992)]. For Speech Communication Best Student Paper Award.

2aSC28. Effect of computer-assisted training on production of English /r/ and /l/ by Japanese. Reiko Akahane-Yamada, Erik McDermott, Takahiro Adachi, and Tomoko Takada (ATR Human Information Processing Res. Labs., 2-2 Hikaridai, Seika-cho, Soraku-gun, Kyoto, 619-0288 Japan)

Japanese speakers were trained to produce American English (AE) /r/ and /l/ using a computer-assisted learning system which was developed to investigate how to provide useful and effective feedback to second-language learners regarding the goodness of their production in an automatic way. In experiment 1, two groups of Japanese speakers were trained to produce /r/ and /l/ in a reproduction task. In one group, spectrographic representations with formant-tracking results overlayed were used as feedback, and in another group, acoustic scores produced by an HMM-based speech recognition system were used as feedback. Learners in both groups significantly improved from pretest to post-test in their production ability as evaluated by AE judges. Experiment 2 investigated the order effect of production training and perception training. One group of trainees received production training before perception training, and a different group received training in the opposite order. Both groups improved in their production ability by an equal amount from pretest to post-test. However, the production-to-perception training group improved more in perception ability than did the perception-to-production training group, suggesting the relevance of training order in perception ability.

2aSC29. Speaking rate, fluency, and accentedness in monolingual English and bilingual Czech, French, and Japanese speakers. Cliff S. Burgess (Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada)

Speaking rates (SRs) and articulation rates (ARs) were calculated for monolingual English speakers and advanced ESL learners with Czech, French, and Japanese L1s; 64 speakers in total. Data were based on two reading tasks, a cartoon description, and a personal anecdote. Bilingual speakers did these tasks in both English and their respective L1s. Short excerpts were then taken from each English language sample (264 tokens plus 64 repeats) and played to 10 native English speakers who made scalar judgments on fluency and degree of accentedness. Judgments showed a wide range of correlation with SRs and ARs that varied among language group and task type. The influence of pause type, length, and frequency was also examined. The results suggest that cues to accent and fluency perception are multidetermined and that listeners weight these cues differentially. [Work supported by SSHRC.]

2aSC30. Phonological representation in Japanese and American rap music: Further evidence for mora timing. Atsuko Iwai (Dokkyo Univ., 1-1, Gakuen-cho, Soka, Saitama, 340-0042 Japan) and Takashi Otake (Dokkyo Univ., Saitama, 340-0042 Japan)

This paper investigated how words in Japanese and American rap music were represented in terms of two phonological units, morae and syllables. The main question addressed in this study was to find out which unit was preferred to represent words in Japanese and American rap music. One hundred CVN syllables in two or multi-syllable words were identified from 9 pieces of 2 Japanese musical bands and 27 pieces of 3 American musical bands. Then, the first author set these syllables to music one by one, using a Japanese computer music software to determine how these syllables were assigned to notes. The analysis showed that 73% of the CVN syllables in Japanese rap music were assigned to a note which was two times longer than that of the following note and 27% were assigned to two separate equivalent notes, while 93% of the CVN syllables in American rap music were assigned to a single note which was equivalent to the following note. In other words, the computation of duration was considered to choose notes in the former, while it was not in the latter, suggesting that words in Japanese and American rap music were represented in terms of morae and syllables, respectively.

2aSC31. Focus on Japanese-accented English interrogatives and declaratives. Laura G. Knudsen (Dept. of Linguist., Indiana Univ., Memorial Hall 317, Bloomington, IN 47405, lwright@indiana.edu)

This paper presents the results of a study of focus realization in Japanese English intonation, offering a modified and expanded replication of an earlier study [M. Ueyama and S.-A. Jun, UCLA Phonet. Lab Wkg. Paper No. 94 (1996)]. Fourteen subjects and three controls read two sets of test interrogatives and declaratives, one with fixed utterance length but focus in varying locations, and one with fixed focus location but varying utterance length after focus. F0 and absolute time measurements were taken at four or more points in each utterance, yielding measurements of focus contour, high- or low-plateau length, and utterance-final contour. Each test item was also analyzed qualitatively and coded phonologically using the ToBI transcription system. The current study includes twelve female and male EFL subjects in four levels, plus two more advanced ESL subjects in a fifth level. Results showed first of all that Japanese acquirers of English had large gaps in their inventory of intonational types. More advanced speakers did show evidence of having acquired aspects of the English intonational contours; however, not employing them as do native English speakers.

2aSC32. The effect of the Lombard reflex on speech produced by Cantonese speakers of English. Herman Chi Nin Li (Dept. of Linguist., Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, clia@sfu.ca)

The Lombard reflex is a phenomenon according to which, in the presence of noise, people unconsciously raise their vocal levels in order to compensate for the level of the ambient noise. It is well documented that speech produced in noisy environments exhibits increases in duration, fundamental frequency, and amplitude. Most studies associated with the Lombard reflex have been carried out with native speakers, but research with second language speakers is much less common. The present study describes a pilot project that aims at studying the effect of cafeteria noise on the production of native English and Cantonese-accented speech. Participants read aloud to the experimenter a set of simple true/false English statements under both no-noise and noise conditions. In the latter, masking noise of 70 dB SPL was fed through headphones to the speaker to elicit the Lombard speech. The characteristics of the sentence-length utterances produced under quiet and noise conditions by the two speaker groups, and by male and female speakers are discussed. [Work supported in part by the SSHRC.]

2aSC33. Acquiring *dynamic* coarticulatory information. Eunjin Oh (Dept. of Linguist., Stanford Univ., Stanford, CA 94305)

This paper investigates the acquisition of dynamic coarticulation information in French [du] by native English speakers. English [du] characteristically shows a concave-downward trajectory, while French [du] shows a concave-upward transition. As an index of coarticulation degree, an absolute understoot measurement [i.e., difference between the undershoot $F2_v$ (the $F2$ frequency at the steady state of the vowel) and the estimated target value] is potentially problematic since $F2$ transitions with the same $F2_v$ values would be treated as equivalent even when their dynamic natures are quite different [C. S. Crowther, UCLA Working Papers in Phonetics **88**, 127–148 (1994)]. The learning effects of the dynamic coarticulatory information were quantified by calculating the value of the second derivative of the $F2$ transition curve and comparing the values of their English and French transition curve. The preliminary results showed (i) the dynamic coarticulatory information can be acquired by L2 learners; (ii) the advanced learners showed closer approximation to the native speakers in the value of the second derivative for $F2$ transition curve than the beginning learners. The results suggest that the L2 learners are sensitive to the detailed acoustic parameters and could progressively acquire the L2-specific dynamic patterns of coarticulation.

2aSC34. Vowel epenthesis in productions of English consonant clusters by Japanese. Keiichi Tajima and Rieko Kubo (ATR Human Information Processing Res. Labs., 2-2 Hikaridai, Seika-cho, Soraku-gun, Kyoto, 619-0288 Japan)

Phonotactic contraints on syllable structure vary across languages. Japanese has a more restricted set of consonant clusters (e.g., ''ho*nd*a'') than English (e.g., ''i*nstr*uct''). This presents a problem for Japanese learners of English, who often avoid consonant clusters by inserting epenthetic vowels between consonants. This study examines how often Japanese speakers produce epenthetic vowels in English and the phonetic environments in which they commonly occur. A group of native Japanese speakers, and a control group of native English speakers, read English words embedded in a sentence, and isolated words spoken at normal and slow rates. Measurements were made from spectrograms of the duration of the target clusters, including any epenthetic vowels produced. Preliminary results suggest that Japanese speakers show more instances of epenthetic vowels than do English speakers, and that such vowels are more commonly observed in slow, careful speech than in normal-rate speech. Also, in the absence of vowel epenthesis, Japanese speakers' productions of consonant clusters differ from those of native speakers in the durational characteristics of component segments. Thus, Japanese speakers' difficulties in producing English consonant clusters are reflected not just through frequent vowel epenthesis, but also through differences in durational properties in the clusters.

Session 2aSP

Signal Processing in Acoustics: Time-Frequency Applications in Acoustics I

Patrick J. Loughlin, Chair
Department of Electrical Engineering, University of Pittsburgh, 348 Benedum Hall, Pittsburgh, Pennsylvania 15261

Chair's Introduction—8:25

Invited Papers

8:30

2aSP1. Adapting time-frequency responses to speech and acoustic classification tasks. Les E. Atlas (Dept. of Elec. Eng., Univ. of Washington, Box 352500, Seattle, WA 98195-2500, atlas@ee.washington.edu)

Much past effort, including our own, has gone into the design of high-resolution time-frequency representations. However, as found in most data-trained automatic classification applications, high resolution can significantly degrade performance. This behavior can be understood from the standpoint of complexity: The higher the resolution and complexity of the time-frequency representation, the larger the size of the training data needed for an accurate classifier. For most applications, where many regions in time and frequency are not salient to the classification task at hand, lowering resolution in these regions should actually improve classifier performance and generality. A new time-frequency approach, called ''class-dependent kernels,'' which selectively smoothes in time and frequency, has been developed. This approach can be summarized by considering the points of a discrete auto-ambiguity function, which is the two-dimensional Fourier transform of a time-frequency representation, as a set of features, which can be ranked in terms of saliency to the particular classification task. Keeping and classifying based on only a small number of these ambiguity plane points corresponds to a flexible and data-adaptive smoothing of the corresponding time-frequency plane. This approach will be shown to be useful for a wide range of acoustic and vibration problems including monitoring acoustic emissions in titanium removal and as an adjunct to cepstra in speaker-independent phoneme recognition.

8:50

2aSP2. Principal features for nonstationary signals from moments of the singular value decomposition of Cohen–Posch (positive time-frequency) distributions. Dale Groutage (Naval Surface Warfare Ctr., Puget Sound Detachment, 530 Farragut Ave., Bremerton, WA 98314-5215)

This paper presents a new method for determining the principal features of a nonstationary time series process based on the singular value decomposition (SVD) of the Cohen–Posch positive time-frequency distribution. This new method uses density functions derived from the SVD singular vectors to generate moments that associate with the principal features of the nonstationary process. Since the SVD singular vectors are orthonormal, the vectors whose elements are composed of the squared-elements of the SVD vectors are discrete density functions. Moments generated from these density functions are the principal features of the nonstationary time series process. These descriptive features can be used in a variety of ways to gain information about the specific aspects of the time series process. This new technique has application to a broad area of fields and applications such as the medical field (nonintrusive diagnostics and condition classification), the defense industry (classification of acoustics transient signals), the field of machinery diagnostics, the field of marine mannal acoustics, and the manufacturing industry. An example is presented which illustrates how the new technique can be used to classify acoustic transient signals from underwater vehicles.

9:10

2aSP3. Cross-spectral methods with applications to speech processing. Douglas J. Nelson (U.S. DoD, 9800 Savage Rd., Fort Meade, MD 20755)

Cross-spectral methods were first presented in the mid-1980's as a method for accurately estimating stable parameters such as modem baud rate and carrier frequencies. The stability of these signal parameters makes it possible to integrate large amounts of data to accurately estimate parameters even under degraded conditions. Since biological signals, such as speech, are not stationary, classical analysis methods, including normal cross-spectral methods, are poorly suited to the problem. Presented here are methods which take advantage of the structure of speech and the phase properties of the Fourier transform. They are based on the cross-spectral

methods of the 80's, but have the advantage that these newer methods provide good accuracy and resolution for nonstationary signals such as speech. In addition, they provide a simple method for taking advantage of signal structure, such as the harmonic properties of speech, which results from the quasi-periodic pitch excitation. Specific problems addressed are accurate pitch and formant estimation, and problems such as blind recovery of carrier frequencies for single side band AM radio transmissions.

9:30–9:50 Break

9:50

2aSP4. Time-varying coherent AM–FM demodulation and denoising of acoustic signals. Patrick J. Loughlin and Ferhat Cakrak (Dept. of Elec. Eng., Univ. of Pittsburgh, Pittsburgh, PA 15261)

Noise removal via linear time-invariant (LTI) filtering is most effective when the signal and noise spectra have minimal overlap in frequency. In particular, it can be difficult to extract, via LTI filtering, broadband signals from broadband noise, because often their spectra overlap. However, many broadband signals are locally narrow band (e.g., AM–FM signals with large FM and moderate to small AM), and this characteristic can be exploited to improve noise suppression for such signals. We present a method for extracting locally narrow band signals from broadband noise, based on an AM–FM decomposition of the signal and time-varying filtering. The center frequency and passband of a linear time-varying filter are determined from estimates of the instantaneous frequency and instantaneous bandwidth of the signal. Results on both synthetic signals and recorded whale sounds in ambient noise demonstrate a significant improvement in SNR compared to LTI-based filtering. [Supported by ONR Grant N00014-98-1-0680.]

10:10

2aSP5. Reconstruction of Formula 1 engine instantaneous speed by acoustic emission analysis. Giorgio Rizzoni (Dept. of Mech. Eng. and Ctr. for Automotive Res., The Ohio State Univ., 206 W. 18th Ave., Columbus, OH 43210-1107, rizzoni.1@osu.edu)

This talk presents some results of a method aimed at extracting instantaneous engine speed information from acoustic emission measurements obtained from Formula 1 (F1) vehicles during a race, using joint time-frequency analysis methods. The analysis method used in this work is applied to acoustic emission data recorded by the microphone of the in-car cameras mounted on F1 vehicles. The data analyzed were acquired during the 1998 Grand Prix of San Marino (Imola), and pertain to the performance of the Ferrari and McLaren–Mercedes vehicles. The analysis presented in the paper is based on data acquired in three different sections of the Imola circuit: the starting straightaway and two curves, to highlight the capabilities of the method. The result of the analysis demonstrates that it is possible to estimate a number of useful variables from sound measurements using joint time-frequency analysis methods. These estimates are related to engine performance (e.g., engine speed and its acceleration, top engine speed), to engine architecture (e.g., gear ratios), to driving strategy (e.g., shifting strategy), and to vehicle performance (tire adhesion, aerodynamic behavior). The analysis includes validation against engine speed data obtained via telemetry measurements.

10:30

2aSP6. Applications of time-frequency analysis in musical acoustics. Gregory H. Wakefield, Maureen Mellody, Rowena Guevara, and William Pielemeier (Dept. of Elec. Eng. and Computer Sci., Univ. of Michigan, Ann Arbor, MI 48109-2122, ghw@eecs.umich.edu)

The modal distribution (MD) is a member of Cohen's class of time-frequency distributions that is designed specifically for signals that are well-modeled as the sum of time-varying partials, such as those generated by many musical instruments [W. Pielemeier and G. H. Wakefield, J. Acoust. Soc. Am. **99**, 2382–2396 (1996)]. When combined with local operators on the time-frequency surface, MD analysis provides substantial improvement over techniques based on the spectrogram with respect to simultaneously resolving variations in amplitude and frequency. Furthermore, the analysis degrades gracefully as the acoustic signal varies from the specified model and, in several cases, can be extended to handle a broader class of signals. Examples drawn from violin vibrato, singing, and the piano are used to illustrate MD analysis and its extensions. Issues of system identification, time-varying signal models, and musical synthesis are also discussed on the basis of these examples. [Research supported by grants from the Ford Motor Company, the Office of Naval Research, the National Science Foundation, and the Office of the Vice President for Research at the University of Michigan.]

Session 2aUW

Underwater Acoustics: Scattering

John Oeschger, Cochair
Coastal Systems, Station Code R21, 6703 West Highway 98, Panama City, Florida 32407-7001

Christopher Feuillade, Cochair
Naval Research Laboratory, Stennis Space Center, Mississippi 39529

Contributed Papers

8:15

2aUW1. Acoustic band gaps and localization in water with air-cylinders. Emile Hoskinson (Dept. of Phys., Univ. of Berkeley, CA) and Zhen Ye (Natl. Central Univ., Chungli, Taiwan, ROC)

Multiple scattering of waves may lead to many peculiar phenomena such as complete band gaps in periodic structures and wave localization in disordered media. Within the gaps excitations are evanescent, when localized waves remain confined in space until dissipated. Here we show that waves are not always localized in 2-D, but when localized there is a coincidence between the gaps in periodic structures and the ranges of localized states in the case of acoustic propagation through water containing an array of air-cylinders. We report that inside the gap or localization regime, an interesting collective behavior of the cylinders appears. Exact numerical calculations reveal the unexpected result that localization is relatively independent of the precise location or organization of the scatterers.

8:30

2aUW2. Validity of the sonar equation and Babinet's principle for object scattering in a shallow water waveguide. Nicholas C. Makris, Purnima Ratila, and Yisan Lai (MIT, 77 Massachusets Ave., Cambridge, MA 02139)

It has recently been shown that the sonar equation can provide a reasonable approximation to nonforward scattering from a noncompact (ka >1) sphere submerged in an ocean waveguide. The omnidirectional scattering characteristics of the sphere in nonforward directions enables the sphere's far-field plane wave scattering function to be factored in the single-scatter waveguide solution just as it is in free space. The highly directional nature of the sphere's scattering in the forward direction, however, prevents this factorization and leads to significant departures from sonar equation predictions in the forward direction. By Babinet's principle, the forward scattered field from a noncompact sphere subject to an incident plane wave in free space will be nearly the same as that of a disk of the same projected area. Accordingly, it is shown that the sonar equation can significantly overestimate the field scattered from an upright plate or disk submerged in an ocean waveguide. These flat objects yield some of the most directional scattering possible from a finite body. It is also shown that in an ocean waveguide the upright disk and an equivalently located sphere with the same great circle area can have nearly identical forward scattered fields, just as in free space.

8:45

2aUW3. Nonlinear scattering of crossed ultrasonic beams in the presence of turbulence: Multiple scattering effects. Rebecca A. Manry and Murray S. Korman (Dept. of Phys., U.S. Naval Acad., Annapolis, MD 21402, korman@nadn.navy.mil)

The nonlinear scattering of two finite-amplitude mutually perpendicular crossed ultrasonic beams—interacting in the presence of turbulence—generates a scattered sum frequency component that radiates outside the interaction region. In the absence of turbulence, virtually no scattering, at the sum frequency, is observed [M. S. Korman and R. T. Beyer, J. Acoust. Soc. Am. **84**, 339–349 (1988); **85**, 611–620 (1989)]. Here, two primary cw ($f_1=2.05$ MHz and $f_2=1.95$ MHz) beams are generated by 2.54-cm-diam circular plane array piston transducer units (T_1 and T_2). A 4-MHz receiving unit (R) is of similar construction. All beam axes form a common plane and overlap region with the axis of a submerged circular water jet, which generates the turbulence. With R fixed, T_1 and T_2 rotate on radius arms—always keeping the beams perpendicular. Symmetry suggests an angle θ_*, where $\theta_*=0°$ defines forward scattering. This geometry allows a nonlinear forward scattering intensity component to exist without concern for a coherent component. Here, multiple scattering effects are needed to predict their results. Supplemental scattering experiments will be presented in an attempt to identify a transition from single (Born approximation) to multiple scattering. [Work supported by Naval Academy Research Council.]

9:00

2aUW4. An indirect method for computing troublesome coefficients in the boundary integral equation method. Ronald T. Kessel (Defence Res. Establishment Atlantic, 9 Grove St., P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada)

The boundary integral equation (BIE) method is ideal for modeling scattering of waves from arbitrarily shaped objects embedded in unbounded domains. Its singular or ''self''-coefficients require special numerical treatment because the kernel of the integral equation exhibits the singularity of the Green's function for a fundamental point source. Regularization for direct numerical integration is particularly complicated for elastic waves in layered media where the Green's function is a tensor whose elements cannot be written in closed form. Here the singular coefficients are computed indirectly, by inferring their values from a family of related propagation scenarios whose solutions are known in advance by virtue of Huygens' principle. The method can be applied quite generally to compute other potentially troublesome coefficients, which may occur in thin platelike objects, for example, when one boundary element lies close and parallel to another on the opposite face, or in horizontally stratified media when one element lies directly above another, making the Green's function poorly convergent. The method is verified here for scattering from elastic spheres (low to moderate ka), and is demonstrated for scattering from an ice keel in floating sea ice, and for a sphere half-buried in seafloor sediments.

9:15

2aUW5. Scattering enhancements for penetrable tilted circular cylinders in water: The computed evolution away from the meridional plane. Philip L. Marston and Florian J. Blonigen (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Significant contributions to the high-frequency backscattering by penetrable bluntly truncated solid circular cylinders include meridional leaky Rayleigh waves [K. Gipson, Ph.D. thesis, Washington State University (1998)] and the caustic merging transition [F. J. Blonigen and P. L. Mar-

ston, J. Acoust. Soc. Am. **102**, 3088 (1997)]. The latter is important for plastic objects (where Rayleigh waves become subsonic relative to water) and is associated with merging transmitted and internally reflected rainbow rays. One way to explore the dependence of these processes on tilt is to compute the exact scattering by infinite circular cylinders away from the meridional plane. The meridional Rayleigh-wave feature for infinite metallic cylinders is a dip in the total scattering and a peak in the background-subtracted scattering [P. L. Marston, J. Acoust. Soc. Am. **102**, 358–369 (1997)]. We find it evolves in a way bounded by the locus of those rays reflected with their local angle of incidence matching the Rayleigh-wave coupling angle. In the caustic merging case, the evolution of the rainbow enhancement is similar to optical observations [Mount *et al.*, Appl. Opt. **37**, 1534–1539 (1998)]. [Supported by the Office of Naval Research.]

9:30

2aUW6. Backscattering enhancement mechanisms for thin tilted circular plates in water. Brian T. Hefner and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

A thin tilted circular glass plate in water was illuminated with high-frequency sound. When the angle of incidence along a diameter of the plate corresponds to the extensional and flexural wave coupling angles, there is an enhancement in the backscattered response. At each of these angles, the largest response arises from a leaky wave which travels along the diameter of the tilted plate and reflects from the plate edge. A similar enhancement was studied previously for the scattering of sound from the flat end of a tilted solid cylinder and, in that case, the reflection of the leaky Rayleigh wave was modeled using methods in ray theory [K. Gipson and P. L. Marston, J. Acoust. Soc. Am. **104**, 1754 (1998)]. For the circular plate, in addition to the scattering associated with this central ray, further enhancement was found near the extensional wave coupling angle associated with rays traveling along off-center paths. The mechanism for this response is believed to depend upon multiple reflections around the rim due to the geometry of the scatterer and may have a significant effect on the overall response of the scatterer. [This work is supported by the Office of Naval Research.]

9:45

2aUW7. Anderson (1950) revisited. C. Feuillade (Naval Res. Lab., Stennis Space Center, MS 39529-5004) and C. S. Clay (Univ. of Wisconsin-Madison, Madison, WI 53706)

The Anderson fluid sphere scattering model [J. Acoust. Soc. Am. **22**, 426–431 (1950)] has been reexamined to clarify three issues which have been the source of misunderstanding among underwater acousticians. First, the accuracy of the Morse large-range approximation for the spherical Hankel functions is investigated. It is shown that the minimum range for use of the approximation is strongly mode-number dependent, and should be carefully evaluated in short-range and/or high-frequency applications. Second, the precise characterization of the forward scatter region is studied. When the scattered field and the incident plane wave are combined, it is shown that little advantage is obtained in detection and localization applications by using forward scattering, rather than backscattering. Third, the translational response, or ''rebound,'' of the sphere under the action of the incident field is examined. By demonstrating that Anderson's theory is a limiting case of Faran's scattering model [J. Acoust. Soc. Am. **23**, 405–418 (1951)] for an elastic sphere, which contains the rebound response, it is shown that the response is completely explainable within Anderson's theory, and is consistent with a description which uses a normal mode expansion around a fixed origin. [Work supported by ONR.]

10:00

2aUW8. Surface backscattering statistics for observations conducted near Kiel, Germany. Timothy H. Ruppel, Christopher Feuillade, and Stephen J. Stanic (Naval Res. Lab., Stennis Space Center, MS 39529)

In May 1993, an experiment was performed off the coast of Kiel, Germany to study the temporal variability of backscattering from the ocean boundaries. In the sea-surface backscattering component of the experiment, two dual-mode acoustic transducers were used to ensonify the surface with 1 or 3 ms pings at 20–90 kHz and incident angles of 65.1° −80.9° from normal. The transducers also recorded the backscattered sound. Analysis of the overall returns shows that they vary significantly from ping to ping, such that the distribution of backscattered energy appears nearly Gaussian. However, close comparison of all the signals from a given run indicates the presence of strong salient patterns in some cases, which suggest scattering from individual features (presumed to be wave crests) which move through the beam pattern from ping to ping. The statistics of the backscattered signal from these features, and their relation to the overall backscattered energy statistics, is the topic of this paper. [This work was funded by the Office of Naval Research.]

10:15–10:30 Break

10:30

2aUW9. Formulation of the interaction of guided waves with buried structures in the shallow ocean. M. F. Werby (NRL Code 7181, Stennis Space Center, MS 39529)

In this work the interaction of a guided wave with a buried object is formulated based on a unified propagation-object scatter model. The model makes use of a range-dependent normal mode description of propagation and a *t*-matrix approach to scattering. In a sediment, certain modes may propagate into the bottom, while others never penetrate and the remaining modes attenuate quickly into the bottom. These facts are used to derive a mathematically plausible method that also has the advantage of yielding intuitive information. Here ocean refraction, boundary interaction at the water-bottom interface, and absorption in the sediment are taken into account. Since this formulation makes use of general *t* matrices there is little limitation on the type of targets that may be considered at the bottom. A result from this formulation is the expected angular spectrum and their weights, which are useful for the experimentalist.

10:45

2aUW10. High-frequency acoustic scattering from medium variability. John Oeschger (Coastal Systems Station, Code R21, Panama City, FL 32407)

Laboratory results of high-frequency acoustic scattering measurements from thermal and saline driven plumes are reported. These experiments used broadband signals in a multi-static configuration. Far-field weak scattering theory describes the acoustic scatter from the medium variability. Expressions relating the acoustic scatter to the scattering field are given in terms of deviations from ambient of temperature and salinity or alternatively, sound speed and density. The angular dependence of the coefficient for the salinity difference has a stronger angular dependence than does the coefficient for the temperature difference. By using the common Bragg wave number comparisons [J. Oeschger, Proc. of 16th ICA and 135th ASA **2**, 1335–1336 (1998)], multi-static measurements can be used to determine the source of scattering, either salinity or thermal in nature, without resorting to *a priori* information. Each angular dependent source-term normalization factor is applied in turn to the data and the one that yields the best agreement predicts the scattering source correctly. Results are also presented for medium mixing where the relative density fluctuations dominate the scattering mechanism. Again the common Bragg wave number comparisons using the multi-static data are able to determine correctly the source of scattering as predicted by supporting environmental measurements.

11:00

2aUW11. Parabolic equation techniques for diffraction and scattering problems. Michael J. Mills and Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

The parabolic equation method has proven to be useful for solving single-scattering problems [M. D. Collins and R. B. Evans, J. Acoust. Soc. Am. **91**, 1357–1368 (1992)]. Parabolic equation techniques for more difficult scattering problems will be presented. An approach involving parabolic approximations for range derivatives and an iteration scheme for the reflected and transmitted fields has been developed for solving problems such as the Sommerfeld diffraction problem and scattering from baffled membranes [Kriegsmann *et al.*, J. Acoust. Soc. Am. **75**, 685–694 (1984)]. This approach has been generalized to include multiple scattering events, with applications such as scattering from compact objects. The parabolic equation techniques will be described and illustrated with examples. [Work supported by ONR.]

11:15

2aUW12. Bistatic scattering: A new way to improve sonar detection capabilities? Francoise Schmitt and Franck Daout (Ecole navale, batiment des laboratoire, BP 600, Lanveoc Poulmic, 29 240 Brest naval, France)

In the field of seabed sonar imagery, it is necessary to establish local scattering models to improve the performances of the detection or recognition algorithms. In this paper, we present the Probability Density Function (PDF) of the acoustic intensity scattered by a natural profile as a function of the bistatic angle. To do this we have developed a 1-D bistatic scattering model called NEWS (Numerical Estimation for Waves Scattering) that incorporates physical phenomenons like multiple reflections, shadow and the reflection coefficient of the profile. Moreover, NEWS takes into account acquisition parameters like sensors characteristics and their positions in relation to the center of the illuminated area. Gaussian spectra for the profile height fluctuation are considered. Five hundred profiles are generated. For each profile NEWS's algorithm gives the angular distribution of the scattered field in amplitude and in phase for all geometries and as a function of incident and scattered wave. The acoustic intensity is then treated as a random variable, and histograms are established. The PDF of the scattered intensity is compared to the K, Weibull and lognormal distributions and we examine the statistical informations providing by bistatic sonar.

11:30

2aUW13. Building a bottom scatter database from measured bottom scatter data in the eastern Mediterranean Sea. Peter Neumann and Gregory Muncill (Planning Systems, Inc., 7923 Jones Branch Dr., McLean, VA 22102-3304, pneumann@plansys.com)

The current U.S. Navy Standard bottom scatter database for low frequencies (up to 5 kHz) is Lambert's Law with a mu coefficient of −27 dB for the world. In recent years, work to characterize scattering from both rough interfaces and the sub-bottom volume has greatly increased the understanding of acoustic interaction with the ocean bottom. However, the analysis of measured bottom scatter data requires an approach that includes the effects of the measurement geometry, higher-order multipaths, and contributions from the water-sediment interface, sub-bottom volume, and the sediment–basement interface. The SCARAB (scattering and reverberation from the bottom) model along with an automated inversion approach (simulated annealing with a downhill simplex) is designed to invert measured bottom scatter data for seven bottom scatter parameters. These seven parameters characterize the water–sediment interface, sub-bottom volume, and sediment–basement interface using a spectral representation for each. The SCARAB model is being used to analyze bottom scatter data measured by NAVOCEANO in the Eastern Mediterranean Sea and produce a database for prediction of bottom scattering strength for active sonar systems. The database is to be submitted to the OAML-SRB in late 1999. [This work is being sponsored by SPAWAR (PMW-185) under ONR management.]

TUESDAY AFTERNOON, 2 NOVEMBER 1999 KNOX ROOM, 3:00 TO 5:50 P.M.

Session 2pAA

Architectural Acoustics and Musical Acoustics: Integration of Synthesized Musical Instruments with Acoustic Environments

Richard H. Campbell, Cochair
Bang–Campbell Associates, Box 47, Woods Hole, Massachusetts 02543

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

Invited Papers

3:00

2pAA1. Driving the virtual orchestra with a conductor gesture interpreter. Frederick W. Bianchi (Music Div., Humanities Dept., Worcester Polytechnic Inst., 100 Institute Rd., Worcester, MA 01609, bianchi@wpi.edu) and Richard H. Campbell (Worcester Polytechnic Inst., Worcester, MA 01609)

The interactive multichannel computer music system known as the ''virtual orchestra'' (VO) has been used several times in professional opera and theater as an alternative to a live pit orchestra. In each case, the VO has been operated by a ''tapper'' who follows the conductor and operates the computer which controls the VO. Recent research in noncontact, video-based conductor gesture interpretation has shown interesting possibilities in tempo and limited musical dynamics management when the gesture interpreter is directly connected to the VO controller. Two different approaches to conductor gesture detection will be discussed and the virtual orchestra will be demonstrated.

3:30

2pAA2. Incorporating gestures into the musical control stream with the conductor's jacket. Teresa Marrin Nakra (MIT Media Lab, 20 Ames St., Rm. E15-368A, Cambridge, MA 02139)

Much contemporary music is created in a studio, put to tape, and performed through speakers to a passive audience. In the process, some of the excitement and spontaneity of live performances is lost. There is arguably a growing problem that our contemporary musical forms have become so complicated that they cannot readily be performed in live concerts. One solution to this problem is to build real-time sensor-based systems for artists to interactively and spontaneously perform with. However, as has been shown by numerous others, these systems are either too simple in their functionality or require such constrained gestures that they force the performer to conform to unnatural or nonintuitive gestures. This paper presents an approach that adopts a unique synthesis-by-analysis method, whereby the signals created by numerous professional musicians are gathered under various rehearsal and performance conditions, studied, interpreted, and used to inform the choices in a real-time performance system. The final results, which will be presented in this paper, demonstrate a variety of attempts to generate more natural, expressive, and intuitive mappings between gesture and sound. We will discuss three different examples, including one taken from a conducting-style gestural metaphor, one from tai chi, and one fron dance.

4:00

2pAA3. Intelligent computer accompaniment systems. Roger B. Dannenberg (School of Computer Sci., Carnegie Mellon Univ., Pittsburgh, PA 15213, rbd@cs.cmu.edu)

Because real-time computer music systems are automated, portable, and affordable, they are finding increasing application in live performance. Unfortunately, most computer music systems offer little more than a tape recorder in terms of their abilities to interact with live musicians. Computer accompaniment is designed to go beyond this ''tape recorder'' model of human–computer interaction. Computer accompaniment is a process in which a computer ''listens'' to a live musician, follows along in a score, and synchronizes an accompaniment score with the live player. Computer systems can reliably accompany a live musician in spite of tempo changes and wrong notes. Many variations of computer accompaniment have been developed. The original work assumed only one live musician playing a monophonic wind instrument such as a trumpet or flute. One variation is to follow the polyphonic performance of a keyboardist by sensing the motion of keys. In ensemble accompaniment, the task is to accompany multiple players. In addition to playing wrong notes, individual players might drop out or become lost, so there are interesting new issues to be dealt with. This work addresses practical problems of integrating computer-based performers with live performers.

4:30

2pAA4. Local performance recording and reproduction: Application to a string quartet. William M. Hartmann and Zachary A. Constan (Dept. of Phys. and Astron., Michigan State Univ., East Lansing, MI 48824)

A goal of traditional multichannel sound recording is to transport the listener into the environment of a musical performance. Ideally, the sound field at the ears of the listener resembles the sound field of a live performance, including ambiance cues to the acoustical character of the performance space. The goal of the local performance recording and reproduction technique (LPR/R) is the reverse. It attempts to transport the music into the environment of the listener. In this technique both the recording and the reproduction steps are different. The LPR/R technique requires at least one separate recording channel for each musical instrument. Recorded channels contain no crosstalk among instruments and no reflected sound from the room surfaces. On reproduction, the channels from different instruments are similarly kept separate; they are never electrically mixed. Instead, the channel(s) for each individual instrument are reproduced by a dedicated array of loudspeaker drivers, attempting to simulate the radiation pattern of the instrument. The LPR/R technique is particularly suited to small ensembles. Listening tests show that it is especially effective when the listener moves. An eight-channel application of the technique to a Mozart string quartet is described and demonstrated.

5:00

2pAA5. Evaluation of multitrack recording practices in rock and roll. Alexander U. Case (Fermata, 117 Atlantic Ave., North Hampton, NH 03862, alex@fermataco.com)

Pop and rock music reaches its listeners using loudspeakers as the interface between the recording medium and the acoustic environment. The art of creating pop music recordings relies on a terrific amount of audio equipment to not just capture but also enhance the musical instruments recorded. In the never-ending search for a more compelling loudspeaker playback experience, the equipment of the recording studio is often used in unexpected and creative ways. The application of signal processing to multitrack pop music occurs using approaches that are accessible, but not necessarily intuitive to the acoustician. In this presentation the snare drum is studied to provide a view of some modern multitrack music recording and mixing techniques. The sound of the drum is analyzed both before and after some typical signal processing in an attempt to quantify the motivation behind these recording practices. Microphone selection and placement priorities are discussed. The application of equalization, compression, and ambience effects is analyzed. The perceptual significance of some processes is demonstrated through the playback of audio examples, both before and after signal processing. A live multitrack mixdown in which discreet elements of the mix can be isolated for evaluation is presented.

Contributed Paper

5:30

2pAA6. A probabilistic method for tracking a vocalist. Lorin Grubb (Andersen Consulting, 1909 Andover Dr., Dover, PA 17315, lorin.grubb@ac.com)

When a musician gives a recital or concert, the music performed generally includes accompaniment. To render a good performance, the soloist and the accompanist must know the musical score and must follow the other musician's performance. Both performing and rehearsing are limited by constraints on the time and money available for bringing musicians together. Computer systems that automatically provide musical accompaniment offer an inexpensive, readily available alternative. Computer accompaniment requires software that can listen to live performers and follow along in a musical score. An implemented system automatically accompanies a singer given a musical score. The focus is a method for robustly detecting the score position and tempo (performance rate) of the singer in real time. Robust score following requires combining information obtained both from analyzing a complex signal (the signer's performance) and from processing symbolic notation (the score). The singer's score position is characterized statistically using a model that combines the available musical information to produce a probabilistic position estimate. By making careful assumptions and estimating statistics from a set of actual vocal performances, a reasonable approximation of this model can be implemented in software and executed in real time during a performance.

TUESDAY AFTERNOON, 2 NOVEMBER 1999 MARION ROOM, 12:55 TO 5:00 P.M.

Session 2pAB

Animal Bioacoustics: Detecting and Identifying Animals Using Acoustics I

David A. Helweg, Chair

SPAWARSYSCEN-San Diego, Code D351, 49620 Beluga Road, San Diego, California 92152

Chair's Introduction—12:55

Invited Papers

1:00

2pAB1. Detection and species identification of baleen whale calls. David A. Helweg (Code D351, SPAWARSYSCEN San Diego, 49620 Beluga Rd., San Diego, CA 92152)

Baleen whales live over extensive home range and time scales. Study of how these animals use their vocalizations for communication requires massive data sampling over long periods. This paper describes a system for automating the sampling and analysis of baleen whale calls. Many species produce stable, homogeneous call structures which lend themselves to automated species identification. We have benchmarked a series of bioacoustical call identification algorithms against a set of blue and fin whale calls while systematically manipulating the signal-to-noise ratio. Blue (*Balaenoptera musculus*) and fin (*B. physalus*) whale calls are very stereotypical. Blue whale ''A'' and ''B'' calls have fundamental frequencies of approximately 17 Hz, narrow bandwidth, well-defined harmonic structure, and typical duration of 15–25 s. Fin whale ''pulses'' have fundamental frequencies of approximately 17 Hz, but are broadband in nature and short (approximately 1-s) duration. The results demonstrated a typical tradeoff of speed versus accuracy. The best algorithm was inserted into an underwater sound recording system and its signal-detection theoretic performance was quantified. Results will be discussed with respect to technological, ecological, and conservation aspects of baleen whale bioacoustics. [Project CS-1082 of the Strategic Environmental Research and Development Program.]

1:20

2pAB2. Acoustic feature extraction for characterizing and classifying animal sounds. Kurt Fristrup (Cornell Lab. of Ornithology, 159 Sapsucker Woods Rd., Ithaca, NY 14850)

Standardized measures for animal sounds were investigated as the basis for characterizing and classifying animal sounds. These measures were computed from a noise-compensated spectrogram. The performance of this system was evaluated in terms of the stability of measurements under varying noise conditions, and the ability to correctly identify sounds to species or individual in a variety of trials. Results from analyses of bird and marine mammal sounds will be presented. General considerations regarding feature and classifier selection will be discussed.

1:40

2pAB3. Using vocal identifiers of individuality in research. Robert Gisiner (Office of Naval Res., 800 N. Quincy St., Arlington, VA 22217-5660, gisiner@onr.navy.mil)

Many animals use vocal cues for individual recognition to mediate interactions between parents and offspring, between mates or between territorial neighbors. These cues or other features of vocalizations with no apparent social role can also be used by researchers to identify individuals or estimate the gender, age, size or other features of the vocalizing individual. In order to establish the reliability of the vocal cue the researcher must first have access to a pool of individuals of known identity, or gender, size, etc. Simple

statistical measures such as the within and between measures of variance from an ANOVA can provide a quantitative measure of the potential utility of a vocalization as an identifier. Sorting of individuals can often be accomplished with a discriminant function analysis using relatively few features (three to five may be all that are needed), but more often a simple neural network classifier can achieve comparable classification results without preselecting acoustic features for sampling. Field application of a trained classifier can greatly aid research requiring rapid identification of individuals when visual identification is difficult or impossible.

Contributed Papers

2:00

2pAB4. Preliminary results on the acoustic characterization of the Northern Right whale. Francine Desharnais and Mark G. Hazen (Def. Res. Est. Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, desharnais@drea.dnd.ca)

The Northern Right whale (*Eubalaena glacialis*) is the most endangered large whale species. Collisions with ships is the cause of a large percentage of documented deaths. An experiment is planned for the summer of 1999 to record vocalizations of the Northern Right Whale. The recordings will be made in the Bay of Fundy (Southeast of Grand Manan Island) and Roseway Basin (South of Nova Scotia). These areas were chosen because of the large occurrence of whale sightings in the summer months. Preliminary analysis of acoustic signals from the Right Whale will be shown. The statistical analysis of these signals should eventually allow the development of processing algorithms to recognize whale calls. Identification of unique features in the whale calls may allow localization and tracking of the Northern Right whales with a field of sonobuoys.

2:15

2pAB5. The dependence of target strength of the northern right whale (*Eubalaena glacialis*) on the acoustic properties of blubber. James H. Miller, Thomas Weber, Angela Tuttle (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882, miller@uri.edu), and David C. Potter (Northeast Fisheries Sci. Ctr., NMFS, Woods Hole, MA 02543)

Whale detection and tracking using active sonar is a subject of recent study [J. H. Miller, D. C. Potter, T. Weber, and J. Felix, J. Acoust. Soc. Am. **105**, 992 (1999)]. In that previous work, the measured target strength of a northern right whale (*Eubalaena glacialis*) was found to be less than that measured for a humpback whale (*Megaptera novaeangliae*) of the same size and aspect (about 0 and 5 dB, respectively). The difference was explained by modeling the thicker blubber of the right whale as a lossy layer in a plane wave reflection coefficient. However, values for the acoustic properties of the blubber were estimated from tissue properties of other mammals [R. C. Chivers and R. J. Parry, J. Acoust. Soc. Am. **63** (1978)]. In April, 1999, a 60-ton northern right whale named Staccato was found dead near Cape Cod. Blubber samples were acquired by the National Marine Fisheries Service and acoustic properties determined using a multi-sensor core logger in URI Marine Geomechanics Laboratory. Sound speed and density data have been used to refine the plane wave reflection coefficient model and corroborate the earlier work. [Work supported by NMFS.]

2:30

2pAB6. Visual and acoustic surveys of whales: A Monte Carlo model. Kathleen J. Vigness (Marine Acoust., Inc., 901 N. Stuart St., Ste. 708, Arlington, VA 22203)

There has been much interest lately in decreasing the high variability of cetacean abundance estimates by coupling passive acoustics with visual surveys. To estimate the advantages and disadvantages of a dual-mode survey, a Monte Carlo model was created. The model can reflect the species and the methodology of a particular survey. The whale population is defined by the parameters of group size and encounter rate, swimming speed and direction, surfacing rate and pattern, and source level and rate of vocalization. The survey process is constrained by the additional parameters of vessel speed, sighting probability, hydrophone array characteristics, and sound propagation. With results from the model, researchers can compare the abundance estimates and variability from a visual survey only, a passive acoustic survey only, or a dual-mode survey. They can also determine which parameters are important for a species, region, and methodology, and design their surveys accordingly. Sample results from a hypothetical minke whale population demonstrate that adding passive acoustics significantly increased the number of groups detected and decreased the time the groups were in the survey area before being detected.

2:45

2pAB7. Detection and censusing of blue whale vocalizations along the central California coast using a former SOSUS array. Ching-Sang Chiu, Christopher W. Miller, Therese C. Moore, and Curtis A. Collins (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943)

To investigate the feasibility of automating the detection and censusing of blue whale vocalizations over a large coastal region using the Naval Postgraduate School (NPS) Ocean Acoustic Observatory (OAO), a four-day experiment was conducted along Central California in the summer of 1997. During the experiment, array data were archived continuously at the NPS OAO, a former SOSUS array. In addition to shore-based acoustic monitoring, an aircraft was assigned to locate blue whales in the Monterey Bay National Marine Sanctuary, and a research vessel, manned with observers and instrumented with a towed hydrophone array, was used to confirm locations of the blue whales and classify the vocalized near-field signals. The shipboard measurements were required to provide a means to separate the source signal characteristics from their multipath signatures. The towed array data were deverberated, source level and characteristics were estimated, and call-to-call variability was studied. Based on robust signal parameters, two simple autodetection correlation filters for the ''A'' and ''B'' blue whale calls were constructed, respectively. By applying the autodetection filters to the OAO data, the number of blue whale calls for the summer of 1998 was tallied. [Research supported by SERDP/ONR.]

3:00–3:15 Break

3:15

2pAB8. Geographic variations of the Hawai'ian spinner dolphin (*Stenella longirostris*) whistle repertoire: Do they exist? Carmen Bazua-Duran and Whitlow W. L. Au (MMRP, Univ. of Hawai'i, P.O. Box 1106, Kailua, HI 96734, bazua@soest.hawaii.edu)

In the present work, groups of spinner dolphins (*Stenella longirostris*) off four of the main Hawai'ian islands (Hawai'i, Lana'i, Maui, and O'ahu) have been studied. The whistle repertoire of dolphin groups from each island will be compared to search for geographic variations. In addition, behavioral, ecological, and acoustical variables have been quantified in order to correlate the dolphin whistle repertoire with specific behavioral and ecological factors. Geographic variations in the whistle repertoire of coastal bottlenose dolphins (*Tursiops truncatus*) from the Gulf of Mexico have been found. The differences were not attributable to different behavioral states, but were related to group structure (group size and contacts established between groups) and coastal habitat. These findings on the whistle repertoire of coastal bottlenose dolphins support the hypothesis that delphinid whistles serve to coordinate group behavior. Preliminary results on group structure of spinner dolphin groups off Hawai'i suggest that geographic variations should exist. Gulf of Mexico bottlenose and Hawai'ian spinner dolphin societies share several characteristics. Thus, it

is expected that the Hawai'ian spinner dolphin whistle repertoire will show variations as does the bottlenose dolphin whistle repertoire. [This work was supported by a Leonida Memorial Scholarship and Seed Money Grant from the University of Hawai'i.]

3:30

2pAB9. Source level estimation of dolphin communication calls as a potential method for species identification. Kristin Kaschner (Marine Mammal Res. Unit, Univ. of British Columbia, Hut B-3, 6248 Biol. Sciences Rd., Vancouver, BC, Canada), A. David Goodson, and Paul R. Connelly (Loughborough Univ., LE11 3TU, UK)

Research into the acoustic structure of communication calls of different odontocete species has attempted to identify distinctive features that might assist in discrimination of individuals, pods or species. Most studies have concentrated on frequency modulation characteristics, such as the general shape or contour of the call. Although the source level (SPL *re*: 1 μPa at 1 m) of echolocation signals has been studied and appears to reflect some body size dependency, signal intensities of ''whistles'' have not been investigated closely. In this study maximum source levels of communication calls were estimated in open sea conditions, using an acoustic multi-path ranging technique. Preliminary results suggest species-specific differences in maxium call intensities that may provide a useful additional cue to acoustically distinguish species in the wild. Data used for spectral and statistical analysis were recorded using a single hydrophone attached to a pelagic trawl fishing in continental shelf waters in the Bay of Biscay. Maximum SPLs of social calls were extracted and distances between hydrophone and vocalizing animals were obtained exploiting information inherent in recorded muli-path signals. Concurrent visual observation provided species identification. Source levels of calls were calculated based on a calibrated reference signal. Methodology and limitations of this technique are discussed.

3:45

2pAB10. Vocalizations in two captive born Florida manatees (*Trichechus manatus latirostris*). Katherine J. Frisch (Western Illinois Univ., Macomb, IL 61455-1390)

Florida manatees (*Trichechus manatus latirostris*) are an endangered species with a population of less than 3000 individuals. They are known to produce sounds. Some attempts have been made to characterize their vocal repetoire. However, little has been documented about intermanatee differences in these vocalizations. This pilot study was undertaken to examine the acoustic differences in vocalizations of two captive born manatees at Mote Marine Laboratory in Sarasota, FL. The manatees were recorded at random intervals while engaged in typical activities. The recordings were analyzed along a number of dimensions. Despite a great deal of overall variability, there was a clear bimodal distribution of frequency characteristics, particularly near the end of the vocalization. This suggests that manatee vocalizations may have an individually determined natural frequency range. These acoustic cues could provide additional means of identifying and distinguishing individuals from each other in the wild. Such a finding has important applications, for example, in the effort to preserve the species as an aid in the cataloging of individual animals.

4:00

2pAB11. Vocal discrimination of two species of manatees (*Trichechus inunguis*) and (*T. manatus manatus*) in Brazil. R. S. de Sousa Lima (PGECMVS/Depto. Zoologia/Univ. Federal de Minas Gerais, Belo Horizonte, Brazil), A. P. Paglia (UFMG, Belo Horizonte, Brazil), and G. A. B. Fonseca (UFMG/CABS/Conservation Intl.)

Localizations from 15 West Indian manatees (*T.m. manatus*) and 14 Amazonian manatees (*T. inunguis*) were recorded and selected for spectral analysis. Several physical parameters of these vocalizations were used to describe and compare the vocal repertoires of these two species. Five of those variables were used in a principal components analysis (PCA) in order to verify if the vocal parameters could be grouped by species. Both approaches (repertoire comparison and PCA) discriminated these two species. The main differences were in the limits of the fundamental frequency (range) of each species, 1.07 to 8 kHz for (*T. inunguis*) and 1.07 to 4.98 kHz for (*T.m. manatus*), and the number of notes per vocalization. Only the Amazonian manatee presented more than one note (1 to 4) per vocalization. The first axis of the PCA explained almost 70% of the data variation. The most important variables in this axis were related to the fundamental frequency. Therefore, the characteristics of the fundamental frequency are responsible for the discrimination of the vocal patterns of (*T. inunguis*) and (*T.m. manatus*). Such patterns are specie-specific and could be used as a taxonomic feature. [Work supported by FBPN, MacArthur Foundation, Conservation International, USFWS, and CNPq.]

4:15

2pAB12. A two-stage process for automatic recognition of harbor seal vocalizations. David K. Mellinger[a] (MBARI, 7700 Sandholdt Rd., Moss Landing, CA 95039, dave@mbari.org)

Male harbor seals (*Phoca vitulina*) were continuously recorded for a year from an array of hydrophones in shallow water off the coast of central California. A two-stage automatic recognition system was used to extract sounds of interest. The first stage, fast but crude, processed the entire sound archive. It operated by: (1) making a spectrogram; (2) normalizing the spectrogram in several ways to remove some background noises and interfering sounds; and (3) detecting sounds in the 100–1000 Hz range with a minimum duration of 1 s. The second stage, slow but accurate, operated on the sounds extracted by the first stage and classified them as being either harbor seal roars or not. Classification was done by measuring a variety of acoustic characteristics—duration, frequency span, amplitude variation, etc.—in several frequency bands, and applying statistical pattern recognition techniques to the resulting feature vectors. Training data consisted of 1011 roar examples and 850 nonroar sounds. Recognition accuracy greater than 95% was achieved, with the principal errors occurring because of close resemblance between seal roars and breaking-waves. These results show that acoustic monitoring combined with automatic recognition can be a viable method for continuous monitoring of populations of wild animals. [a] Currently at PMEL, 2115 S. E. OSU Dr., Newport, OR 97365.

4:30

2pAB13. Passive detection and localization of marine mammals in the Ligurian Sea. Angela D'Amico, Joseph E. Bondaryk, and Nicola Portunato (SACLANT Undersea Res. Ctr., Viale S. Bartolomeo 400, 19138 La Spezia, Italy, damico@saclantc.nato.int)

SACLANTCEN sponsored an environmental acoustic measurement in the Ligurian Sea in August 1999. The objectives of this cruise were to detect, localize and track marine mammals using passive acoustic techniques. During this cruise, the R.V. ALLIANCE passively operated several advanced sonar arrays near Corsica. Acoustic techniques were used to localize and track several species of cetaceans in position and depth over time from their vocalizations only. The acoustic results are compared to ground truth provided by visual observations and data from several radio tags with time-depth recorders. Statistics on whale vocalizations and behavior are tabulated. At the time of this abstract submission, it is anticipated that data on Fin, Sperm and other cetaceans will be available.

4:45

2pAB14. Passive acoustical survey of finless porpoises in the Yangtze River, China. Tomonari Akamatsu (National Res. Inst. Fisheries Eng., Ebidai, Hasaki, Kashima, Ibaraki, 314-0421 Japan, akamatsu@nrife.affrc.go.jp), Ding Wang, and Kexiong Wang (The Chinese Acad. of Sci., Wuhan 430072, PROC)

Finless porpoises (*Neophocaena phocaenoides*) are distributed in Asian waters. The narrow-band and high-frequency pulse sonar signals produced by this species are distinctive from background noises. Underwater sound monitoring by a hydrophone (BK8103) along board sides of a

research vessel concurrent with visual observation were conducted in the Yangtze River from Wuhan to Poyang Lake in 1998. The peak-to-peak detection threshold level was 133 dB re: 1 μPa, which could be roughly converted to the theoretical detection range of 316 m, hypothesizing the porpoise directing to the hydrophone and 168-dB source level of the sonar signal under a shallow water sound propagation (combination of spherical and cylindrical spreading). In a total of 1064-km cruise, 717 finless porpoises were observed. The acoustical monitoring system could detect the sonar signals from finless porpoises, found within 200 m from the research vessel. The sonar signals could also be detected at night and under windy weather conditions. Basically, the acoustical observation system was operated automatically and free from the individual difference of observers. The high-frequency acoustical monitoring seems to be an effective method for the survey of small cetaceans which produce sonar signals.

TUESDAY AFTERNOON, 2 NOVEMBER 1999 KNOX ROOM, 1:00 TO 2:30 P.M.

Session 2pBB

Biomedical Ultrasound/Bioresponse to Vibration: Role of Vibration in Medicine

Diane Dalecki, Chair

Department of Electrical and Computer Engineering, University of Rochester, Hopeman Building, Rochester, New York 14627

Contributed Papers

1:00

2pBB1. Lung response to low-frequency underwater sound. Diane Dalecki, Carol H. Raeman, Sally Z. Child, Stephen A. McAleavey, and Edwin L. Carstensen (Dept. of Elec. and Computer Eng. and the Rochester Ctr. for Biomed. Ultrasound, Univ. of Rochester, Rochester, NY 14627)

The response of the lung to exposure to low-frequency underwater sound was investigated for frequencies of 100–500 Hz. Several different experimental and theoretical approaches demonstrated that the lung responds to low-frequency underwater sound as a resonant structure. Measurements of acoustic scattering near murine and rat lung indicated that the response of the lung to low-frequency acoustic fields can be described well by the theory of linear oscillations of a bubble in water. The measured mean resonance frequency was 325 Hz for adult mice (30 g), 420 Hz for young mice (10 g), and 175 Hz for rats (320 g). Noninvasive measurements of the displacement amplitude of the lung using an ultrasonic pulse-echo ranging technique confirmed that the lung oscillates in response to exposure near the resonance frequency. At the resonance frequency the response of the lung is maximized and the thresholds for damage to the lung and surrounding tissues (such as liver) were lowest. For exposure of adult mice at the resonance frequency, the threshold for lung hemorrhage and for liver hemorrhage was ~184 dB re: 1 μPa (i.e., 1.6 kPa).

1:15

2pBB2. Measurements of lung vibration in response to low-frequency sound. Stephen A. McAleavey, Carol H. Raeman, Kevin J. Parker, and Diane Dalecki (Dept. of Elec. and Computer Eng., Univ. of Rochester, Rochester, NY 14627)

The amplitude of vibration of lung in response to low-frequency (100–500 Hz) sound was measured in mice and rats *in vivo*. Low-frequency sound fields were generated in an inertial water column driven at its base with an electromagnetic transducer. Estimates of vibration amplitude of the lung were obtained by calculating the variations in the round-trip delay of pulsed wideband ultrasound bursts. The relative time delays were calculated by determining the maximum of the cross-correlation of the first echo with subsequent echoes. The ultrasound bursts were emitted at a rate of 1–5 kHz, allowing ten displacement estimates to be calculated per cycle of lung oscillation. The mean resonance frequencies obtained through measurements of vibration amplitude were 330 Hz for mice and 189 Hz for rats. The maximum observed displacements were on the order of 0.1 mm. The resonance observed through measurements of displacement amplitude was equivalent to that obtained through acoustic scattering measurements and consistent with observations of lung hemorrhage.

1:30

2pBB3. The effect of age on vibrotactile adaptation. Alan K. Goble (Dept. of Psych., Bennett College, 900 E. Washington St., Greensboro, NC 27401-3239, goble@bennett1.bennett.edu)

The aging process is known to affect the processing of somatosensory information. For example, a number of recent studies have shown that detection thresholds for vibrotactile stimuli are elevated at low and high frequencies in elderly individuals, as compared to comparable measurements in young adults. The present study is the first to directly measure the effect of aging on vibrotactile adaptation. Detection thresholds for 25-Hz sinusoidal vibrations presented to the thenar eminence via a 5-mm-diameter contactor were measured in a small group of senior citizens ($n=4$) before and after exposure to a 25-Hz 20-dB SL adapting stimulus and compared to comparable measurements previously obtained on younger adults ($n=4$). As expected, baseline thresholds were significantly higher in the older observers by some 12 dB ($p<0.0000$). Furthermore, older individuals exhibited significantly less adaptation than their younger counterparts (71 vs 14.4 dB, $p<0.0000$). These results provide evidence of age-related changes in both peripheral and central nervous system function, and suggest directions for future research. [This research was supported by NIH Grant No. 2 S14 GM44780-08.]

1:45

2pBB4. Excitation and response of surface waves on isotropic and nonisotropic viscoelastic half-spaces with application to medical diagnosis. Thomas J. Royston (Dept. of Mech. Eng., Univ. of Illinois at Chicago, Chicago, IL 60607), Hussein A. Mansy, and Richard H. Sandler (Rush Medical College, Chicago, IL 60607)

An analytical solution is proposed for the problem of surface wave generation on a viscoelastic half-space by a finite rigid circular disk located on the surface and oscillating normal to it. The solution is an incremental advancement of the work reported in two articles published in the 1950s in the Proceedings of the Royal Society. While the application focus of those articles was seismology, the application of interest here is medical diagnostics. Consequently, the solution is verified experimentally using a viscoelastic phantom with material properties that approximate biological soft tissue. Also, the effect of an inclusion on surface wave behavior within the otherwise isotropic medium is investigated. Measurement of wave motion on the skin surface caused by internal biological functions or external stimuli has been studied by a few researchers for rapid, nonintrusive diagnosis of a variety of medical conditions. Conditions considered

include those associated with cardio-vascular dynamics, oedema and other skin ailments, hardened tissue regions, e.g., tumors, and extraluminal air in the abdomen, a.k.a. pneumoperitineum. It is hoped that the developments reported here will advance these techniques and also provide insight into related diagnostic techniques, such as sonoelastic imaging and other vibro-acoustic methodologies. [Work supported by the Whitaker Foundation.]

2:00

2pBB5. Dynamic response characteristics of upper respiratory system. Ahmed Al-Jumaily (Diagnostics and Control Res. Ctr., Auckland Inst. of Technol., Auckland, NZ) and Ammar Al-Saffar (Univ. of Sci. and Technol., Irbid, Jordan)

A theoretical model is presented to study the general dynamic characteristics of the upper human respiratory system for a healthy as well as an unhealthy lung. The model is formulated by using perturbation technique on the fluid momentum and continuity equations taking into consideration the structural dynamic features of the air passage walls. A recursion formula is generated to compute the input impedance along the principal path. Also the distributed pressure response along a single pathway in the breathing system is investigated for different frequencies. The results indicate that for a healthy lung the magnitude of the impedance at the throat decreases to a minimum value at a frequency of 379 Hz, then increases again, and continues to fluctuate between maximum and minimum values. This prediction agrees very well with available experimental data and disputes earlier theoretical observations of double minimum values. However, it is indicated that a lung with some abnormality at a Horsfield order of 24 and above has its first minimum at a frequency of 238 Hz. A detailed explanation of the dynamic characteristics in terms of the impedance ration and the pressure distribution is presented in the paper.

2:15

2pBB6. Electrical bioimpedance monitoring of cardiovascular effects of noise. Goran Belojevic (Inst. of Public Health of Serbia ''Dr Milan Jovanovic Batut,'' Dr Subotica St. No. 5, 11000 Belgrade, Serbia, FR Yugoslavia), Vesna Stojanov (Inst. for Cardiovascular Diseases, Belgrade, Serbia, FR Yugoslavia), Branko Jakovljevic, and Jelena Ivanovic (Belgrade Univ. School of Medicine, Belgrade, Serbia, FR Yugoslavia)

Cardiovascular effects of recorded traffic noise ($L_{eq}=89$ dBA) were monitored with an ''AVL 2001'' electrical bioimpedance apparatus on 12 medical students (4 male and 8 female), and compared to quiet conditions ($L_{eq}=40$ dBA) before and 10 min after exposure. There were significant changes of the following cardiovascular parameters in quiet–noise–quiet conditions (Mean$\pm$SD): cardiac work/kgm/m^2/(7.0$\pm$2.0; 6.4$\pm$1,5; 7,1 $\pm$2,1; respectively; $P<0.01$, Friedman ANOVA test); global flow [L/min/m^2] (5.8$\pm$2.0; 5.4$\pm$1,5; 6.0$\pm$2.0; respectively; $P<0.01$), pump output/beat [ml/m^2] (75$\pm$26; 69$\pm$18; 78$\pm$21, respectively; $P<0.05$) and peripheral vascular resistance (afterload) [Ω m^2] (1347$\pm$476; 1387 $\pm$458; 1273$\pm$430; respectively; $P<0.01$). No significant effects of noise were observed concerning pulse frequency, systolic and diastolic arterial blood pressure, preload, pump efficiency, thoracic fluids, and contractility. These results indicate that intense noise has a strong vasoconstrictive effect, which in longer terms may lead to disturbances in the regulation of arterial blood pressure and in coronary circulation.

TUESDAY AFTERNOON, 2 NOVEMBER 1999 HARRISON ROOM, 1:00 TO 4:00 P.M.

Session 2pEA

Engineering Acoustics, Physical Acoustics and Structural Acoustics and Vibration: Acoustic Nondestructive Evaluation: New Directions and Techniques, Part II

P. K. Raju, Chair

Mechanical Engineering Department, Auburn University, Ross 201, Auburn, Alabama 36849-5341

Invited Papers

1:00

2pEA1. Impulse response characterization of composite materials and structures for design and manufacturing. Ronald F. Gibson (Mech. Eng. Dept., Wayne State Univ., Detroit, MI 48202, gibson@eng.wayne.edu)

Impulse response testing may be used for fast and efficient characterization of both the elastic and viscoelastic properties as well as the structural integrity of composite materials and structures. Impulsive excitation is used to induce vibrations in the structure and modal frequencies and damping factors are extracted from the response using frequency domain and/or time domain techniques. The use of measured modal frequencies in the solution to the inverse problem for composite beams or plates yields the elastic constants for the composite. A variation on this method yields the spatial distribution of fibers in the composite. For certain configurations, the damping has been found to be a good indicator of interlaminar fracture toughness of a composite. Adhesively bonded composite structures having various flaws and defects have also been tested using this method, and both frequencies and damping factors have been found to vary from those of the ''good'' structures. PC-based virtual instrumentation software has been developed for data acquisition and analysis, and the method has also been used to test full scale composite automotive vehicle components as well as small laboratory samples. The long range goal of the research is the development of rapid, inexpensive screening tests for composite components.

1:30

2pEA2. Elastodynamic response of ordered materials. Vikram K. Kinra (Aerosp. Eng., Texas A&M Univ., College Station, TX 77843-3141)

The interaction of a normally incident plane longitudinal wave with an ordered or disordered single layer of spherical inclusions embedded in a polyester matrix was measured. Inclusions were arranged in a periodic (square or hexagonal) and random arrays with area fraction ranging from 0.07 to 0.65. Measurements were carried out at wavelengths that are large, equal, or small compared to the two characteristic lengths of the composites, namely, the particle radius and the interparticle distance. The transmission and reflection

spectra for periodic layers are characterized by several resonances, the frequencies of which are close to the cut-off frequencies of the appropriate shear lattice modes. The excitation of resonances is accompanied by the propagation of mode-converted shear waves, which are propagated along certain symmetry directions within the plane of the particles. These waves were detected by the use of a shear wave transducer. At the first lattice resonance frequency, the reflection coefficient of a layer drops down to near-zero value. The vibrations of an individual particle have been measured by a laser interferometer. Finally, a simple model of the elastodynamic event was constructed. In spite of its simplicity, the model captures the essential features of the experimental data.

Contributed Papers

2:00

2pEA3. Differential ultrasonic stress–strain measurements. Sissay Hailu (Dept. of Elec. Eng. and Computer Sci., CWRU, Cleveland, OH 44106), Gary R. Halford (NASA, John H. Glenn Res. Ctr., Cleveland, OH 44135), Dov Hazony, and Gerhard Welsch (CWRU, Cleveland, OH 44106)

Fatigue tests are often encumbered by lack of the specimen's exact length due to mechanical grip effects and by environmental noise factors. Such issues may be mitigated when differential measurements are undertaken between consecutive data states. Our principle-monitoring tool is an ultrasonic pulse-echo process where the primary and secondary echoes along the principle specimen's axis provide both length and gage length, respectively [D. Hazony, Circuit Systems Signal Process. **14**(4), 525–538 (1995)]. The process is also useful for crack detection [I. Mostafa *et al.*, Int. J. Fracture **85**, 99–109 (1997)]. Compared with absolute measurements, differential stress-strain monitoring is shown to be highly reproducible and more sensitive providing changes in Young's modulus and Poisson's ratio as well as a focus for the detection of an emerging crack. [Work supported by NASA.]

2:15

2pEA4. Acoustic detection of pressure in sealed drums. R. Daniel Costley (Miltec, Inc., Natl. Ctr. for Physical Acoust., University, MS 38677, dcostley@mil-tec.com) and Mark Henderson (Mississippi State Univ., Starkville, MS 39762)

At many waste sites, transuranic (TRU), low-level, and mixed wastes are stored in 55-gallon drums. Many of these drums contain hazardous, organic wastes as well. Radiolysis or other physical or chemical processes may result in gaseous emissions inside these drums. When this happens the pressure within the drum will increase, sometimes to unacceptable levels. In more drastic cases, these emissions may produce flammable or explosive atmospheres (e.g., hydrogen from radiolysis). Currently regulatory procedures requires that each drum be individually opened and inspected for the presence of hazardous organic waste. This situation will be dangerous for workers. A nonintrusive technique has been developed which will detect an increase in pressure over ambient levels and alert workers of potential danger and greatly increase safety. When a drum lidis tapped, it vibrates at specific frequencies. It turns out that the natural frequencies of vibration of the drum lid increase as the pressure inside the drum increases. Thus the pressure within the drum can be determined by measuring the frequency at which the drum lid vibrates. Experimental results and plans to incorporate this into a simple handheld device will be discussed. [Work supported by DOE.]

2:30–2:45 Break

2:45

2pEA5. Acoustic sensors and NDE techniques for process control in drug, cosmetic and food manufacturing industries. Hasson M. Tavossi (Dept. of Eng. Sci. and Mech., The Penn State Univ., 227 Hammond Bldg., University Park, PA 16802-1401)

New techniques of acoustic sensors for process control in manufacturing industries are presented. Two practical applications of acoustic sensors are considered. The first case involves remote determination of bulk-temperature of a paste heated in an agitated mixer, when other means for temperature measurement are not tolerated. During chemical processing the precise value of bulk-temperature of the paste must be known for quality control of the final product. To determine bulk-temperature, thermal expansion of a rod, placed in the mixer, is measured by acoustic sensors, in the pulse-echo and the through-transmission modes. The results show that in the range of 60 °F–210 °F a temperature change of 1 °F can be measured. The second case relates to the nondestructive measurement of the solid content of the packaged food products. Accurate value of the solid-weight-ratio is required for correct thermal processing. Nondestructive methods with acoustic sensors in the through-transmission mode are utilized to measure solid-weight-ratio of a binary mixture of water and a food product. Using Wood's equation for porous media, a precision of less than 5% in the value of solid-weight-ratio is obtained by this new technique.

3:00

2pEA6. Profiling a surface using its echoes. Gareth Block and John G. Harris (216 Talbot Lab., 104 S. Wright St., Urbana, IL 61801)

The electromechanical reciprocity relation is used to construct a theoretical model of the imaging of a sinusoidal fluid–solid interface using a cylindrically focused acoustic beam. The electromechanical reciprocity relation is used to connect the change in the voltage measured at the electrical terminal of the transducer to the perturbation in the mechanical wave field caused by a change in the profile of the interface. It does so by mixing an unperturbed reference wave field with one containing the perturbed wave field. We use a regular boundary perturbation expansion to obtain an approximate boundary condition that fits directly into the reciprocity relation. This perturbation assumes both the amplitude and the slope of the profile are small. We limit our discussion to imaging a sinusoidal interface with the understanding that resolution of this profile is central to understanding that of more general profiles. We find that depending on the choice of parameter values the measured profile is not simply a replication of the original sinusoidal profile. Time permitting, we briefly indicate preliminary work on a related model of the imaging of the mechanical properties of a thin solid film using a confocal arrangement of point-focused transducers. [Work supported by NSF.]

3:15

2pEA7. Detection of cracks in plates using guided waves. Christoph Eisenhardt, Laurence Jacobs (School of Civil and Environ. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0355), and Jianmin Qu (Georgia Inst. of Technol., Atlanta, GA 30332-0405)

This research combines laser ultrasonic techniques with the two-dimensional Fourier transform (2-D-FFT) to investigate the effect of cracks on the dispersion of Lamb waves propagating in thin aluminum plates. The high fidelity and broad bandwidth of these optical techniques are critical elements to the success of this work. The experimental procedure consists of measuring a series of equally spaced, transient waves in aluminum plates containing notches; a crack is simulated with a 1-mm-thick saw-cut notch. The frequency spectrum (dispersion curves) for each plate is obtained by operating on these transient waveforms with the 2-D-FFT; this procedure extracts steady-state behavior from a series of transient waveforms. This study quantifies the effect of notch depth (two notch depths are examined: one-fourth and one-half of plate thickness) on the dispersion curves of three different plate thicknesses (nominal thicknesses of 1, 1.5, and 3 mm). These dispersion curves show that a notch reduces the transmitted energy by an amount that is directy proportional to a notch's depth. In addition, scattering by a notch causes definitive reductions in energy (evident in all modes) at certain frequency–wave-number combinations, thus providing experimental evidence of the relationship between crack size and the scattered Lamb wave field.

3:30

2pEA8. Detection of fatigue crack initiation and growth in steel specimens from Rayleigh scattering of 5-MHz Rayleigh waves. Daniel A. Cook and Yves H. Berthelot (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

Detecting the formation of surface microcracks remains an important challenge for condition-based maintenance (CBM) and remaining life prediction algorithms. A relatively new method [see M. Resch and D. Nelson, ASTM STP **1149**, 169–196 (1992)] consists of monitoring the reflection of pulsed surface waves off the crack. The method is extended to monitor the Rayleigh forward and backscattering (low-ka regime) with pulsed 5-MHz surface waves propagating on steel specimens under tensile cyclic loading. The cross-correlation of the received signals measured as a function of the number of fatigue cycles can be used to detect the presence and the growth of small cracks. Experimental results will be presented and discussed. [Work supported by the Office of Naval Research, N00014-95-1-0539, MURI Center for Integrated Diagnostics.]

3:45

2pEA9. Vibroacoustical nondestructive evaluation of fatigue crack size. Nadejda I. Bouraou (Dept. of Orientation and Navigation Systems, Natl. Technolog. Univ. of Ukraine, 37, Peremogy Pr., Kiev, 252056 Ukraine, nadye@burau.inec.kiev.ua) and Alexander N. Tyapchenco (Natl. Technolog. Univ. of Ukraine, Kiev, 252056 Ukraine)

For the vibroacoustical nondestructive evaluation of the fatigue crack's relative size the feature set, components of which are ratios of the testing object free or forced oscillations spectral amplitudes of higher and basic harmonics, is used. The features being used are functions of the relative crack size and do not depend on initial excitation intensity. The evaluation of the above-mentioned fault is produced by using the generalized likelihood ratio method. The maximum likelihood equations for using the feature set are solved. New expressions of the relative crack size estimation are obtained. The estimation of each feature is represented as the estimation set component. On the basis of the estimation set the statistics (mean, standard deviation) are computed.

TUESDAY AFTERNOON, 2 NOVEMBER 1999 GRANT–HARDING ROOM, 12:55 TO 3:00 P.M.

Session 2pED

Education in Acoustics: Integrating Acoustics into the Curriculum of Other Disciplines

Daniel R. Raichel, Chair

Department of Mechanical Engineering, City College of New York, 140 Street and Convent Avenue, New York, New York 10031

Chair's Introduction—12:55

Invited Papers

1:00

2pED1. One approach to architectural acoustics in education. J. Christopher Jaffe[a)] (Jaffe Holden Scarbrough Acoustics, Inc., 114A Washington St., Norwalk, CT 06854)

In the fall of 1997, Dean Alan Balfour of the School of Architecture at Rennselaer Polytechnic Institute asked me to introduce an undergraduate course entitled ''Sonics in Architecture.'' I had previously taught a number of two-credit survey courses at Julliard, RPI, and City College of New York. My limited exposure as an adjunct professor did not prepare me for the difficulty of developing and integrating a 14-credit certificate course in what was in reality a vertical studio (a studio with students from different classes and disciplines). This paper discusses the curriculum I developed, the strengths and weaknesses of my initial plan, examples of student work and my revised curriculum for next year. In addition I will share my concerns regarding the teaching methods currently prevalent in many schools of architecture today, and how building science professionals might assist in addressing these issues. [a)] Distinguished Visiting Professor at the Rensselaer Polytechnic Institute.

1:30

2pED2. Integrating acoustics into mechanical engineering education. M. G. Prasad (Dept. of Mech. Eng., Stevens Inst. of Technol., Hoboken, NJ 07030)

Acoustics is generally considered as an area of specialization. However, the significance of acoustics can be seen in several disciplines including science, engineering, and arts, etc. Thus acoustics as knowledge that deals with production, propagation, transmission, and reception of sound plays an important role in many different fields. Acoustics can be integrated into mechanical engineering education through core as well as elective courses. As core material, acoustics nicely fits into a course in dynamical systems. The topics such as room acoustics and Helmholtz resonator are effective examples of first- and second-order systems, respectively. Acoustics along with vibrations can be part of core course material in design and machine dynamics. Acoustics with mechanical engineering applications such as noise control using source-path-receiver concepts, active control, noise control materials, product design, acoustical measurements, etc. can be included as elective courses. It is important to integrate acoustics in engineering curriculum because it gives the students an experience of the system effects and their impact on design. The integration of acoustics through both core and elective courses into mechanical engineering will be presented.

2:00

2pED3. A course in musical acoustics for nonengineering majors at the University of Hartford. Robert Celmer (Acoust. Prog. and Lab., College of Eng., Univ. of Hartford, 200 Bloomfield Ave., W. Hartford, CT 06117, celmer@mail.hartford.edu)

The University of Hartford's College of Engineering provides a service course entitled ''Introduction to Musical and Architectural Acoustics'' for nonengineering students majoring in such degree programs as music (at the Hartt School), technology fields, or the liberal arts. The material covers: acoustics terminology and concepts; a study of the science of each musical instrument family; sound radiation patterns of instruments and the implications for sound recording technology; the auditory system and hearing conservation; and an introduction to architectural acoustics related to such applications as band room and auditorium design. This presentation will describe some of the multi-media techniques for in-class presentation of the material as a means of better disseminating complex information using a sensory-rich environment. The extent to which this course fits into the assessment goals and general education science requirements of these nonengineering majors will also be discussed.

Contributed Papers

2:30

2pED4. Development of an acoustics curriculum for architectural engineers. Ralph T. Muehleisen (Dept. of Civil, Environ., and Architectural Eng., Univ. of Colorado, Boulder, CO 80309)

The University of Colorado is currently developing a series of acoustic classes for undergraduate Architectural Engineers and graduate students in the Building Systems Program of the Department of Civil, Environmental, and Architectural Engineering. The two classes currently undergoing development emphasize noise control in buildings and the design of acoustic venues. The goal of the classes is to teach the fundamentals of acoustics as needed by Architectural Engineers in order to design quieter buildings and interact with acoustic consultants. In the classes, the students are given both theoretical and practical problems which involve both analysis and design. In addition, students get hands on experience by taking measurements of real rooms using standard equipment.

2:45

2pED5. Should every physics student study acoustics? Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115)

Should every physics student learn about sound? Many of my colleagues say ''no;'' I say ''yes.'' My arguments for restoring acoustics to the physics curriculum will be presented.

TUESDAY AFTERNOON, 2 NOVEMBER 1999 MCKINLEY ROOM, 1:00 TO 3:45 P.M.

Session 2pMU

Musical Acoustics: African Musical Instruments and Traditions

David Avorgbedor, Cochair
School of Music, The Ohio State University, Columbus, Ohio 43210

James M. Pyne, Cochair
School of Music, The Ohio State University, Columbus, Ohio 43210

Invited Papers

1:00

2pMU1. Voiced noise: The ''heterogeneous sound ideal'' as preferred acoustic environment in selective sub-Saharan African instruments and ensembles. Daniel Avorgbedor and James Pyne (School of Music, Ohio State Univ., 110 Weigel Hall, Columbus, OH 43210-1170)

This paper draws on the African–American composer Olly Wilson's notion of a ''heterogeneous sound ideal'' in which the author discusses the common preference for certain sound combinations and timbres in African and African–American musical traditions. The purpose of this presentation is to demonstrate, in detail, specific aspects of this concept by drawing on a variety of instrumental and ensemble traditions from sub-Saharan Africa with focus on the manipulation and intentional modulation of timbres that characterize much of the musical traditions of sub-Saharan Africa and which are often overlooked by analysts. The paper argues that this common emphasis on ''unusual'' timbres constitutes one of the major parameters distinguishing sub-Saharan African musical traditions from those of European art music traditions. A ''heterogeneous sound ideal'' suggests new approaches to the identification, measurement, evaluation, and appreciation of acoustic phenomena in the contexts of sub-Saharan African musical traditions. Additional demonstrations involving spectral–timbral projections of individual and ensemble instruments will be presented.

1:30–1:45
Performance Demonstration

1:45

2pMU2. ''Greenotation:'' A system for representing African drum sounds and techniques. Doris Green (Pan African Performing Arts Preservation Assoc., Inc., 700 Southern Pkwy., Uniondale, NY 11553)

Within the past few decades technology has made it possible not only to write African music on paper like western notation, but also to retrieve and perform it from the printed page. In this paper I will present an overview of and demonstrate a system for writing African music I designed and tested over the past 20 years. Greenotation (after my name, Doris Green) was created for percussion instruments of Africa because Western musical notation could not notate the nuances and actions found in percussion music. Greenotation can notate the music of bells, rattles, drums, talking drums, sticks, stamping tubes, xylophones, hand clapping, and water drums; it is also able to represent dance movements that are integral to many sub-Saharan African drum and dance traditions.

2:15

2pMU3. African influences on jazz: A lecture–demonstration. Ted W. McDaniel (Div. of Jazz, School of Music, 110 Weigel Hall, Ohio State Univ., Columbus, OH 43210-1170)

This presentation focuses on selective aspects of sound patterns and performance traditions from sub-Saharan African societies that are retained in various ways in African–American jazz and performance styles. The talk will examine specific timbres and their related practices of repetition, collective improvisation, the acoustic phenomenon of pitch-bending, and texture. Recorded examples as well as live performances will highlight these elements and establish their relationships to the sub-Saharan African examples. A group of jazz musicians drawn from the OSU Jazz Division will perform specific items that further illustrate the concepts and practices outlined.

2:45–3:00
Performance Demonstration

3:00

2pMU4. The influence of African musical traditions on gospel music. Jan McCrary and Raymond Wise (School of Music, Ohio State Univ., 110 Weigel Hall, Columbus, OH 43210-1170, mcclary.5@osu.edu)

The paper examines historical and cultural influences of African music traditions on African–American gospel and spiritual music. Gospel music is one of several African–American genres of sacred music developed during the 1920's and 30's when musicians combined elements of blues and jazz music with church hymns and spirituals. The paper examines scholarly investigations of the impact of African music traditions on the development of gospel and spiritual music in America. In addition, the paper will explore unique African and American cultural influences on specific vocal-style characteristics. The session will feature The Ohio State University African American Music Chorale, which will perform representative short examples of African–American gospel and spirituals. The chorale also will present a short concert of complete works immediately following the session.

3:30–3:45
Performance Demonstration

TUESDAY AFTERNOON, 2 NOVEMBER 1999 GARFIELD ROOM, 1:30 TO 5:00 P.M.

Session 2pNS

Noise: Active Noise Control, Sound Quality and Noise Control Analysis

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

Contributed Papers

1:30

2pNS1. An inverse structure for active noise control or combined active noise control/transaural sound reproduction. Stephan Quednau and Martin Bouchard (School of Information Technol. and Eng., Univ. of Ottawa, 161 Louis Pasteur, Ottawa, ON K1N 6N5, Canada)

In this presentation, the use of inverse models of the acoustic plants is investigated for active noise control (ANC) systems. It is shown that the use of the inverse models combined with a predictor can: (1) greatly improve the convergence speed of stochastic gradient descent algorithms such as the multichannel filtered-X LMS or the modified filtered-X LMS and (2) significantly reduce the computational load of the resulting ANC system. Moreover, for systems combining active noise control and transaural sound reproduction (TSR), the reduction of the computational load will be even greater, since TSR systems implicitly need the use of the inverse models of the acoustic plants, and the proposed ANC inverse structure already performs this inverse filtering. For broadband feedforward ANC controllers, the drawback of using the typically noncausal inverse models of the acoustic plants is to increase the delay required between the reference signal(s) of the controller and the disturbance signal(s) to be reduced. In the case of feedback controllers, this additional delay would likely limit the application of the proposed structure to the control of periodic disturbances only.

1:45

2pNS2. Multichannel RLS algorithms and FTF algorithms for active noise control and sound reproduction systems. Martin Bouchard (School of Information Technol. and Eng., Univ. of Ottawa, 161 Louis Pasteur, Ottawa, ON K1N 6N5, Canada)

In the fields of active noise control (ANC) and transaural sound reproduction (TSR), multichannel FIR adaptive filters are extensively used. For the learning of such FIR adaptive filters, recursive-least-squares (RLS) algorithms are known to typically produce a faster convergence speed than stochastic gradient descent techniques, such as the basic least-mean-squares (LMS) algorithm or even the fast convergence Newton-LMS, gradient-adaptive-lattice (GAL) LMS and discrete-cosine-transform (DCT) LMS algorithms. In this presentation, multichannel RLS algorithms and multichannel fast-transversal-filter (FTF) algorithms are introduced, with the structures of some stochastic gradient descent algorithms used in ANC: the filtered-x LMS, the adjoint-LMS and the modified filtered-x LMS. The new algorithms can be used in ANC systems or for the deconvolution of sounds in TSR systems. Also, heuristic techniques are introduced, to compensate for the potential ill-conditioning of the correlation matrix in ANC or TSR systems, and for the potential numerical instability of the multichannel FTF-based algorithms. Simulations of ANC and TSR systems will compare the performance of the different multichannel LMS, RLS, and FTF-based algorithms.

2:00

2pNS3. Reduction of time-domain aliasing in adaptive overlap-add algorithms. Uwe Rass and Gerhard H. Steeger (Georg-Simon-Ohm Univ. for Appl. Sci., FB NF, P.O. Box 210320, D-90121 Nuremberg, Germany, Gerhard.Steeger@fh-nuernberg.de)

The adaptive overlap-add algorithm (OLA) is an attractive means for acoustical signal processing (e.g., filtering, compression), since it is computationally effective and provides magnitude and phase information. It performs well when the input data blocks, the DFT, and the filter impulse response are of suitably chosen lengths. If the filter frequency response is altered on-line by an adaptation rule formulated in the frequency-domain, the resulting impulse response will, in general, be too long. This results in serious distortions, due to time-domain aliasing during the inverse DFT. Limiting the filter length by a filter design procedure needs much computation time, which is inhibitive for many acoustical applications. Time-domain aliasing distortions are particularly disturbing when musical signals are processed, due to their nonharmonic nature. An algorithm is proposed which convolves the filter frequency response with a short window sequence, solely in the frequency domain. Thereby, the aliasing components can be reduced by a predictable amount. The computational burden is low (approximately 12% add-on to the basic OLA algorithm) and, in addition, can be traded with the attenuation of the aliasing components. Discrete prolate spheroidal sequences have proven to be the optimal window type for this purpose.

2:15

2pNS4. A method for evaluating automotive steering column noise. Allan K. Kennedy (Delphi Automotive Systems, 3900 Holland Rd., Saginaw, MI 48601) and William E. Niehoff (Automotive Eng. Management Services, Inc.)

Automotive component suppliers are often called on to supply components which meet very vague requirements such as no audible noise from component. As odd as this might seem, this type of specification usually has a number associated with it. This number could be the measured A-frequency weighted sound pressure level, the total loudness, some averaged band sound pressure level, or even a vibration response level. The irony comes from the fact that most of these components do not generate noises on their own but only when subjected to intense vibration. To deal with this, component suppliers must somehow construct a measurement system which will capture these squeaks and rattles as the component is vibrated. This paper describes a measurement system and method for evaluating the noises generated by automobile steering columns. Descriptions are given of the various vibration inputs used, the equipment required, and the parameters measured.

2:30

2pNS5. Modifications of a handheld vacuum cleaner for noise control. Gerald C. Lauchle and Timothy A. Brungart (Appl. Res. Lab. and Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804, GCL1@psu.edu)

Sound power measurements are described for a commercially manufactured, handheld vacuum cleaner. An objective assessment of the measured results is performed to identify the aeroacoustic sources of noise, suggested routes to noise control are then implemented, and a re-evaluation of the modified units is conducted and compared to the baseline results. The blade rate tone of the vacuum working fan is found to be the most annoying source, and it is reduced by up to 8 dB through a modification of the shroud that surrounds it. Unevenly spaced fan blades are also implemented as a noise control measure. This results in a comparable 8-dB reduction of the blade rate tone, but it introduces several new, side band tones to the sound power spectral data; the overall sound power increases slightly. Regardless, a subjective jury survey indicated that the modified unit having the unequally spaced fan blades is preferred 3:1 to the unit with the modified shroud only. [Work supported by Royal Appliance Manufacturing Co.]

2:45

2pNS6. Measurement of acoustic properties by the two-cavity method. Minor Nice (Owens Corning Testing Systems, 2790 Columbus Rd., Rte. 16, Granville, OH 43023-1200)

In this paper, results of the measurement of the infinite thickness properties (characteristic impedance, propagation constant) of acoustic materials by the two-cavity method will be presented. Results for some standard fiberglass materials will be shown along with comparison to some well-known correlations (Delaney and Bazley, Beranek). Results for materials other than fiberglass will also be presented. The impact of various measurement parameters (cavity spacing, averaging, impedance tube size, etc.) will be discussed relative to the measurement technique.

3:00–3:15 Break

3:15

2pNS7. Vibration analysis and design of wiper motors. Manuel Recuero, Juan Sancho, and Antonio Minguez (Departamento de I+D de Acustica, INSIA - UPM, Carretera de Valencia, Km. 7, 28031, Madrid, Spain, mrecuero@insia.upm.es)

This work has two objectives: (1) to characterize the manner of vibration of each part of a wiper motor; and (2) to find the sources of vibration so the noise emitted by the wiper motor will be known. Different measurement systems to analyze the vibrations of the wiper motor have been used. Results have been compared and the better system has been found. The results of the measurements have also been used to make some design recommendations for a new more silent wiper motor. This work started with four wiper motors. Two of them were characterized by the manufacturer of the motors as noisy and the other two as acoustically OK.

3:30

2pNS8. Holographic sound-field imaging as a diagnostic tool. Karl B. Washburn and Richard D. Godfrey (Owens Corning Testing Systems, 2790 Columbus Rd., Rte. 16, Granville, OH 43023-1200, karl.washburn@owenscorning.com)

Approaching its third decade, sound-field imaging using near-field acoustical holography has matured as an analysis technique. It has been established as the tool of choice for source motion mapping in structural acoustics and for source identification in noise control. The strength of near-field holography lies in its complete description of the sound field,

from source to far field. This opens a door to using sound-field imaging as an acoustical diagnostic tool. Several current trends in applying holography, including moving sources, reverberant spaces, and lower-cost systems, are surveyed. New diagnostic applications of sound-field reconstruction in audio-source imaging, transmission path analysis, and boundary material characterization are described.

3:45

2pNS9. Disk brake squeal analysis using the ABLE algorithm. G. Lou, T. W. Wu (Dept. of Mech. Eng., Univ. of Kentucky, Lexington, KY 40506), and Z. Bai (Univ. of Kentucky, Lexington, KY 40506)

Disk brake squeal noise is due to unstable friction-induced vibration. In order to study the stability of the disk system, the finite element method (FEM) is used to model a typical assembly that includes one rotor and two pads. Coulomb's friction law is applied at the contact interface between each pad and the rotor. Due to continuity of displacement in the normal direction, some degrees of freedom are condensed out. The FEM matrices of the dynamic system then become unsymmetric, which will yield complex eigenvalues. Any complex eigenvalue that has a positive real part will indicate an unstable mode. In real-world applications, the FEM model could include thousands of degrees of freedom. Fortunately, the resulting FEM matrices are sparse and a dynamic allocation scheme can be used to store only the nonzeros. Then a recently developed iteration method called ABLE (Adaptive Block Lanczos Method) is used to search only the complex eigenvalues with a positive real part in a certain user-specified frequency range. This algorithm has been shown to be very efficient and effective in predicting the unstable vibration modes.

4:00

2pNS10. New geometric sound absorbers. Glenn E. Warnaka (Future Technologies, L.L.C., 1612 South Allen St., State College, PA 16801)

This paper presents new concepts for simplified geometric sound absorbers that have a number of advantages over previous geometric sound absorbers and conventional sound absorbing materials. The new absorbers can be made of nearly any substance. Hence, the material for the absorbers can be selected on the bases of material costs, construction costs, and environmental considerations. They may be painted and cleaned without affecting their sound absorbing qualities. They also can be made very rugged for use in highway noise barriers and other outdoor applications. With the new geometric sound absorbers, it is possible to design absorbers that fulfill nearly any frequency range required. An infinite variety of designs is possible. In addition, the amount of sound absorption can be changed even after the sound absorbers have been installed. This may be of interest in multi-purpose rooms where the sound absorption can be altered for different functional uses of the room. The geometric sound absorbers may also be formed into flat treatments. Since the frequency range is a matter of design, high acoustic absorption can be maintained to frequencies well below 100 Hz, depending on design requirements, in treatments that are 6–16 mm thick.

4:15

2pNS11. Analysis of impact noise of nail ejection in wire nail machines. Jack Ding, Ahmed Al-Jumaily (Diagnostics and Control Res. Ctr., Auckland Inst. of Technol., Auckland, NZ), and Doug Wilson (Unitech Inst. of Technol., Auckland, NZ)

A wire nail production machine can be classified as one of the punch press machines. By nature, the production of wire nails involves the development of a series of force impulses, which in turn generate various sound pressure levels with different frequencies. Typically, the machine operations generating impact noise include pressing, cutting, ejecting, wire gripping and feeding. The punch pressing is the movement of punch header to press forming the nail head. It produces a large force impulse, which in turn generates a high level of impact noise and consequently high levels of ancillary impact noise such as the impact noise by the backlashes. It has been found that this type of impact noise is not necessarily the most annoying one. The impact noise generated by nail ejection, however, has been identified as one of the annoying noise components. The ejector used in a nail machine for executing the movement of nail ejection produces ringing noise at high frequencies when it is struck by the punch header in each cycle of the machine operations. This paper presents the analyses of and experiments on the ejection force impulse and the characteristics of the impact noise generated by nail ejection.

4:30

2pNS12. Acoustical analysis of infant/toddler rooms in daycare centers. Tom Frank (Dept. of Commun. Disord., Penn State Univ., 5-A Moore Bldg., University Park, PA 16802) and Matthew V. Golden (Penn State Univ., University Park, PA 16802)

The purpose of this ongoing research is to determine ambient noise levels, signal-to-noise ratios (SNRs), and RTs in daycare centers. To date, 248 measures of ambient noise levels have been obtained in occupied infant/toddler rooms of 11 daycare centers and 109 measures in 7 of the 11 daycares unoccupied. The mean occupied levels were 57 dBA and 66 dBC and the mean unoccupied levels were 38 dBA and 56 dBC. A 1/3 OB analysis revealed that the occupied levels were relatively constant (46–50 dB) from 63 to 2000 Hz while the unoccupied levels decreased as frequency increased especially after 125 Hz. Using several different speech spectrums compared with the occupied 1/3 OB levels, estimated SNRs ranged from -3 to 12 dB. Dosimeter measurements ($N=932$) revealed that the occupied daycares had an average peak sound level of 103 dB, high-threshold level of 62.5 dB, time-weighted average of 55.9 dB, and noise dose of 1.3%. Additional noise measurements and RTs are currently being collected. Overall, the unoccupied dBA levels and the estimated SNRs were less than recommended in the ASHA 1995 guidelines for classroom acoustics. [Work supported by PHS/NIH (1-R01-HD31540-01A2) Otitis Media, Behavior and Attention in Daycare.]

4:45

2pNS13. Prediction of cutoff noise in centrifugal fans. Zhichi Zhu, Song Li, and Dongtao Huang (Dept. of Eng. Mech., Tsinghua Univ., Beijing 100084, PROC)

The most important noise in centrifugal fans is aerodynamic noise, in which cutoff noise is the most main component. Cutoff noise is mainly caused by the fluctuating pressure on the cutoff in the volute impacted by the nonuniform flow from the exit of the rotating impeller. A numerical method is presented for prediction of cutoff noise in centrifugal fans. The contents of the method are mainly the following. (1) The fluctuating pressure serving as the source of cutoff noise is given out by using the existing computation software of 3D viscous flow in centrifugal fans and the time-frozen hypothesis. (2) The sound field in the volute of a centrifugal fan and the sound power levels of cutoff noise are computed by using the forth-order MacCormack scheme and making some important numerical measures. The sound field in the volute and the sound power levels of cutoff noise for three practical volutes with different cutoff clearance have been computed and the acoustic pressure distribution in space and change with time are discussed. Finally, the sound power levels of cutoff noise have been measured. The errors between the predicted and the tested sound power levels of cutoff noise are less than 3 dB. [Work supported by NSF.]

Session 2pPA

Physical Acoustics: Theoretical Physical Acoustics

R. Glynn Holt, Chair

Department of Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215

Contributed Papers

1:30

2pPA1. *A priori* assessment of the effect of small scales on sound radiation from a subsonic axisymmetric jet. Wei Zhao, Steven H. Frankel, and Luc Mongeau (School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907)

The effect of small scales on sound generated from a subsonic axisymmetric jet was investigated by filtering near-field data from a direct numerical simulation to determine the accuracy of the large eddy simulation technique for jet sound predictions. Lighthill's acoustic analogy was employed to predict the far-field sound. The direct numerical simulation results were in agreement with published results [Mitchell *et al.*, J. Fluid Mech. **383**, 113–142 (1999)]. A new approach to handle the large spatial extent of the Lighthill source term yielded predictions of the far field in good agreement with the simulation results at low frequencies even for shallow angles from the jet axis. It was found that the small scales have little effect on the low-frequency sound, which is dominant in this case. The levels computed from both the filtered and unfiltered computed sound pressure field were in good agreement with those predicted by direct computations. As expected, filtering removed the small scale fluctuations in the near field, thereby reducing the magnitude of the source term for the high-frequency sound. [Work supported by NIH DCO 3577-02, RO1 grant from NIDCD.]

1:45

2pPA2. Hot jet noise computed using large eddy simulation, computation of compressible free jet turbulence. David B. Schein (Northrop Grumman Corp. for UCLA/MAE Dept., 9HI 1/GK, 8900 E. Washington Blvd., Pico Rivera, CA 90660) and William Meecham (Univ. of California Los Angeles, Los Angeles, CA 90095)

A computational fluid dynamics model for free, heated jet flow and resultant far-field sound has been developed which uses large-eddy simulation (LES) and Lighthill's acoustic analogy. A deductive, subgrid scale model (based on a Taylor series expansion of the weighting function in the Favr E9 expansion) is used for the turbulent simulation. The simulation has been tested using published experimental mean flow field and rms fluctuation data [W. R. Quinn and J. Militzer, Phys. Fluids **31** (1988)] for a turbulent, free, square jet (known to be the same as a round jet of the same area). The ultimate objective is to address large Reynolds number, high subsonic (compressible) flow with realistic geometries, more representative of aircraft engine exhausts than can be considered using direct numerical simulation (DNS). In the simulation, Gaussian random velocity fields are introduced at the jet exit to excite the turbulence. The far-field sound and directivity is computed using the time-derivative form of Lighthill's source-integral result (formulated in terms of quadruple sources from the simulated flow field), which is integrated in time and contains the fluctuations set up by the time-varying stress tensor. Simulation for a WR 19-4 turbofan engine exhaust (Re-2×106 based on exit velocity and diameter) was performed, and propagated jet noise results compared with experimental acoustics data.

2:00

2pPA3. Calculation of second harmonic beam patterns for ultrasonic sources of arbitrary shape. Brian Landsberger (Caterpillar, Inc., P.O. Box 1875, Peoria, IL 61656-1875)

Ultrasonic transducers with nonaxisymmetric radiating surfaces are common in both industrial and medical applications. This presentation describes a numerical technique for calculating the second harmonic field produced in the field radiated by a transducer of arbitrary shape. The angular spectrum of the source is calculated with a two-dimensional Fourier transform, a weighted integral over the source spectrum is evaluated, and the inverse transform of the result yields the second harmonic pressure in any plane parallel to the source plane. Arbitrary absorption and dispersion are taken into account. The method is similar to that used by Alais and Hennion [Acustica **43**, 1–11 (1979)] to evaluate the field of a parametric array. In the present theory, however, we do not make use of the parabolic approximation. Beam steering and focusing are introduced by appropriate phasing of the source function. Calculations of radiation patterns produced by rectangular and other sources are presented.

2:15

2pPA4. Nonlinear modeling of focused acoustic fields. B. Edward McDonald (Saclant Undersea Res. Ctr., 19138 La Spezia, Italy) and William A. Kuperman (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

A series of simulations was performed to address the following question: If a circular piston projector were capable of producing finite amplitude beam pulses of arbitrary amplitude and bandwidth, what parameters would lead to the greatest concentration of energy within the smallest focal volume? The simulations were performed using the NPE time domain nonlinear acoustics model [B. E. McDonald and W. A. Kuperman, J. Acoust. Soc. Am. **81**, 1497 (1988)] adapted to azimuthal symmetry about the beam axis [G.-P. J. Too and J. H. Ginsberg, J. Acoust. Soc. Am. **91**, 59 (1992)]. For a given amplitude of the focusing wave, simulations were performed over a range of values for the parameter of nonlinearity and the effective piston aperture at the focus. Results show a strong maximum in focal gain as a function of the parameter of nonlinearity for each aperture considered. The maximum can be interpreted as competition between increasing amplitude at the piston and nonlinear defocusing as the wave propagates. Results may be rescaled to imply an optimum amplitude for a given parameter of nonlinearity. [Work supported by Saclantcen.]

2:30

2pPA5. Effects of small-amplitude fluctuations on thermoviscous shock wave structure. David G. Crighton and Pablo L. Rendon (Dept. of Appl. Mathematics and Theoretical Phys., Univ. of Cambridge, Silver St., Cambridge CB2 9EW, UK)

We study the low-diffusivity limit of the plane Burgers equation when small-amplitude fluctuations are introduced behind the shock region. Experiments in hydraulic jump propagation suggest that these downstream fluctuations can produce large-scale effects in the shock region. A self-consistent model is proposed in which the fluctuations themselves largely determine propagation over the mean field, and are thus permitted to amplify sufficiently so as to broaden the nonuniform part of the mean flow. By means of asymptotic matching we find an expression for the power amplification of any given spectral component in a mean field where a particular gradient is sought in the shock region. Shock thickening and profile distortion are predicted for certain values of fluctuation amplitude and frequency, and these results are confirmed numerically using a pseudospectral method.

2:45

2pPA6. Numerical solution of a statistical version of the Burgers equation. Penelope Menounou (Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712)

The Burgers equation (BE) is transformed into an unclosed set of linear equations that describe the evolution of the joint moments of the sound signal. The set of the equations is represented by a recursion equation termed Statistical Burgers Equation (SBE). Unlike for the BE, the time signal at the source is not required for the solution of the SBE and the power spectral density of the signal (or of a stochastic process) can be given instead [Menounou and Blackstock, J. Acoust. Soc. Am. **99**, 2539(A) (1996)]. The SBE is solved numerically by appropriately selecting only a finite number of equations from the infinite set, provided that the joint moments of the signal (or the stochastic process) are known at the source. The properties of the joint moments are presented for two source conditions: (i) a sinusoidal signal; and (ii) a Gaussian stationary and ergodic stochastic process. The finite difference scheme employed for the prediction of the joint moments' evolution is presented and stability criteria for the algorithm are derived. Finally, the sequence of computing the equations within the set is investigated and its effect on the stability of the numerical algorithm is demonstrated. [Work supported by the F. V. Hunt Postdoctoral Fellowship.]

3:00–3:15 Break

3:15

2pPA7. Mode counts for rooms and waveguides. Christopher L. Morfey, Matthew C. M. Wright, and Seong-Ho Yoon (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton SO16 1BJ, UK)

The density of modes in a room (3-D) or waveguide cross-section (2-D) is known to be approximated at high frequencies by expressions involving room volume and surface area, or (in 2-D) the area and perimeter of the waveguide cross-section. Deviations of the actual mode count from the smoothed approximation are studied numerically and analytically for simple shapes: rectangular (2-D and 3-D) and annular (2-D). Periodic clustering of eigenvalues, associated with cyclic rays [R. Balian and C. Bloch, Ann. Phys. **69**, 76–160 (1972)] is demonstrated at high frequencies. Expressions are given for estimating the standard deviation of 2-D and 3-D mode counts from their smoothed values, in a finite frequency band, when one dimension is much smaller than the others.

3:30

2pPA8. Band-gap engineering in a two-dimensional periodic system of fluids. M. S. Kushwaha (Inst. of Phys., Univ. of Puebla, P.O. Box J-45, Puebla 72570, Mexico, manvir@sirio.ifuap.buap.mx) and B. Djafari-Rouhani (Univ. of Sci. & Technol., 59655 Villeneuve D'Ascq, Cedex, France)

This work emphasizes that periodic binary systems can give rise to genuine acoustic band gaps (or stop bands) within which sound and vibrations remain forbidden. Extensive band structures for two-dimensional (2-D) periodic arrays of air cylinders in water background are computed. Complete, multiple, huge stop bands are found for both square and hexagonal lattices. The lowest stop bands are largest for a range of filling fraction $10\% \leqslant f \leqslant 55\%$, with a gap/midgap ratio of 1.8. The most interesting finding of the present investigation is that the low-frequency, flat passbands for a perfectly periodic system correspond to the discrete modes of a *single* airy cylinder. This almost exact correspondence is attributed to the low-filling fraction and the huge density contrast in air and water. It is stressed that such a simple inhomogeneous system as made up of air and water exhibits the largest stop bands ever reported for 2-D or 3-D elastic as well as dielectric (photonic crystals) composites. [Work partially supported by CONACyT Grant No. 28110E.]

3:45

2pPA9. Dispersion relations of surface waves in semi-infinite periodic scattering arrays. D. Caballero and J. Sánchez-Dehesa (Dept. Theoretical Condensed Matter., Autonomous Univ., 28049 Madrid, Spain)

Lord Rayleigh was the first in proposing the existence of surface waves in elastic systems at the end of the 19th century. Since then, people have observed elastic surface waves in very different contexts: from the huge wavelengthed seismic movements to the ultrasonic waves traveling along electronic devices. The aim of this work is to study the propagation of elastic surface waves through structures with periodically distributed inhomogeneities with cylinder symmetry. The theoretical study is based on variational tools developed by the authors. The periodicity led us to characterize the system by a band structure. In principle, those results can be used to design devices to forbid the transmission or to enhance and drive the propagation of elastic surface waves of selected frequencies. [Work supported by CICyT of Spain.]

4:00

2pPA10. Experimental determination of acoustic bands in two-dimensional sonic band gap crystals. J. Sánchez-Dehesa, D. Caballero (Dept. of Theoretical Condensed Matter, Autonomous Univ., 28049 Madrid, Spain, jsdehesa@uamca3.fmc.uam.es), R. Martinez-Sala, C. Rubio, J. V. Sánchez-Perez, F. Meseguer, and J. Llinares (Unidad Asociada CSIC-UPV, 28049 Madrid, Spain)

The dispersion relation of acoustic bands in sonic band gap crystals built up with periodic arrays of rigid cylinders in air have been characterized both experimentally and theoretically. A technique based on the analysis of the phase-shift experienced by the sound is used to construct the experimental acoustic dispersion relation. Measurements have been performed in square, triangular, and honeycomb lattices for different filling fractions. The experimental setup also allows to detect the so-called ''deaf bands'' [J. V. Sánchez-Pérez *et al.*, Phys. Rev. Lett. **80**, 3080–3083 (1998)]. A variational method is employed to calculate the corresponding acoustic bands. The good agreement between theory and experiment supports the experimental technique here described as an important characterization tool for these systems. [Work supported by CyCIT of Spain and Generalitat of Valencia.]

4:15

2pPA11. Band gap engineering in three dimensional system of air bubbles in water. M. S. Kushwaha (Inst. of Phys., Univ. of Puebla, P.O. Box J-45, Puebla 72570, Mexico, manvir@sirio.ifuap.buap.mx), B. Djafari-Rouhani, L. Dobrzynski (Dept. of Phys., Univ. of Science & Technology, Lille-I, 59655 Villeneuve D'Ascq, Cedex, France)

This work reports systematic and extensive evidence for the existence of complete, multiple, huge stop bands in the band structures for cubic arrays of air bubbles in water. All three important structures: face-centered cubic (fcc), body-centered cubic (bcc), and simple-cubic (sc) arrangements are investigated using the Fourier-series expansion (of position dependent density and elastic constant) method. It is noteworthy that this formulation does not require matching of the messy boundary conditions. The lowest stop bands are largest for a volume fraction $f \leqslant 10\%$, with a gap/ midgap ratio of 1.8, for all three geometries. It is found that the low-frequency, flat passbands for the perfectly periodic systems correspond to the discrete modes of a *single* bubble. This is an artifact of the low filling fraction and huge density contrast in air and water. It is stressed that such a simple inhomogeneous system as made up of air bubbles in water gives rise to the largest stop bands ever reported for elastic/acoustic as well as dielectric composites—save the similar 2D composites discussed in the preceding work. [This work was partially supported by CONACyT grant No. 28110E.]

4:30–4:40 Break

Contributed Poster Papers

Papers 2pPA12 and 2pPA13 will be presented in poster format. Authors will be at their posters from 4:40 p.m. to 5:10 p.m.

2pPA12. Stop bands of tri-dimensional sonic band gap crystals. P. Ribaute, D. Caballero, and J. Sánchez-Dehesa (Dept. of Theoretical Condensed Matter, Autonomous Univ., 28049 Madrid, Spain)

Acoustic and ultrasonic waves are powerful probes for many macroscopic systems ranging from the oceanic environment to devices under mechanical stress. From the technological point of view, tools must be developed in order to understand the propagation of acoustic waves through inhomogeneous media. In this work, a variational method is developed to study the acoustic response of three-dimensional systems with periodically distributed scatterers. The dispersion relations of the corresponding acoustic bands are obtained. They determine the stop bands. Also, it is shown that symmetry plays an important role in the full problem of transmitting acoustic waves through a periodic structure. If the symmetry of the external wave does not match with the band at its corresponding frequency, the acoustic energy cannot be transmitted. The existence of these bands has been previously reported in two-dimensional sonic crystals [J. V. Sánchez-Pérez *et al.*, Phys. Rev. Lett. **80**, 3080–3083 (1998)]. [Work supported by CICyT of Spain.]

2pPA13. A novel time-domain method to study the propagation of acoustic waves across composite media. D. Bosquetti, J. Sánchez-Dehesa, and D. Caballero (Dept. of Theoretical Condensed Matter, Autonomous Univ., 28049 Madrid, Spain, jsdehesa@uamca3.fmc.uam.es)

It has been previously shown that the symmetric-split-operator technique is a useful method to study evolution problems in classical mechanics. In this work the technique is applied to obtain a finite-difference scheme which allows us to analyze the propagation of acoustic waves in time domain. This scheme is developed for a general three-dimensional composite system. Numerical results are presented here for some one-dimensional structures. [Work supported by UAM and CICyT of Spain.]

TUESDAY AFTERNOON, 2 NOVEMBER 1999 UNION E ROOM, 1:30 TO 4:35 P.M.

Session 2pPP

Psychological and Physiological Acoustics and Speech Communication: Honoring the Contributions of Robert C. Bilger: Bilger and Better Science

Lawrence L. Feth, Cochair
Speech and Hearing Science, The Ohio State University, 110 Pressey Hall, 1070 Carmack Road, Columbus, Ohio 43210-1372

Walt Jesteadt, Cochair
Boys Town National Research Hospital, 555 North 30th Street, Omaha, Nebraska 68131

Chair's Introduction—1:30

Invited Papers

1:35

2pPP1. A Bilger journey. Ira J. Hirsh (Central Inst. for the Deaf, 818 South Euclid, St. Louis, MO 63110)

Bob Bilger started (1954) his postdoctoral appointment at Central Institute with Ira Hirsh. The topic was recovery of auditory threshold for tones, following exposure to tones. They confirmed the results of Hirsh and Ward (1952) on recovery of click thresholds after similar exposures. The collaboration continued with papers on masking, remote masking, and additivity of different kinds of masking. The organization at Central Institute was such that he, like other research associates, learned about and participated in work in other laboratories. He particpated in physiological studies; he branched out to time and speech; and his statistical expertise was much sought in clinical research. His move to Pittsburgh allowed him to continue his disregard for disciplinary barriers and his readiness for clinically related matters. Just before leaving Pittsburgh for Illinois, he undertook to write, organize and edit the impressive, monumental, first serious study of the effect of a single-channel cochlear implant on adult human subjects [Bilger *et al.* (1977)]. The Illinois stretch demonstrates that (1) speech and psychoacoustics are not very far apart, and (2) very successful Chairs of Speech and Hearing Departments need not be clinicians.

1:55

2pPP2. Origins of the IWAIF model. Lawrence Feth (Dept. of Speech & Hearing Sci., Ohio State Univ., Columbus, OH 43210, feth.1@osu.edu)

The intensity weighted average of instantaneous frequency (IWAIF) model evolved from early work on the processing of frequency-modulated (FM) tones suggested by Bob Bilger [Feth *et al.*, J. Acoust. Soc. Am. **45**, 1430–1437 (1969)]. This early work on FM signals led to the incorporation of Voelker's Unified Theory of Modulation [Proc. IEEE **54**, 340–353 (1966)] into the original envelope-weighted average of instantaneous frequency (EWAIF) model [L. L. Feth, Percept. Psychophys. **15**, 375–379 (1974)]. This talk will describe the development of the IWAIF model and describe its application in auditory signal processing such as formant tracking in speech processing algorithms and the perception of Doppler-frequency shifts produced by moving sound sources.

2:15

2pPP3. Variability in forward masking and intensity discrimination. Walt Jesteadt, Jason F. Reimer, and Huanping Dai (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, jesteadt@boystown.org)

Jesteadt and Bilger [W. Jesteadt and R. C. Bilger, J. Acoust. Soc. Am. **55**, 1266–1276 (1974)] noted greater individual differences for frequency discrimination than for intensity discrimination and summarized this effect in terms of analyses of variance. In this paper, we report similar comparisons for forward masking and intensity discrimination. Four subjects were tested in a 2IFC adaptive procedure. The forward masker and the pedestal in the intensity-discrimination task were broadband stimuli consisting of 18 components with third-octave spacing from 200 to 10 000 Hz. Thresholds were obtained for forward maskers from 20 to 80 dB SPL and for pedestals from 10 to 80 dB SPL in 10-dB steps, with eight 50-trial blocks per condition. When the results were analyzed in terms of signal power at threshold for forward masking and intensity discrimination, they showed smaller individual differences and less variability across repeated measurements in forward masking than in intensity discrimination. Psychometric functions reconstructed from the adaptive tracks were steeper for forward masking than for intensity discrimination. Thresholds obtained in forward-masking tasks are often considered to be highly variable, but may be more stable than those obtained in studies of intensity discrimination using comparable stimuli. [Work supported by NIDCD.]

2:35

2pPP4. Physiological mechanisms of frequency discrimination. Eric Javel (Dept. of Otolaryngol., Univ. of Minnesota, Minneapolis, MN 55455)

I recall a conversation with Bob Bilger that occurred around 1971, when I was a graduate student in Bioacoustics at Pitt. He said (translated into English from the original Bilgerian), ''Did it ever occur to you that you can apply principles of signal detection theory in physiological studies? That way, you could examine neural and psychophysical performance using comparable measures and possibly shed light on underlying mechanisms.'' This idea, which predated its implementation in auditory research by at least 10 years, was initially lost on me. Much later, I recognized the possibilities and performed some pertinent experiments with various colleagues. We utilized 2AFC adaptive tracking procedures to investigate performance limits of single cat auditory nerve fibers and neural populations on pure-tone frequency discrimination tasks. For single-fiber responses, we found that: (1) decision strategies based on detecting differences in spike counts fail to account for perceptual findings by wide margins; and (2) strategies based on detecting differences in phase-locked activity produce data that match perceptual findings well at low frequencies but rapidly degrade at frequencies >1500 Hz. Examining neural population performance using a stochastic excitation pattern model, we found that a strategy which simply estimates the spatial location of the response peak accounts well for perceptual performance at most frequencies. See, Bob? Your idea works, and I WAS paying attention.

2:55

2pPP5. Development of a screening version of the communication profile for the hearing impaired. Marilyn E. Demorest (UMBC, 1000 Hilltop Circle, Baltimore, MD 21250), David J. Wark (Univ. of Memphis, Memphis, TN 38105), and Sue Ann Erdman (Baltimore, MD 21286)

The 163-item CPHI provides a diagnostic profile of scores on 25 scales that describe a client's adjustment to hearing impairment. Factor structure of the instrument shows that two important factors it assesses are communication performance (an aspect of hearing disability) and psychosocial adjustment to hearing impairment (an aspect of handicap). The goal of this study was to develop a brief instrument, to be used in conjunction with standard audiometric assessment, to screen for disability and handicap. A pseudo-random sample of 1000 cases was drawn from a large, heterogeneous clinical database. Item response theory was used to derive item characteristic curves, and item selection was based primarily on item discrimination. Nine items were chosen to screen for communication performance, and 11 were chosen to screen for psychosocial adjustment. Pass/Fail criteria were developed, and sensitivity and specificity were evaluated in a holdout sample of 319 cases.

3:15

2pPP6. Forward masking recovery and peripheral compression in normal-hearing and cochlear-impaired ears. David A. Nelson and Anna C. Schroder (Dept. of Otolaryngol., Univ. of Minnesota, 396 UMHC, 516 Delaware St. S.E., Minneapolis, MN 55455, dan@tc.umn.edu)

Iso-response temporal masking curves are obtained from normal-hearing subjects at a probe frequency of 1000 Hz for masker frequencies between 500 and 1200 Hz. Time constants calculated from the temporal masking curves varied with masker frequency, from around 70 ms for low off-frequency maskers (500–600 Hz) to around 36 ms for on-frequency maskers (close to the probe frequency). Continuing Bilger's earlier pursuits of nonlinearities in hearing, estimates of peripheral compression were calculated under the assumption that the response to a low off-frequency masker is linear at the probe frequency place. Average compression exponents varied from close to 1.0 for remote off-frequency maskers (both below and above the probe) to below 0.4 for on-frequency maskers. Input–output transfer functions derived from the compression exponents were consistent with BM transfer functions recorded in animals with normal cochlear function. Comparisons of iso-response temporal masking curves in subjects with sizable cochlear hearing losses at the probe frequency yielded linear transfer functions consistent with BM data from cochlear-damaged animals. It is concluded that time constants for recovery from forward masking in ears with cochlear hearing loss are no different than those obtained from normal-hearing ears, once differences in peripheral compression are taken into account. [Work supported by NIH-NIDCD Grant DC00149 and the Lion's 5M International Hearing Foundation.]

3:35

2pPP7. Selected aspects of across-frequency processing in binaural hearing. Constantine Trahiotis, Leslie R. Bernstein (Surgical Res. Ctr., Dept. of Surgery (Otolaryngol.) and Ctr. for Neurological Sci., Univ. of Connecticut Health Ctr., Farmington, CT 06030), and Richard M. Stern (Carnegie Mellon Univ., Pittsburgh, PA 15213)

While considering the many contributions of Dr. R. C. Bilger to knowledge concerning auditory processing, it seemed both fitting and appropriate to honor him by discussing research in binaural hearing that is consistent with key aspects of what we perceive to be his style. Accordingly, the presentation will highlight the integration of theory, measurement, and empirical observation in selected aspects of across-frequency processing in binaural hearing. The topics addressed will be: (1) lateralization as a function of bandwidth, center frequency, and interaural time/phase disparities; (2) binaural interference; (3) the incorporation of peripheral compression, rectification, and low-pass filtering in an index that accounts for binaural detection across frequency.

2p TUE. PM

3:55

2pPP8. Context effects with limited spectral information. Melanie L. Matthies (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215)

Speech recognition has been established and unified as a construct through the systematic study of its domain [Bilger, 1984]. To investigate the effect of limiting spectral information, items from the Revised Speech Perception in Noise (SPIN) test [Bilger *et al.*, 1984] were impoverished to include two, three, or four channels of spectral information [cf. Shannon *et al.*, 1995] and administered to 62 normal-hearing, young adults. Subjects also identified consonants that likewise had been spectrally limited via signal processing. When consonant confusion matrices were analyzed, transmission of place of articulation was strongly affected by spectral limitations followed closely by effects on manner perception, while voicing and resonance (oral/nasal) cues were relatively robust. High-context (HC) SPIN items were significantly easier for the subjects to identify than low-context (LC) items for the four-channel sentences. The context advantage decreased for the two-channel task because listeners were unable to utilize the sentence cues. Across channel conditions, a regression line fit to normalized HC by LC data had a slope of 0.9. Reduced semantic and syntactic information in LC items, therefore, contributed to a perceptual uncertainty network (PUN) when spectral information was limited and this SPIN PUN should be explored further. [Work supported by MURI Grant Z883402.]

4:15

2pPP9. Modeling closed-set phoneme and open-set word recognition by multi-channel cochlear implant users. Ted A. Meyer, Mario A. Svirsky (Indiana Univ. School of Medicine, Dept. of Otolaryngol., Indianapolis, IN), Stefan Frisch (Univ. of Michigan, Ann Arbor, MI), Adam R. Kaiser, David B. Pisoni, and Richard T. Miyamoto (Indiana Univ. School of Medicine, Indianapolis, IN)

There has been phenomenal growth in research on speech perception by cochlear implant (CI) users since the printing of the ''Bilger Report'' [Bilger *et al.*, Ann. Otol. Rhinol. Laryngol. **86** (S38), 1–176 (1977)]. Undoubtedly, average speech perception performance with these devices has improved dramatically. However, despite advances in implant technology, CI users continue to demonstrate a wide range in the ability to perceive speech. Little progress has also been made in understanding how CI users actually perceive speech. Although many correlational analyses have been carried out, little research has focused on mechanisms of speech perception, and many clinical decisions are made on a trial-and-error basis. Using a new approach, we have developed a quantitative, psychophysically based model (Multidimensional Phoneme Identification, MPI) of phoneme perception by CI users [M. A. Svirsky and T. A. Meyer, J. Acoust. Soc. Am. **103**, 2977 (1998)]. The MPI model generates phoneme confusion matrices from performance on psychophysical tasks. In a complimentary line of work, we aim to predict open-set spoken word recognition from phoneme performance [S. Frisch and D. B. Pisoni, Res. Spoken Lang. Proc. 261–288 (1998)]. In both lines of research, we have found predictable relations between perception of phonetic features, phonemes, and words by CI users. [Work supported by NIH, AAO-HNS, DRF, NOHR.]

Session 2pSA

Structural Acoustics and Vibration: Characterization of Structural Properties

Jerry H. Ginsberg, Chair

School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, Georgia 30332-0405

Contributed Papers

1:15

2pSA1. Measurements of the dynamic elastic moduli of viscoelastic materials with micro-inclusions. R. Lance Willis, Lei Wu, and Yves H. Berthelot (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

An indirect method has been proposed [see J. Acoust. Soc. Am. **102**, 3549–3555 (1997)] to determine the complex, frequency-dependent, elastic moduli of polymers containing microscopic inclusions from laser-based measurements and finite element modeling. The method consists in measuring the surface dynamics of the sample under harmonic excitation by noncontact laser Doppler interferometry. The experimental results are then matched with numerical predictions in which the moduli are the adjustable parameters. The method is first validated at ambient pressure by measuring the dynamic moduli of a sample of known properties. Results obtained with samples of the same voided viscoelastic material but with different aspect ratios are presented. To assess the effect of static pressure on the moduli, the sample is placed inside a static pressure air chamber (0–500 psi) and the effect of pressure cycling is investigated by measuring the moduli during the first pressure cycle (0–500, and 500–0 psi) and after eight pressure cycles. [Work supported by ONR, code 334.]

1:30

2pSA2. Determination of the complex shear modulus from torsional waves in viscoelastic bars. Jacek Jarzynski (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, jacek.jarzynski@me.gatech.edu) and John W. Doane (Georgia Inst. of Technol., Atlanta, GA 30332)

The torsional wave method for measurements of shear modulus, described by Garrett [J. Acoust. Soc. Am. **88**, 210–221 (1990)], is modified by the addition of metal end pieces to the sample. This allows measurements of the shear modulus at low frequencies (100–1000 Hz) for small samples. Measurements can be made with the sample either air-loaded or water-loaded. The system is calibrated using a laser vibrometer. Data will be presented for neoprene rubber and compared with measurements of Young's modulus using a resonant rod technique.

1:45

2pSA3. Identification of multi-degree-of-freedom nonlinear vibratory systems consisting of unknown elastic forces. Christopher M. Richards (Caterpillar, Inc., 100 N. E. Adams St., Peoria, IL 61629-9760, richards@iaonline.com) and Rajendra Singh (The Ohio State Univ., Columbus, OH 43210)

Nonlinear system identification techniques often require *a priori* knowledge of the nature and mathematical form of the nonlinearities. Unfortunately, for practical systems, this is not always possible. As a result, nonlinearities are often approximated and questions remain as to whether an accurate model can be determined. In addition, under experimental conditions, the amount of measurement noise present in the identification process must also be quantified. To address these issues, identification of discrete systems consisting of nonlinear elastic forces is examined in the presence of uncorrelated noise. It is assumed that the mathematical form of the nonlinearities is unknown but can be approximated by polynomials. Coherence functions are introduced which are based on a ''reverse path'' spectral approach developed by the authors for multi-degree-of-freedom systems. These coherence functions, as calculated from conditioned spectra, indicate the extent of uncorrelated noise present and the accuracy of the assumed mathematical models. Using several example simulation systems, including a system with a continuous nonlinearity described by a noninteger exponent, both temporal and spectral identification techniques are employed to study the issues described above.

2:00

2pSA4. Experimental characterization of nonlinear rubber isolators in a multi-degree-of-freedom system configuration. Christopher M. Richards (Caterpillar, Inc., 100 N. E. Adams St., Peoria, IL 61629-9760, richards@iaonline.com) and Rajendra Singh (The Ohio State Univ., Columbus, OH 43210)

Experimental characterization has been investigated for three rubber isolators placed in a multi-degree-of-freedom system configuration. The configuration consisted of a rigid mass mounted to a flexible support beam via each of the rubber isolators. Random, sinusoidal, and sine sweep excitations were applied to the rigid mass and accelerations were measured. Data revealed the dependence of the isolators' properties on amplitude and type of excitation. Modeling based on the continuous system theory resulted in quasi-linear representations of the experimental system. Discrete system models consisting of nonlinear polynomial equations for describing the nonlinearities of the isolators were also developed. In addition, sound pressure levels were measured during sine sweep excitation for one of the isolators. Increased sound levels were observed at off-resonant regions of the vibration response. Finally, single-degree-of-freedom static and dynamic experiments were conducted. Discrepancies in the isolators' properties were found between the single- and multi-degree-of-freedom experimental results.

2:15

2pSA5. Simple nondestructive quantitization of specially orthotropic materials. Curt Preissner and Thomas J. Royston (Univ. of Illinois at Chicago, MC 251, 842 W. Taylor St., Chicago, IL 60607, troyston@uic.edu)

Many different plate structures can be classified as specially orthotropic, such as cross-ply composites, unidirectional composites and quarter-cut wood boards. A simple method of determining the four elastic constants of a completely free, specially orthotropic plate has been developed. An improved Rayleigh expression, utilizing beam functions, is used to represent the deflection shape of the plate. A dimensionless objective function is formed from the frequency equation. The material properties are determined by minimizing the objective function with respect to modal analysis and geometry data. The technique has been successfully used on aluminum, composite and wooden plates. The ease of the method lends itself to use in NDE of composite plates and qualification of wood used in the construction of musical instruments.

2:30

2pSA6. Maximum-likelihood estimation of wave components for structural vibrations. Peter J. Halliday and Karl Grosh (Dept. of Mech. Eng. and Appl. Mech., Univ. of Michigan, Ann Arbor, MI 48109-2125)

In the analysis of structural acoustic systems, it is desirable to have a robust method for estimating the wave components (complex wave numbers and the corresponding wave amplitudes) of the structural response. A technique for evaluating steady-state wave propagation on single and multiply connected beam-like structures from noisy data is developed using previous modifications of Prony's method [K. Grosh and E. G. Williams, J. Acoust. Soc. Am. **93**, 836–848 (1993)] and a new maximum-likelihood scheme. An overdetermined exponential solution is fit to evenly spaced data points from which propagating and evanescent wave numbers and their associated amplitudes are determined for each frequency of interest. *In situ* estimation of the elastic modulus may be accomplished by using a least-squares fit of the analytic solution of the Timoshenko beam theory dispersion relationship to the dispersion curve estimated from experimental data. Structural intensity is evaluated from the wave component and material property information. Numerical experiments have shown the effectiveness of these methods even for relatively low signal-to-noise ratios, on the order of 30 dB.

2:45–3:00 Break

3:00

2pSA7. Sensitivity of resonance-frequency shifts to defect location in adhesive-bonded joints. Deborah Hopkins, Seiji Nakagawa, Kurt Nihei (MS 46A-1123, Eng. Div., Lawrence Berkeley Natl. Lab., 1 Cyclotron Rd., Berkeley, CA 94720, DLHopkins@lbl.gov), and Guillaume Neau (Univ. of Bordeaux1, Bordeaux, France)

Techniques based on frequency shifts and mode-shape analysis are being investigated to determine their feasibility for characterizing defects in adhesive-bonded joints in automotive structures. It is well known (Rayleigh–Ritz derivation) that introduction of a crack-like defect into a structure reduces its stiffness and results in a corresponding downward shift in resonance frequencies. Experimental and modeling results show that analysis of resonance-frequency shifts is much more complicated for bonded joints where the defect consists of a gap in the adhesive layer. Structures containing a defective joint sometimes exhibit higher-resonance frequencies than a structure with an undamaged joint. Such results have been observed in laboratory experiments on aluminum plates with adhesive-bonded T joints, and in finite-element simulations for a variety of structures. In all cases, the defects studied are gaps in the adhesive layer of the joints. While the reduction in mass associated with a gap in the adhesive layer is very small, the mass effect overwhelms the frequency-decreasing effect of the reduced stiffness when the defect is located in a low-stress region of the joint. Thus, the direction and magnitude of frequency shifts depend on the resonance mode and the location of the defect.

3:15

2pSA8. Thermoelastic effects on surface acoustic wave propagation. Zhongyu Yan and Peter B. Nagy (Dept. of Aerosp. Eng. and Eng. Mech., Univ. of Cincinnati, Cincinnati, OH 45221)

The effect of thermal stresses on ultrasonic surface wave propagation was investigated. Quasi-static and dynamic thermal stresses and deformations were considered in materials with and without fatigue cracks. The perturbation of ultrasonic surface waves due to these thermal effects were studied both theoretically and experimentally. A long-duration infrared laser pulse was used to irradiate a series of aluminum and titanium specimens with fatigue cracks of known sizes between 0.5 and 1.0 mm. Preliminary numerical results from finite element simulation were found to be in good agreement with the experimentally observed laser-induced modulations of the ultrasonic surface wave. In particularly, both the numerical and experimental results indicated the same characteristic transient behaviors of the thermo-optical modulations for cracked and intact materials. The different behaviors suggest a new promising method for ultrasonic nondestructive evaluation, which cannot only effectively distinguish fatigue cracks from other artifact scatters, but potentially can also provide a method for quantitative evaluation of crack features. [This effort was sponsored by the Defense Advanced Research Project Agency (DARPA) Multidisciplinary University Research Initiatiative (MURI), under Air Force Office of Scientific Research Grant No. F49620-96-1-0442.]

3:30

2pSA9. A definition of a loss factor is unique, some definitions are not. G. Maidanik (Carderock Div., Naval Surface Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817-5700)

Noise control goals are often specified in terms of ratios of quadratic response quantities, among them a loss factor. Thus, the loss factor (η) is defined as the ratio of the external input power density (Πe) and the product of the stored energy density (E) and the frequency (ω), i.e., $\eta = [\Pi e/(\omega E)]$. A unique definition of the loss factor demands that the dynamic system be enclosed so that (Πe) accounts for all the external input power density and (E) accounts for all the stored energy density generated by (Πe). Recently a number of acousticians in seeking noise control goals have defined the goals in terms of ratios that are assigned as loss factors. The definitions of these loss factors do not always conform to that uniquely defined loss factor. Those who maintain the conservative definition of a loss factor have been puzzled by reports of loss factors with values that exceed unity by several orders of magnitude. This paper attempts to examine a few examples of these less unique definitions of a loss factor and to show that a return to the unique definition is mandatory if the concept of a loss factor is not to be lost.

3:45

2pSA10. Acoustical forced oscillation nondestructive testing method for meshing gears. Leonid M. Gelman (Dept. of Nondestructive Testing, Natl. Tech. Univ. of Ukraine, 37, Peremogy pr., Kiev, 252056 Ukraine) and Alexandr S. Iievlev (SPU ''Slavutich,'' Kiev-49, Ukraine)

Nondestructive testing and evaluation method of gears is considered. Differential equations of meshing gears forced oscillations are considered with time-variant piece-constant gear mesh stiffness and exciting force. New expressions of spectral density of mentioned oscillations are received for two cases: with defect and without one. For the first case modulation of gear mesh stiffness and exciting force is considered.

Session 2pSC

Speech Communication: Speech Processing (Poster Session)

Ashok K. Krishnamurthy, Chair
Department of Electrical Engineering, The Ohio State University, Columbus, Ohio 43210

Contributed Papers

To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 4:30 p.m. To allow for extended viewing time, posters will be on display from 9:00 a.m. to 10:00 p.m.

2pSC1. Telephone speech enhancement for elderly hearing-impaired listeners. Amy E. Sheffield, Ashok Krishnamurthy (Dept. of Elec. Eng., The Ohio State Univ., Columbus, OH 43210, sheffield.20@osu.edu), Lawrence Feth, Stephanie Davidson, Evelyn Hoglund, and Lynette Roth (The Ohio State Univ., Columbus, OH 43210)

Many elderly persons with high-frequency hearing loss find telephone use frustrating due to lower intensity levels and reductions in acoustical information that can be useful in deciphering speech. The purpose of this project is to pre-process the speech signal before it is sent over the phone line and provide speech enhancement without the use of amplifying handsets or hearing aids at the receiving end. The enhancement technique takes into account the limited bandwidth of the phone line as well as the hearing characteristics of the user. Two pre-processing schemes, a single channel and a double channel approach, used to increase the intelligibility of speech in these situations are discussed. The single channel method performs amplitude compression of the entire signal. The two-channel method filters the incoming signal into high-frequency and low-frequency channels and performs independent compression on each before recombination. Results comparing the two speech enhancement schemes against no processing for a group of elderly hearing-impaired subjects are presented. [Work supported by a grant from the Franklin County Office on Aging.]

2pSC2. High-frequency transfer function model of the vocal tract sections using FEM matrix condensation and transfer matrix techniques. Samir El-Masri and Nobuhiro Miki (Grad. School of Eng., Hokkaido Univ., North 13, West 8, Kita-ku, Sapporo, 060-8628 Japan, selmasri@cho8-ei.eng.hokudai.ac.jp)

Using numerical tools such as the Finite Element Method (FEM) for acoustical modeling of the vocal tract would require a huge number of nodes and elements. In this paper, the matrix condensation and transfer matrix techniques [A. Craggs, J. Sound Vib. **132**, 393–402 (1989)] adapted to a three-dimensional model of the vocal tract section analysis will be introduced and discussed. Using these new FEM techniques, which use only the nodes at the input and output of the structure, the transfer function can be computed with a very small matrix comparing to standard FEM. The second goal of this research is to attempt to discover some geometrical parameters which could influence the transfer function. This investigation could be useful to make a new electrical vocal tract model which would consist of cascade transfer functions controlled by parameters. The study of the transfer functions will be carried out in low and high frequencies. [Work partly supported by Japanese CREST project.]

2pSC3. Getting two birds with one phone: An acoustic sensor for both speech recognition and medical monitoring. James D. Bass, Michael V. Scanlon, and Thomas K. Mills (Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, jbass@arl.mil)

Automatic speech recognition (ASR) in noisy environments requires innovative use of disparate technology to overcome the special demands caused by multiple speakers and minimal signal-to-noise ratios (SNRs). Use of nonairborne acoustic sensors for ASR imposes special requirements on speech engines due to the changes in spectral information caused by alternative pickup locations. Specifically, the relative power of voiced and nonvoiced components is often reversed when compared to the relative power of these components collected with conventional microphone technology. Our research entails the evaluation of various ASR sampling configurations in conjunction with different body location points for the physiological sensor. Physiology provides clues to speakers' stress or cognition. For ASR, our goal is to build a suite of optimal sampling configurations for several strategic body locations (e.g., throat temple, thorax, etc.). The experimental design includes traditional word error rate and subjective task completion components. These experiments were conducted in environments with SNR ranges of 10, 3, 0, and -1 dB. These SNR ranges cover the optimal commercial ASR environment of 10 dB to ranges where commercial ASR systems with conventional microphone technology are completely ineffective. Pilot studies indicate good performance below 0-dB SNR for a sensor that is throat located.

2pSC4. Estimation of articulatory movement and its application to speech synthesis. Jun Huang and Stephen Levinson (Dept. of ECE, Univ. of Illinois, Urbana, IL 61801, jhuang@ifp.uiuc.edu)

In this paper, two methods are investigated to estimate the movement of the vocal tract modeled by a three-dimensional articulator described by seven physical parameters. First, we use the cubic spline method to interpolate the articulatory parameters between consecutive phonemes. Second, the articulatory parameters are piecewise linearly interpolated and then passed by a cubic spline smoothing filter. The cubic spline smoothing filter is a low-pass filter with the maximum flatness property. The filter bandwidth can be easily adjusted by a control parameter. A graphical animation is built to visualize the articulatory movement of phonetic strings and to compare the results from the two different schemes. Some English phonemes are synthesized based on the articulatory model and the governing Webster equation in frequency domain considering viscous losses. An overlap-and-add method is used to convert the frequency domain spectrum to time domain speech signal. Finally, some example of the synthesized speech phonemes will be demonstrated. [We want to thank the National Science Foundation (NSF) for the support of our work.]

2pSC5. Vocal-tract length normalization for acoustic-to-articulatory mapping using neural networks. Sorin Dusan and Li Deng (Dept. of Elec. and Computer Eng., Univ. of Waterloo, Waterloo, ON N2L 3G1, Canada, sdusan@speech2.uwaterloo.ca)

A new method of estimating the overall vocal-tract (VT) length and the normalization of acoustic parameters of different speakers is reported in this paper for acoustic-to-articulatory mapping. The main goal of this work was a high accuracy of VT length estimation from a short speech utterance. An articulatory model, originally developed by Maeda, was used as a reference female VT. Linear scaling was used to synthesize training data for VT lengths between 100% and 125% of the reference VT length (14.96 cm). These data had 250 utterances, resulted from different VT lengths, each containing six vowels. A neural network with two hidden layers was trained using vectors of 10 mel-frequency cepstrum coefficients and the corresponding VT lengths of these utterances. For the same VT length range, similar test data were synthesized using the training vowels but in different contexts. With the trained network, evaluation of this method on test data has shown an average error of less than 1% and a maximum error of 3.2% in estimating VT length from single test utterances. Frequency warping was used to normalize the cepstrum parameters according to estimated length factors ranging between 1.0 and 1.25. [This work was supported by NSERC.]

2pSC6. Automatic ToBI prediction and alignment to speed manual labeling of prosody. Ann K. Syrdal, Julia Hirschberg (AT&T Labs–Res., Florham Park, NJ 07932, syrdal@research.att.com), and Julie T. McGory (Ohio State Univ., Columbus, OH 43210)

ToBI (Tones and Break Indices) prosodic labeling of a speech corpus is a slow, labor-intensive process that typically takes from 100 to 300 times real time, even with experienced labelers. An experiment was conducted to determine: (1) whether manual correction of automatically assigned ToBI labels would speed up the labeling process; and (2) whether default labels introduced any bias in label assignment. A group of four graduate linguistics students previously trained in ToBI labeling were paid participants in the study. A large speech corpus of one female speaker reading several types of texts was labeled over a period of nine months. Half of each recording was labeled in the normal fashion ''from scratch'' without default labels, and the other half was presented with preassigned default labels for labelers to correct. Default ToBI labels were predicted from text using techniques developed for text-to-speech synthesis. Both labeling methods used standard Entropic waves+ tools developed for ToBI transcription. A log file was created during the labeling of each file that identified labeler, filename, method, and beginning and ending times of each session. Results indicate that labeling from defaults was faster than standard labeling, and that defaults had relatively little impact on label assignment.

2pSC7. Evaluation of quality of speech enhanced by HMM and AR model-based systems. Anisa Yasmin (Dept. of Elec. and Computer Eng., Univ. of Waterloo, Waterloo, ON N2L 3G1, Canada, ayasmin@crg3.uwaterloo.ca), Paul Fieguth, and Li Deng (Univ. of Waterloo, Waterloo, ON N2L 3G1, Canada)

Speech enhancement algorithms have demonstrated their application potential in a wide variety of speech communication contexts in which the quality or the intelligibility of speech has been degraded by the presence of background noise: hearing aids, cellular phones, public telephones, hands-free telephones, air-ground communications, etc. By far the two most popular choices for model-based speech enhancement are the Wiener-filter-based hidden Markov model (HMM), and the autoregressive (AR) model-based Kalman filter. Although researchers have been studying such enhancement systems for some time, relatively little has been undertaken in comparing the quality of enhanced speech produced by these two systems. This paper studies the Wiener/HMM and AR/KF models and conducts a comprehensive comparative study of the relative quality of enhanced speech, based on utterances from the TIMIT database contaminated by simulated and sampled empirical noises. The Wiener/HMM and AR/KF comparison includes both subjective and qualitative evaluations: subjective assessments are based on mean opinion scores (MOS) and the inspection of temporal and spectrogram plots; objective evaluations are based on average and segmental signal-to-noise ratios. HMM enhanced speech has most of the noise removed, but with interruptions and discontinuities present due to the switched nature of the HMM. AR model-based enhancement possesses more audible background noise in the high-frequency region above 4 kHz, however the speech is smoother, with fewer discontinuities.

2pSC8. Continuum mechanical model of the tongue and mouth floor. Reiner Wilhelms-Tricarico (Institut de Commun. Parlee, INPG, Grenoble, France)

In this new model, the mouth floor and the tongue body are treated as blocks represented by tri-quadratic finite elements in which the muscles are modeled as continuous directed fields of the fiber direction. Collision with the hard palate and other rigid structures is modeled by subdividing the surface into small triangles for rapid collision detection and through calculating forces by a penalty method to enforce impenetrability of the rigid structures. A similar method is provided for the more complicated case of the collision between tongue body and soft floor of the mouth. The model can be coupled with a model of the jaw and hyoid where muscles may be modeled as contracting strings. The moving hyoid and jaw provide kinematic constraints for the tongue/mouth-floor model. Active muscle stress is generated by a Hill model of muscle shortening combined with a rational extension for muscle elongation, and a polynomial curve represents the relation between muscle length and stress. The stiffness matrix and damping matrix of the system (used for the implicit time stepping method solving the equations of motion) are derived directly from the continuum mechanical muscle model. An overview of the theory and numerical examples will be presented.

2pSC9. Measurement error compensation using integration for hidden Markov-model-based speaker recognition. Marie A. Roch (Dept. of Computer Sci., The Univ. of Iowa, Iowa City, IA 52242) and Richard R. Hurtig (The Univ. of Iowa, Iowa City, IA 52242)

Like all such measurements, the source data for speaker recognition is subject to errors in measurement due to transducer, channel, and quantization effects. When training and test equipment are known, one may calibrate the hardware and introduce a noise compensation procedure into the recognition process. In many applications, it is highly desirable for speaker recognition tasks to function with a wide variety of unknown equipment, making calibration impractical. A study is presented with the *a priori* assumption that for each feature vector $\vec{\tilde{o}}$ observed with measurement noise, an error compensated vector $\vec{\hat{o}}$ lies within some uniformly distributed interval $\pm\epsilon$ of the observed vector. A statistic derived from the observation set is computed and used to estimate an empirical interval in the neighborhood of each observation. An approximation to integration over the interval is carried out and is used in place of the density measurement at $\vec{o}$. Tests using a 15-component cepstral feature vector derived from telephone quality speech (King corpus, San Diego speakers, sessions 1–5) have shown reductions of error rate on the order of 15% as compared to a baseline system. Techniques to reduce the algorithmic cost of the integration will also be discussed.

2pSC10. Detecting filled pauses in spontaneous speech. Douglas O'Shaughnessy (INRS-Telecommunications, 16 Pl. du Commerce, Nuns Island, QC H3E IH6, Canada)

Practical speech recognizers must accept normal conversational voice input (including hesitations). However, most automatic speech recognition work has involved read speech, whose acoustic aspects differ significantly from speech found in actual dialogues. Hesitations, filled pauses, and restarts (after aborted utterances) are common in natural speech, yet few recognition systems handle such disfluencies with any degree of success.

Among other problems, filled pauses (e.g., ''uhh,'' ''umm''), unlike silences, resemble phones as part of words in continuous speech. The work reported here further develops techniques to allow identification of filled pauses. A distinction is made between disfluencies in actual dialogs (e.g., in the Switchboard database of natural telephone conversations, which have poor recognition rates so far) and simulated ones (e.g., the ATIS Wizard-of-Oz-style database of airline travel inquiries). It appears that speaking with actual people influences disfluencies, e.g., filled pauses tend to be shorter and more variable in pitch patterns, although unfilled pauses adjacent to filled ones remain important in both styles. While most automatic recognition methods rely entirely on spectral envelope (e.g., low-order cepstral coefficients), identifying hesitation phenomena seems to require use of fundamental frequency and duration in addition to such spectral parameters.

2pSC11. DSP circuit to improve speech intelligibility. Michael J. Metz (Audiology Assoc., P. C., 14150 Culver Dr., Ste. 207, Irvine, CA 92604) and Neil A. Shaw (MenloScientific Acoustics, Inc., Topanga, CA 90290-1610)

Two studies, one equipment based and one clinically based, were attempted to verify the contention that a DSP circuit is capable of changing the intelligibility of speech in noise. The first used two intelligibility metrics (RASTI and %ALcons) to measure the change in metric score under the conditions of test signal with (1) no DSP speech signal processing of the test signal, and (2) with DSP speech processing of the test signal in the presence of pink noise in the background. The second study measured the articulation functions of normal ears using phonetically balanced word lists under conditions of (1) no speech signal processing, and (2) ''maximum'' DSP processing when each condition was varied in the presence of a set of level of pink noise. The algorithm used in this study is the ''Voice Intelligibility Processor,'' or VIP signal processor. The studies indicate an increase in the intelligibility metric scores and the measured articulation functions. For the articulation studies, when the processor is ''normalized'' to account for the perceived increase in loudness level produced by the processor algorithm, the mean shift in ''50% point'' of the articulation function was approximately 6 dB. For the RASTI study, similar improvement was also found.

2pSC12. The use of spectral versus temporal cues to recognize speech. Arindam Mandal, Laura J. Davis, Carol Espy-Wilson, and Melanie Matthies (Boston Univ., Boston, MA 02215)

Previous work has shown that listeners are able to use temporal information to identify spectrally limited consonants [Shannon *et al.*, Science 303–304 (1995)]. In this study, we used spectrally impoverished speech to: (1) investigate the performance of a temporal-based automatic classifier; (2) describe the performance of human listeners; and (3) compare the performance of the classifier with human listeners. For the automatic classifier, acoustic events are extracted from a cross-spectral temporal measure that computed onsets and offsets of acoustic energy. The strength and time difference between these acoustic events served as input to the classifier. The results from human listeners showed that their ability to identify nasality and voicing remained fairly good. In addition, recognition of manner for stops and fricatives remained high. As the speech signal was degraded, listeners often confused sonorants with fricatives. The temporal-based classifier did not make such confusions. The classifier is able to identify manner, voicing and nasality well and the results remain fairly consistent across different levels of degradation. Thus, these results show the importance of temporal information and they suggest that human listeners may not optimally combine all cues to identify spectrally impoverished consonants. [Work supported by ONR Supervised MURI grant Z883402.]

2pSC13. Comparative study of $F0$ extractors for high-quality speech synthesis. Hideki Kawahara (Faculty of Systems Eng., Wakayama Univ./ATR/CREST, 930 Sakaedani, Wakayama, 640-8510 Japan) and Parham S. Zolfaghari (ATR Human Information Processing Res. Labs., Kyoto, Japan)

Performance of a new $F0$ extraction algorithm based on fixed point analysis of filter center frequency to output instantaneous frequency [Kawahara *et al.*, Eurospeech'99] was compared with numbers of $F0$ extraction algorithms based on different definitions of fundamental frequency. The proposed method uses partial derivatives of the mapping at fixed points to estimate carrier to noise ratio of $F0$ information. It also enables integration of distributed $F0$ cues among harmonic components to provide a reliable $F0$ estimate. Objective evaluations were conducted using simulations and a speech database with simultaneous EGG (electroglottograph) recording. Subjective evaluations were based on reproduced speech quality assessment by a high-quality speech analysis/modification/synthesis method STRAIGHT [Kawahara *et al.*, Speech Commun. **27**, 187–207]. Discussions about the relevance of various $F0$ definitions for high-quality speech synthesis will be presented based on these test results. It was also indicated that the proposed method is tunable to specific needs.

2pSC14. A vowel synthesizer based on $F0$-modulated formant sinusoids. Ingo Hertrich and Hermann Ackermann (Dept. of Neurology, Univ. of Tuebingen, Hoppe-Seyler Str. 3, D-72076 Tuebingen, Germany)

A vowel synthesis algorithm was developed resembling parallel formant synthesizers. However, formants are computed as amplitude- and phase-modulated sinusoids instead of being represented by recursive filters or resonators. The fundamental frequency is imposed on the signal in the following way: Each pitch period starts with a short initial rise of a duration of pi/2 of the first formant frequency; at the end of this ramp phase all formants start as sinusoids at pre-defined phase angles and amplitudes, and successively undergo linear attenuation toward zero amplitude at the end of each pitch period. Adjacent pitch periods, thus, do not influence each other. In principle, the signals produced by this algorithm resemble the output of parallel formant synthesizers [e.g., D. H. Klatt, J. Acoust. Soc. Am. **67**, 971–995 (1980)]. However, by incremental sample-by-sample computation of the formants' phase angles, formant frequencies are allowed to continuously change within single pitch periods. This algorithm produces vowels as well as formant transitions signaling stop consonant–vowel syllables such as /ba/, exhibiting a homogeneous, stereotypic voice quality. Because of its additive procedure, this method might allow for the implementation of a variety of further acoustic aspects of the human speaking voice in a well-controlled manner.

2pSC15. Problems of signal processing in modern wireless terminals and its impact on speech quality. Ekkehard Diedrich and Hans Wilhelm Gierlich (Goslarer Ufer 35, D-10589 Berlin, Germany)

Modern terminal applications in speech communications are more and more including signal-processing technologies, such as switching devices, echo cancellers, nonlinear processors, level control, and compander a.o. Apart from the benefits of such technologies, there are doubtless impacts on the speech transmission quality like echo, level fluctuations, and clipping. Even modern terminal developments, e.g., for mobile communications lead to shorter and compact handset designs with some negative influences on the speech transmission quality from an acoustical point of view. One example is echo because of the stronger internal coupling between microphone and loudspeaker in very short handsets and the need for echo control to avoid the negative impact on speech quality. New test and measurement procedures (subjective and objective) are described and discussed to give the possibility of classifying these impacts on speech transmission quality.

2pSC16. Improvements to segmentation accuracy using a large speech database. Alistair Conkie (AT&T Labs.–Res., 180 Park Ave., Florham Park, NJ 07932, adc@research.att.com)

Current text-to-speech (TTS) systems rely increasingly on large speech databases for both the raw speech necessary for concatenative synthesis and for material used for modeling prosody. It is time-consuming to label the databases phonetically so that they can be useful. It is nevertheless important for the quality of the resulting synthesis that the labeling be accurate. One currently used method involves some kind of automatic labeling – usually using hidden Markov modeling (HMM) techniques, followed by manual correction. This is still time-consuming. Recently, work has been done on post-processing the output from an HMM recognizer with a fuzzy logic system [Torre Toledano *et al.*, Third ESCA/COCOSDA Workshop on Speech Synthesis, Australia, November 1998] to improve accuracy. We examine this promising idea by applying post-processing to automatically generated segmentation of the speech signals in our TTS database. The effectiveness of the technique is evaluated both in terms of labeling accuracy in comparison to hand-labeled data, and with informal listening tests for synthesis quality.

2pSC17. Flexible post-lexical processing for speech synthesis from a large unit selection database. Mark Beutnagel (AT&T Labs.–Res., Shannon Lab., 180 Park Ave., Florham Park, NJ 07932, mcb@research.att.com)

Online unit selection from large speech databases provides an opportunity to essentially play back words, phrases, and even sentences which were included in a recorded corpus. This capability can be extremely useful for limited domains, e.g., application prompts. Without switching voices, such a synthesizer could integrate high-quality synthesis with near-perfect recorded material. However, traditional post-lexical processing (PLP) considers only the phoneme specifications and not the sequences which actually exist in the target database. Phonemes supplied by the dictionary are typically rewritten into a single sequence of phones with reduced vowels, flapped t's, etc. Given the enormous variability of human speech, any single sequence is unlikely to match an entire phrase or prompt as spoken and labeled. This paper addresses the use of flexible PLP, allowing multiple transcription possibilities which are essentially equivalent, at least for the speaker in question. By building the equivalences from the specific dictionary used by the synthesizer and the detailed phonetic labeling of a specific voice database, longer regions of the database can be selected, reducing the number of concatenation points in ordinary synthesis and increasing the odds of selecting complete recorded phrases.

2pSC18. Distributed acoustic model of the entire respiratory tract. Paul Harper (1285 Elec. Eng. Bldg., Purdue Univ., West Lafayette, IN 47907, pharper@ecn.purdue.edu), Hans Pasterkamp (Univ. of Manitoba, Winnipeg, Canada), Steve S. Kraman (VA Medical Ctr., Lexington, KY), and George R. Wodicka (Purdue Univ., West Lafayette, IN 47907)

The anaylsis of breathing sounds measured over the extrathoracic trachea has been used to detect and monitor respiratory tract changes such as those found in asthma and obstructive sleep apnea. To begin to link the attributes of these easily measured sounds to the underlying anatomy, a lumped-element model of the acoustic properties of the entire respiratory tract (supraglottal plus subglottal airways) with varying cross-sectional area, yielding walls, and dichotomous branching in the subglottal component was developed over the frequency range from 100 to 3000 Hz. The portions of the model above and below the larynx (supra- and subglottal) predict well the distinct locations of the spectral peaks from speech sounds such as /ah/ as measured at the mouth and the trachea, respectively, in healthy subjects. When combining the supra- and subglottal portions in a complete tract model, the predicted peak locations compare favorably with those of tracheal sounds measured during normal breathing in the same subjects. In addition, the model predicts both the direction and magnitude of the relatively small spectral shifts that occur in tracheal sounds during a neck flexion/extension maneuver, indicating that measurable acoustic parameters are sensitively tied to respiratory tract dimensions.

2pSC19. Effect of a tracheal tumor on breathing sounds: A case study. Steve S. Kraman (VA Medical Ctr., 2250 Leestown Rd., Lexington, KY 40511, sskram01@pop.uky.edu), Paul Harper (Purdue Univ., West Lafayette, IN), Hans Pasterkamp (Univ. of Manitoba, Winnipeg, Canada), and George R. Wodicka (Purdue Univ., West Lafayette, IN)

A 67-year-old man came to the hospital with a history of shortness of breath on exertion and cough progressing to severe breathlessness at rest over several weeks. Physical exam revealed stridorous sounds in inspiration and expiration heard at the mouth and over the trachea. Lung function tests revealed severe airway obstruction in expiration and probably inspiration. Fiberoptic airway inspection revealed a large mass above and arising from the main carina (first airway bifurcation). This mass obstructed the distal trachea except for a narrow ($\sim$2 mm) circumferential space between the mass and the tracheal wall. Breathing sounds were recorded from in front of the mouth and the anterior neck overlying the trachea both before and after laser ablation of the mass. Before ablation, the tracheal sound exhibited unusual resonances and antiresonances that rose in frequency by nearly 100% during inspiration and fell during expiration. These variable sounds disappeared after laser ablation of the mass. It is postulated that the mass behaved as a dynamic acoustic obstruction and thereby altered the distal boundary condition and resonance behavior of the respiratory tract during breathing. This provides insights into the links between the airway anatomy and measurable acoustic properties of breath sounds.

2pSC20. Speaker verification performance comparison based on traditional and electromagnetic sensor pitch extraction. T. J. Gable, L. C. Ng, G. C. Burnett, and J. F. Holzrichter (Lawrence Livermore Natl. Lab., P.O. Box 808, Livermore, CA 94551)

This work compares the speaker verification performance between a traditional acoustic-only pitch extraction to a new electromagnetic (EM) sensor based pitch approach system. The pitch estimation approach was developed at the Lawrence Livermore National Laboratory (LLNL) utilizing Glottal Electromagnetic Micropower Sensors (GEMS, also see http://speech.llnl.gov/). This work expands previous pitch detection work by Burnett *et al.* [IEEE Trans. Speech and Audio Processing (to be published)] to the specific application of speaker verification using dynamic time warping. Clearly, a distinct advantage of GEMS is its insensitivity to acoustic ambient noise. This work demonstrates the clear advantage of the GEMS pitch extraction to improve speaker verification error rates. Cases with added white noise and other speech noise were also examined to show the strengths of the GEMS sensor in these conditions. The EM sensor speaker verification process operated without change over signal-to-noise (SNR) conditions ranging from -20 to -2.5 dB; the acoustic algorithms became unusable at SNR exceeding -10 dB. [Work supported by NSF and DOE.]

2pSC21. The use of glottal electromagnetic micropower sensors (GEMS) in determining a voiced excitation function. Gregory C. Burnett, John F. Holzrichter, Larry C. Ng, and Todd J. Gable (Lawrence Livermore Natl. Lab., P.O. Box 808, L-271, Livermore, CA 94551)

Recent experiments using a portable, extremely low-power electromagnetic motion sensor to detect the motion of the posterior tracheal wall during speech production will be presented. The motion of the wall may be related to the driving subglottal pressure through a lumped element circuit model, leading to an approximation to the voiced excitation function of the human vocal tract. Using the excitation and the recorded spoken audio, a stable and accurate transfer function of the vocal tract may be calculated every few glottal cycles in near real-time. The excitation function may be used to calculate very accurate pitch information at low cost, and the transfer functions may be employed as an additional feature vector to enhance the performance of a new class of speech recognizers and synthesizers. [Work supported by NSF and DOE.]

2pSC22. Background speaker noise removal using combined EM sensor/acoustic signals. Lawrence C. Ng, John F. Holzrichter, Gregory C. Burnett, and Todd J. Gable (Lawrence Livermore Natl. Lab., P.O. Box 808, L-054, Livermore, CA 94551)

Recently, very low-power EM radarlike sensors have been used to measure the macro- and micro-motions of human speech articulators as human speech is produced [see Holzrichter *et al.*, J. Acoust. Soc. Am. **103**, 622 (1998)]. These sensors can measure tracheal wall motions, associated with the air pressure build up and fall as the vocal folds open and close, leading to a voiced speech excitation function. In addition, they provide generalized motion measurements of vocal tract articulator gestures that lead to speech formation. For example, tongue, jaw, lips, velum, and pharynx motions have been measured as speech is produced. Since the EM sensor information is independent of acoustic air pressure waves, it is independent of the state of the acoustic background noise spectrum surrounding the speaker. By correlating the two streams of information together, from a microphone and (one or more) EM sensor signals, to characterize a speaker's speech signal, much of the background speaker noise can be eliminated in real time. This paper presents several algorithms to demonstrate the added noise suppression capability of the glottal EM sensors (GEMS). [Work supported by NSF and DOE.]

TUESDAY AFTERNOON, 2 NOVEMBER 1999 GRANT–HARDING ROOM, 3:15 TO 5:15 P.M.

Session 2pSP

Signal Processing in Acoustics: Time-Frequency Applications in Acoustics II

Patrick J. Loughlin, Chair

Department of Electrical Engineering, University of Pittsburgh, 348 Benedum Hall, Pittsburgh, Pennsylvania 15261

Chair's Introduction—3:15

Invited Papers

3:20

2pSP1. Reduced interference distribution application and interpretation for transient acoustic events. William J. Williams (EECS Dept., Univ. of Michigan, Ann Arbor, MI 48109, wjw@eecs.umich.edu)

Transient acoustic events often present difficulties when using conventional time-series or spectral methods. The Reduced Interference Distribution (RID) and its derivatives have been under development [W. J Williams and J. Jeong, *Time-Frequency Signal Analysis*, edited by B. Boashash (Longman Cheshire, New York, 1992)] for several years. The RID was specifically motivated by the need for a better analysis tool for transient acoustic events [ONR Grants N00014-90-J-1654, N00014-97-1-0072]. The deficiencies of the spectrogram for marine mammal sound analysis were early recognized [William Watkins, Marine Bioacoustics **2**, 15–43 (1966)]. Watkins' illustrations of spectrogram deficiencies and updated results obtained using the Wigner distribution (WD) and the RID will be presented. The RID represents an attempt to retain the desirable mathematical properties of the WD while reducing the often troublesome WD interference terms. Examples drawn from several acoustic transients such as marine mammal sounds, joint clicks and vortex formations in turbulent air flow will be used to illustrate the usefulness of the RID approach. The importance of retaining such properties as proper time and frequency marginals; time and frequency support along with proper group delay and instantaneous frequency will be discussed. The value of covariance with time and frequency shifts and scale changes will also be emphasized.

3:40

2pSP2. Characterization of densities by moments. Leon Cohen (Dept. of Phys., City Univ., New York, NY 10021)

For two-dimensional classical densities it is very often the case that the general properties of a density can be characterized by low-order joint moments and conditional moments. The advantage of this is that moments and conditional moments are constants and one-dimensional functions, respectively, and hence can be more effectively used for characterization, classification, and detection than the full density. In the time-frequency case unique problems arise. First, the definition of conditional moments is problematic and has not been fully investigated. Second, since a time-frequency density may go negative the methods of probability theory may not be taken over directly. We investigate these issues and show that time-frequency conditional moments can be defined in such a way that they appropriately characterize the general features of a time-frequency density. [Work supported by ONR.]

4:00

2pSP3. Further applications of time-frequency methods in acoustics. G. C. Gaunaurd (Naval Surface Warfare Ctr., Carderock Div., Code 683, W. Bethesda, MD 20817-5700) and H. C. Strifors (Defense Res. Establishment, FOA, Stockholm, S-17290, Sweden)

Time-frequency (t-f) methods have been repeatedly established as effective tools to successfully analyze a variety of signatures from many types of sources or scatterers. Many such signatures have been acoustic in nature, and we have studied these most extensively in the past. We have presented several of these examples before [viz., G. C. Gaunaurd and H. C. Strifors, J. Acoust. Soc. Am. **104**, 1746, 136th ASA Mtg. (1998)]. Here, we will cover acoustical applications that we have analyzed, but which were not adequately covered in 1998. We will also outline additional ones—more extensively discussed in Appl. Mech. Rev. **50**, 131–149 (1997)—and a recent radar application [i.e., *ibid.* IEEE Trans. Antennas Propag. **46**, 1252–1262 (1998)]. In this case, the present t-

f processing was shown capable of successfully identifying an air target that had been covered with a dielectric RCS-reducing layer. We believe that this general approach, jointly with the way we have used it to physically interpret broadband sonar/radar signatures, can lead to straightforward implementations for many practical target-classification problems associated with many types of sensors. [Work partially supported by the ILIR Program of the authors' Institutions and the ONR.]

4:20–4:30 Break

Contributed Papers

4:30

2pSP4. Time-frequency analysis of the backscattering by tilted finite cylindrical shells. Scot F. Morse[a] and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814, morse@lpsa2.nrl.navy.mil)

Previous experiments and analysis have shown that the backscattering from thick-walled finite cylindrical shells can be significantly enhanced in and above the coincidence frequency region over a large range of angles of the incident sound [S. F. Morse *et al.*, J. Acoust. Soc. Am. **103**, 785–794 (1998)]. In this presentation, impulse backscattering measurements performed with a wide bandwidth PVDF sheet source are analyzed in terms of a simple joint time-frequency representation. When plotted for a sequence of specific tilt angles, it is possible to identify and then observe the progression of individual ray contributions as the coupling conditions to the shell are changed. In the frequency range investigated, where $ka < 45$ and $kh < 3.4$, several of the surface waves excited are highly dispersive and as a result are launched on the shell over a wide range of frequencies and tilt angles, including end-on incidence. The analysis described also confirms various features of the responsible scattering mechanisms predicted by ray theory. The results show the importance of time-frequency analysis to the identification of back scattering in a situation where no exact reference solution is available. [Work supported by the Office of Naval Research.] [a] Currently with Naval Research Laboratory, Code 7136, 4555 Overlook Ave. SW, Washington, DC 20375.

4:45

2pSP5. A new method for an almost-on-the-fly estimation of the instantaneous frequency and of the amplitude of a signal. Gerard Girolami and Lucile Rossi (URA 20253 CNRS, SDEM, Equipe Ondes et Acoustique, Universite de Corse, Faculte des Sci., BP 52, 20250 Corte, France, girolami@univ-corse.fr)

The instantaneous frequency (IF) of a signal is a parameter whose estimation is of prime importance in a number of domains ranging from musical acoustics to seismics and astronomy. Its determination is usually based on one of the following three general methods: the derivation of the phase of the associated analytic signal, the Wigner–Ville transform and others Cohen's class transforms, or the three-point algorithms like the Teager–Kaiser ''energy operator.'' A new three-point method is described which is simple, fast, precise, robust to noise, and estimates the IF and the amplitude with a very small delay. This method gives excellent results on almost any kind of chirps (polynomials, hyperbolics, exponentials, etc.) and frequency modulations (sinusoidal, FSK, Gaussian pulse, etc.), whether or not associated with amplitude modulation (sinusoidal, Gaussian, etc.). The relative errors on the amplitude and on the IF are usually around 1% in the no-noise case, and increase to about 5% for a 20-dB signal-to-noise ratio, even when these different modulations are associated.

5:00

2pSP6. Detection and identification for highlight echo using the time-frequency postaccumulation method. Zhu-Ye and Gong-suying (Inst. of Acoust., Academic Sinica, P.O. Box 2712, Beijing 100080, PROC)

The highlight distribution of underwater target in active sonar is the main characteristic for target identification. The pulse compression with chirp signal is a current method for target echo detection. But sonar highlight echo is always time-frequency fading and the background reverberation is nonstationary, so that the matched filter is not optimum to detect the highlight echo. Based on that a linear-frequency modulation signal, the peak trace of the ambiguity function (AMF) or the Wigner–Ville distribution (WVD), is linearity, the accumulations along the peak trace (APT) (analog to Radon transform in image processing) of both the WVD and cross-AMF of multi-target echo are developed in this paper. The method is applied to a simulated multi-target echo and practical highlight echoes of the target. The detection characteristics and performances are comparison for both accumulations with matched filter. The results of the analysis and simulation are shown: (1) In detection performances, the APT–WVD and APT–CAMF are superior to the matched filter if the accumulating time is selected properly and especially if the background spectra is not even. (2) In multi-target resolution, the APT–WVD is superior to the APT–CAMF. (3) The APT–WVD need not know the reference signal which can be estimated by this method.

TUESDAY AFTERNOON, 2 NOVEMBER 1999 FAIRFIELD ROOM, 1:00 TO 5:20 P.M.

Session 2pUW

Underwater Acoustics and Engineering Acoustics: Robert J. Urick Memorial Session

David L. Bradley, Chair

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

Chair's Introduction—1:00

Invited Papers

1:05

2pUW1. Source level and Urick. James E. Barger (BBN Technologies, 70 Fawcett St., Cambridge, MA 02138)

Source level (SL) is the first term in Bob Urick's sonar equation, and it represents the root-mean-square pressure of sound radiated at the target, in decibels. The fundamental measure of source level is the signal-energy level (SEL) represented by the time integral of sound pressure squared, to which the detection index is proportional, in decibels. Despite using a form of sonar equation more

natural for long-pulse transducers than for impulsive ones, he only described explosives in detail in his chapter on generation of underwater sound. When ''Principles of Underwater Sound for Engineers'' was published, there was a tendency to use impulsive sound sources to study sound interactions with the sea and to use narrow-band long-pulse transducers in sonar systems. The diminution of this tendency has been a very important development since Bob's book was written, and it has occurred because of improvements in both kinds of transduction. Transducer bandwidth has been increased, both by materials with larger electromechanical coupling coefficients and by innovative designs. These include ''hybrid'' transducers that combine magnetostrictive and electrostrictive actuators and projector arrays designed to have no mechanical reactance. Improved impulsive sound sources such as air guns, sparkers, and explosive arrays now find sonar use.

1:35

2pUW2. Transmission loss since Urick. Charles W. Spofford (Science Applications Intl. Corp., 1710 Goodridge Dr., McLean, VA 22102)

Advances in understanding propagation of ''underwater sound'' are traced from the view expressed in Urick's 1967 landmark text to our current state through two remarkable transitions. First, propagation in 1967 was an arcane phenomenon with annoying, questionably modelable ''anomalies'' from spherical spreading which forced extensive global measurements. It is now known to be a stable, well-understood property of the ocean whose predictability is limited only by our knowledge of the controlling environmental inputs. In fact, models are now being used to deduce that environment through inversions. Second, the ''TL'' term in the sonar equation has expanded to include coherence measures inextricably coupled to the ''gain'' terms. These are being routinely modeled, again limited mainly by our knowledge of the environmental inputs. The variety of approaches to modeling propagation hinted at in Urick has blossomed to form a powerful set of tools for designing and analyzing a wide range of sonars. Geo-political ASW concerns gave birth to ''underwater sound'' for engineering sonar systems of ever increasing complexity and challenged us in 1967 to understand the physics of deep-water propagation. In 1999 our challenge is to extrapolate that physics to the world's enormously variable shallow-water areas.

2:05

2pUW3. Target strength. Louis Dragonette (Naval Res. Lab., Code 7132, Washington, DC 20375-5350)

Chapter 9 of Urick's *Principles of Underwater Sound* is titled ''Reflection and Scattering by Sonar Targets: Target Strength'' [R. Urick, *Principles of Underwater Sound* (McGraw–Hill, New York, 1975), p. 263], and while this chapter gives an excellent review of target strength definitions, simple echo formation processes, and echo characteristics, it is the table, ''Target Strength of Simple Forms,'' compiled by Urick from several obscure sources, that is invaluable to those who deal with target strength issues on a routine basis. A typical first approximation to the target strength (TS) of a body of interest is made by approximating the scatterer as a combination of the simple shapes listed in the table, and then summing up the separate rigid-body responses to produce a first-order target strength prediction. Some common-sense principles that derive from such an exercise are discussed as well as comparisons between such first-order predictions and the results and understanding obtained from modern theoretical and experimental approaches. [Work supported by ONR.]

2:35

2pUW4. Bob Urick and ambient sea noise. Raymond C. Cavanagh (Science Applications Intl. Corp., 1710 Goodridge Dr., McLean, VA 22120)

Selected highlights of progress in the measurement and understanding of ambient sea noise are presented, from the time of publication of the first edition of Urick's pioneering textbook (1967) through his noise monograph (1984) and beyond. Urick's presentation of the fundamentals became the primer for a generation of ocean acousticians and the basis for many of today's theories and predictive capabilities. Valuable insights were derived from his direct involvement in the measurement and interpretation of noise data, dating to the early 1950's. His treatments of arctic and shallow-water noise anticipated research initiatives of the 1980's and 1990's, as did his framework for noise fluctuations. Some notable advances since Urick's publications are discussed, including: mechanisms for wind-wave noise sources, predictive models for ship-generated noise properties, the noise ''floor'' and noise increases over time, directionality, and the effects of coastal sources. His attention to the details of metrics and units for ocean acoustic quantities is acknowledged as a special contribution to the field.

3:05–3:20 Break

3:20

2pUW5. Properties of transducer arrays: Directivity index (Urick, 1967). Arthur B. Baggeroer (MIT, Cambridge, MA 02139, abb@arctic.mit.edu)

The chapter, ''Properties of Transducer Arrays: Directivity Index,'' in Urick's text includes not only the directivity index (DI) of an array, but also many of the important metrics of spatial performance which remain important in assessing the performance of a sonar system. Beampatterns, tapers and sidelobes, signal gain, noise gain and even superdirectivity are all included. The DI metric in the sonar equation indicates gain against 3-D isotropic noise; alternatively, it is used to convert a beam noise level to power per steradian. While easy to compute and appropriate when a field is diffuse, the DI is not a good metric for spatially discrete noise such as shipping which can often dominate a noise field. Recently, the signal and noise gain metrics are used more often. Signal gain is an

issue as sonar systems use very large apertures; noise gain is also an issue when the directionality of the noise fields is exploited. Certainly, the most significant evolution in spatial processing not included in Urick's chapter was the use of adaptive processing which has been enabled by the availability of high-speed DSP for real time processing. The presentation will review Urick's important contributions to describing the performance of arrays and how they have impacted sonars.

3:50

2pUW6. Detection threshold. Henry Cox (ORINCON Corp., 4350 No. Fairfax Dr., Ste. 470, Arlington, VA 22203)

Detection threshold or recognition differential is defined as the signal-to-noise ratio at the beamformer output required for detection. An introduction to this topic is provided in Uricks' *Principles of Underwater Sound*. The subject is frequently a source of confusion due to the multiple possible definitions of signal-to-noise ratio, the effects of nonlinearities in the signal processing, and the overall complexity of the sonar operators job of signal recognition in real-world clutter. In this paper, a review of the basic theory of mathematical relationships are provided, followed by a discussion of practical limitations in applying simple formulas to performance predictions. The relationships of detection threshold to the signal-processing approach and background-noise statistics are discussed. Active sonars in both noise and reverberation, as well as passive sonars involving energy detection, spectral analysis, and cross correlators are discussed.

4:20

2pUW7. Reverberation: Urick and beyond. D. V. Holliday (Marconi, 4669 Murphy Canyon Rd., San Diego, CA 92123-4333)

In his classic textbook, *Principles of Underwater Sound for Engineers*, Urick dedicated more space to reverberation, than to any other subject except propagation loss. His characterizations of reverberation were practical, easily understood, and remarkably complete for the era in which they were written. The chapter on scattering, especially the references, was an invaluable starting point for both students and practicing designers of active sonars. Urick, the sonar designer, pragmatically asserted that reverberation was a ''particularly obnoxious'' phenomenon. However, Urick, the scientist, demonstrated an unbridled curiosity about the underlying causes of the phenomena that he observed in the sea. Near the end of his active career in acoustics, the patterns of research into sound scattering changed dramatically. Urick personally encouraged young scientists to pursue a detailed understanding of the causes of scattering. Reverberation, which clearly had been a part of the ''noise'' in the sea became a ''signal'' which could be used to study and describe the sea surface, the seabed, and the various particles, animals, and bubbles found in the water column. This paradigm shift and some of its consequences, e.g., the extension of measurements to both higher and lower frequencies will be examined. [Work supported by ONR Code 322BC.]

Contributed Papers

4:50

2pUW8. Perspectives on teaching: A tribute to Bob Urick. Paul C. Etter (Northrop Grumman Corp., Oceanic and Naval Systems, P.O. Box 1488, M.S. 9115, Annapolis, MD 21404, paul_c_etter@md.northgrum.com)

Based on 5 years of teaching short courses with Bob Urick, an assessment is made of his teaching style and elements of content. Urick's influence on the underwater-acoustics community is exemplified by the popularity of his books and short courses. The fact that much of Urick's nomenclature became institutionalized within the international community provides further evidence of his authoritative command of the technical literature and his lucid conveyance of fundamental concepts. Specific examples are drawn from personal experience to demonstrate the effective aspects of his instructional methods, including both style and content.A synthesis of Urick's methods can instruct and inform contemporary educators in the field of underwater acoustics and related disciplines.His equally important, but less tangible, impact on the professional development of emerging sonar engineers is also discussed from the dual perspective of a co-instructor and former student. Finally, Urick's conception of the complementary roles of field measurements and numerical modeling is reviewed to characterize the proper relationship between measurements and modeling in modern ocean-acoustics research.

5:05

2pUW9. Array signal-gain measurements in shallow-water. William M. Carey (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215) and Peter G. Cable (BBN Technologies, New London, CT 06320)

Shallow-water transverse coherence lengths can be estimated by the measurement of the narrow-band coherence function, broadband correlation function, or the signal gain either from a direct measurement with a filled aperture or the steered beam response of a sparsely filled aperture. Signal-gain measurements with an array of sensors have a larger number of degrees of freedom, spatial filtering of unwanted noise sources, and higher signal-to-noise ratios. Given a Gaussian coherence function, the coherence length may be estimated from these gain measurements. This paper discusses signal-gain measurements performed in five shallow-water sandy-bottomed areas with known environmental conditions using horizontal-bottomed arrays, omnidirectional explosives, and continuous sources between 100 and 600 Hz to ranges of 40 km. The coherence lengths determined by using a Gaussian coherence function yield a consistent representative value of approximately 30 wavelengths at a range of 40 km and a frequency of 400 Hz, which decreases with an increase in frequency. Comparisons show these lengths are consistent with previous measurements. Analytical considerations indicate these lengths are due to the combined effects of the water column variability and bottom. Implications for shallow-water array processing are discussed.