PROGRAM OF

The 145th Meeting of the Acoustical Society of America

Nashville, Tennessee • Nashville Convention Center • 28 April–2 May 2003

NOTE: All Journal articles and Letters to the Editor are peer reviewed before publication. Program abstracts, however, are not reviewed before publication, since we are prohibited by time and schedule.

MONDAY MORNING, 28 APRIL 2003 ROOMS 103/104, 8:00 A.M. TO 12:00 NOON

Session 1aUW

Underwater Acoustics, Acoustical Oceanography and Physical Acoustics: Frederick Tappert Memorial Session on Propagation Phenomena and the Parabolic Equation

David R. Palmer, Chair
Atlantic Oceanographic and Meteorological Laboratory, 4301 Rickenbacker Causeway, Miami, Florida 33149

Chair's Introduction—8:00

Invited Papers

8:05

1aUW1. The summer of 1974. Stanley M. Flatté (Phys. Dept., Univ. of California at Santa Cruz, Santa Cruz, CA 95064, flatte@physics.ucsc.edu)

In the early 1970s, Garrett and Munk summarized oceanographers' knowledge of ocean internal waves in a reference statistical spectrum. Almost at the same time, Hardin and Tappert introduced the parabolic equation into ocean acoustics. In the summer of 1974, Flatté and Tappert [J. Acoust. Soc. Am. **58**, 1151–1159 (1975)] helped bring these two ideas together by simulating 100-Hz acoustic propagation through an approximate version of the Garrett–Munk spectrum to a range of 100 km, in order to help understand the effects caused by internal waves on acoustic fluctuations. The history of that summer will be discussed, along with a movie made at that time. The influence of that work on subsequent developments will be discussed.

8:30

1aUW2. Waves, rays, and the predictability of underwater sound fields. Michael G. Brown (RSMAS-AMP, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149)

Recent results relating to the application of ray-based methods to long-range underwater sound propagation are discussed, including Fred Tappert's influence on the development of some of the ideas. Understanding limitations on predictability linked to chaotic motion of ray trajectories is a central issue. Related issues that are discussed include semiclassical breakdown and the important role played by the background sound-speed structure in controlling ray and wavefield stability. [Work supported by ONR.]

8:55

1aUW3. Computing rough surface Doppler effects on broadband pulse propagation using a split-step Fourier parabolic equation model. Kevin B. Smith (Code PH/Sk, Dept. of Phys., Naval Postgraduate School, Monterey, CA 93943, kbsmith@nps.navy.mil) and Richard Ead (Autonomous Undersea Vehicles Div. Naval Undersea Warfare Ctr., Newport, RI 02841)

The current interest in broadband pulse propagation in shallow water is increasing with the need for improved active sonar systems and the growth of applications utilizing underwater acoustic communications. Such shallow-water propagation is dominated by boundary interactions. If the ocean surface is rough on the scale of the acoustic wavelength, considerable scattering can occur that can significantly influence the coherent propagation. Because the rough ocean surface is also evolving dynamically, such scattering can introduce Doppler shifting and spreading of the acoustic pulse spectrum. Following the method of F. D. Tappert and L. Nghiem-Phu [J. Acoust. Soc. Am. Suppl. 1 **77**, S101 (1985)], an exact formulation for scattering from a rough surface is introduced into the

Monterey-Miami Parabolic Equation Model [JCA **9**, 243–285 (2001)]. An algorithm is then developed for computing the solution of pulse propagation as the rough surface evolves dynamically. The result can be shown to produce the proper spatial scattering as well as introduce the correct Doppler shifts from the dynamic surface. A simple sinusoidal rough surface interface will be employed to test the algorithm followed by a more complex, realistic rough ocean interface. Spatial and temporal properties of the scattered field will be examined and discussed.

9:20

1aUW4. Can chaotic rays be used for long-range propagation? Michael A. Wolfson and Frank S. Henyey (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105-6698)

Fred Tappert was skeptical of the applicability of chaotic rays to long-range propagation experiments. In this talk, the best ray predictions we can make are compared to PE predictions. In particular, one expects the predictions of ‘‘deep arrivals’’ to be the most robust to uncertainties in how to use ray information, and the ray predictions fail to agree with the PE distribution of the intensity of the deep arrivals. The discrepancy can be seen in the propagation over the first 50 km from an assumed single mode source array. A comparison with a toy model shows that the difficulty with the rays is that the scattering is too weak, in the sense that almost all of the phase fluctuations in a correlation length are smaller than a cycle.

Contributed Papers

9:45

1aUW5. Chaos and wave propagation regimes. John Colosi (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Ray chaos theory and parabolic equation numerical modeling were two thrusts of Fred Tappert's research that were perpetually in tension. Fred was interested in the problem of identifying wave propagation regimes, most notably the strong focusing caustic regime and its evolution into the saturation regime. On the one hand, chaos theory held the seed of the complexity Fred believed existed in ocean acoustic wavefields; on the other hand ocean acoustic ray chaos theory (which Fred helped to pioneer) was a disdainful approximation to the full wave treatments offered by parabolic equation calculations. Fred was convinced that the saturation limit could not be obtained using ray theory and therefore he examined a new field of inquiry: a blend of chaotic ray insight and full wave dynamics called wave chaos. This talk will discuss some of Fred's insights on this topic and how they relate to observations from basin scale acoustic transmissions.

10:00–10:15 Break

10:15

1aUW6. Dozier–Tappert theory for long-range propagation. Frank S. Henyey and Michael A. Wolfson (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105-6698)

Among Fred Tappert's contributions to ocean acoustics is the mode transport equation in the paper he wrote with Dozier. In light of the results reported by Wolfson (this session), we need to have a theoretical approach other than chaotic rays for long-range propagation experiments; mode transport is a promising candidate. The principal assumptions made by Dozier and Tappert are the Markov approximation and the assumption that cross-mode coherence is small. Although eminently reasonable at the time they wrote their paper, the latter approximation must be questioned due to newer data and simulations. Deep arrivals constitute a phenomenon directly predictable from mode transport equations that extend that of Dozier and Tappert by including travel time information.

10:30

1aUW7. Low-frequency phase rate, source–receiver motion, and the parabolic approximation. George V. Frisk (Appl. Ocean Phys. & Eng. Dept., M.S. #11, Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, gfrisk@whoi.edu) and Travis L. Poole (MIT/WHOI Joint Prog. in Oceanogr. and Oceanogr. Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

A theory relating the phase rate of low-frequency cw signals to the range rate between the source and receiver is presented. A key component of this theory is the parabolic approximation, which was introduced into underwater acoustics by Tappert. It is shown that, after shifting the received signal to the base band, the leading-order behavior of the residual phase rate is simply equal to the product of a typical wavenumber in the water column and the source–receiver range rate. This result holds true even for situations where the acoustic field magnitude clearly displays a complicated multimodal (or multipath) interference pattern. The theory is supported by a variety of experimental measurements obtained in a broad range of ocean environments. These include deep- and shallow-water situations, range-independent and range-dependent cases, and short- and long-range scenarios. Implications of the theory for phase tracking and localization of quasi-cw sources are also discussed. [Work supported by ONR.]

10:45

1aUW8. Statistics of normal-mode amplitudes in shallow water. Daniel Rouseff (Appl. Phys. Lab., College of Ocean and Fishery Sci., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Dozier and Tappert published a statistical theory of acoustic propagation in a random ocean [J. Acoust. Soc. Am. **63**, 353–365 (1978)]. Their theory predicts the statistical moments of the acoustic mode amplitudes. The acoustic modes are coupled by random variability in the water column. Creamer extended the theory to include bottom loss, an important factor in shallow-water applications [J. Acoust. Soc. Am. **99**, 2825–2838 (1996)]. In the present paper, the extended theory is applied to propagation through a realistic shallow-water internal wave model. An efficient computational algorithm is developed that makes practical the theory's application to observed sound speed and buoyancy profiles. Model parameters such as bottom loss and internal-wave strength can be varied with minimal computational cost. Numerical results are presented that demonstrate the competing effects of modal attenuation and mode coupling. [Work supported by ONR.]

11:00

1aUW9. Coupled-mode solutions in the complex plane and benchmarking with the parabolic equation method. Steven A. Stotts, David P. Knobles, and Robert A. Koch (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029, stotts@arlut.utexas.edu)

A complex plane extension of a previously developed two-way coupled mode model is presented. Coupling coefficients based on horizontal layer propagators using complex Airy solutions are evaluated analytically [Stotts, J. Acoust. Soc. Am. **111**, 1623–1643 (2002)]. This approach significantly reduces memory storage requirements and improves the computational efficiency over numerical depth integration for the coefficients. The integral equation method introduced by Knobles [J. Acoust. Soc. Am. **96**, 1741–1747 (1994)] is used to solve the coupled equations. Benchmarking with the parabolic equation (PE) approach will be presented for several examples including the ASA continental shelf benchmark and upslope/down-slope wedge propagation. Since Tappert's introduction, PE has been accepted as the de-facto model for range variable calculations. Nevertheless, having an independent verification by a normal mode model for general environments is advantageous. Differences between real and complex k-plane solutions are emphasized. Comparisons to numerically calculated key quantities are presented. Application to a range-dependent data set recorded recently off the coast of Florida is discussed.

11:15

1aUW10. Vertical modes with sediment absorption in generalized Pekeris waveguides. Frank S. Henyey, Dajun Tang, and Stephen A. Reynolds (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

The eigenvalue spectrum for vertical modes in a system consisting of a water layer overlying sediments modeled as an infinitely deep fluid region with absorption is investigated. These eigenvalues are the square of the horizontal (complex) wavenumbers. The sound speed in the water is assumed smaller than that in the sediments. The dependence of modes on the absorption coefficient as a parameter is of primary concern. The nature of the eigenvalues is related to the amount of absorption in sediments. The main result is that as the amount of absorption is raised, leaky modes (which are not eigenvalues in the absence of absorption) become true discrete modes. This behavior, not present in perturbation theory, is physically explained. [Work supported by ONR.]

11:30

1aUW11. Coupled hydrodynamic ship wake and PE-based acoustic propagation model. Aditya Balani, R. Lee Culver, David L. Bradley, Eric G. Paterson (Appl. Res. Lab., Penn State Univ., P.O. Box 30, N. Atherton St., State College, PA 16804-0030, agb127@psu.edu), Xiao Di, and Robert F. Kunz (Penn State Univ., State College, PA 16804-0030)

A full-wave, one-way, 2D parabolic equation (PE) code is used to predict sound propagation through a complex hydrodynamic bubble field. The normalized velocity component obtained from a hydrodynamic ship wake model is the basis for constructing the bubble distribution. A three-dimensional field is used to represent the complex environment: down wake (toward or away from the ship), cross wake (perpendicular to the wake axis), and with depth. Although the hydrodynamic wake model generates three-dimensional wake-bubble fields, the PE program is a 2D model. The problem is assumed to be two-dimensional by taking various two-dimensional cuts of the wake field, which are then used in the PE simulation. The wake of a ship is a complicated and challenging environment in which to model acoustic propagation. Predictions are made for the representation of the wake at various distances from the source ship. From this study a few results are seen: (1) The sound level corresponding to the strength of a spherically spreading sound field can be obtained. (2) Sound propagation through the wake bubble field is a three-dimensional problem. (3) Small changes in void fraction can produce noticeable differences in the propagation. [Work supported by ONR under Award No. N00014-02-1-0156.]

11:45

1aUW12. Propagation modeling issues for macro/micro scale environments. John McCoy (The Catholic Univ. of America, Washington, DC)

The propagations of acoustic signals through a porous solid and through a bubble cloud are both representative of a class of experiments for environments that are observable on macro and micro length scales. Intuitive models for these propagations are described by theories for poroelastic solids and for frothy fluids, respectively. These theories reflect a different intuitive understanding of the underlying physics that applies to the micro scale, and accommodate different phenomenology that can be observed on the macro scale. A more precise understanding of the linkage between micro and macro scale observations is a fundamental problem that has attracted a number of different approaches that are usually left unrelated. Moreover, there are other phenomenology that, while observable on a macro scale, are not modeled by either theory, and other models that are developed for these other phenomenology. An example of another phenomenology is a continuous backscattering occasioned by micro scale heterogeneity, which results in a kind of localization in one dimension. An example of another model is a radiative transport model. These issues are discussed in the light of still another approach to accommodating macro/micro scale variation, this one based on a phase space, spatial filtering. [Work supported by ONR.]

MONDAY AFTERNOON, 28 APRIL 2003 ROOMS 108/109, 2:00 TO 4:45 P.M.

Session 1pAA

Architectural Acoustics: Architectural Acoustics—Research and Application

Martha M. Larson, Cochair
Kirkegaard Associates, 801 West Adams Street, 8th Floor, Chicago, Illinois 60607-3034

Lily M. Wang, Cochair
Architectural Engineering, University of Nebraska–Lincoln, 200B Peter Kiewit Institute, 1110 South 67th Street, Omaha, Nebraska 68182-0681

Contributed Papers

2:00

1pAA1. A modal approach to the prediction of the sound reduction index. Alain Tisseyre, Cécile Courné, Thomas Buzzy, and André Moulinier (Tisseyre & Associés, 16, chemin de Manel, F 31400 Toulouse, France)

The calculation of the sound reduction index in modal analysis is presented in a general way; different possible approaches are described. These calculations are done in two steps: a vibratory study to determine the transverse displacement of the plate and a study of radiation. The specificity of orthotropic plates is presented. This study led to programming a calculation algorithm. Initial hypotheses are indicated, as well as results obtained for various plates or partitions. Modal analysis calculation results are then compared to the Cremer–Sewell approach results.

2:15

1pAA2. Experimental determination of porous media properties for predicting absorption of bulk-reacting surfaces. Jerry W. Rouse, Linda P. Franzoni, and Justin W. Jaworski (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708, jwrouse@me1.egr.duke.edu)

Porous materials are commonly used and located along enclosure boundaries for the purpose of sound attenuation. An accurate representation of the boundary condition is needed for the analytical modeling of these enclosures. The proper characterization of porous materials requires the determination of the effective complex phase speed and effective complex density of the fluid during propagation through the porous material. This work presents a two-step process in which these characteristic properties are determined using a standing wave tube device. The normal incidence impedance of a porous material is obtained for two cases: a zero velocity condition, and a zero pressure condition, behind the material sample. The effective phase speed and density are then obtained from formulas containing the normal incidence impedances. For verification, standing waves are predicted theoretically using the previously obtained parameters as inputs for the case of a two-dimensional slender duct with the porous material lining a side wall. These predictions are compared to experimental measurements of sound pressure in the two-dimensional duct and good agreement is observed for the open-cell foams considered in this work.

2:30

1pAA3. Dependence of reverberant energy on source distance in enclosures. Erik Larsen and Albert S. Feng (Beckman Inst. for Adv. Sci. and Technol., Univ. of Illinois at Urbana–Champaign, 405 N. Mathews Ave., Urbana, IL 61801, elarsen@uiuc.edu)

It is known that the direct-to-reverberant energy ratio has a strong influence on speech intelligibility, and also on the performance of adaptive beamforming algorithms. It is also a cue for source distance perception. According to traditional acoustic assumptions, reverberation is homogenous and isotropic, yielding a decay of -6 of direct-to-reverberant energy ratio per doubling of source distance. However, the extent to which real-world data deviates from this ideal prediction and the underlying physical parameters that affect the outcome are unknown. A series of experiments and simulations were conducted in which the reverberant energy was determined at various distances from the sound source, in various listening rooms. It was found that reverberant energy decreased systematically with an increase in source distance, while reverberation time increased orderly. Also, source distance at which direct-to-reverberant energy ratio was zero corresponded well with the critical distance as given by standard acoustic theory. The decay of reverberant energy with distance varied for the different rooms tested, roughly between -1 and -3 dB per doubling of source distance. An evaluation of these results in terms of the room's acoustic parameters will be given.

2:45

1pAA4. Aperture size, materiality of the secondary room and listener location: Impact on the simulated impulse response of a coupled-volume concert hall. Michael Ermann, Marty E. Johnson (Virginia Tech Dept. of Architecture, 201 Cowgill Hall 0205, Blacksburg, VA 24061, mermann@vt.edu), and Byron W. Harrison (The Talaske Group, Oak Park, IL)

By adding a second room to a concert hall, and designing doors to control the sonic transparency between the two rooms, designers can create a new, coupled acoustic. Concert halls use coupling to achieve a variable, longer and distinct reverberant quality for their musicians and listeners. For this study, a coupled-volume concert hall based on an existing performing arts center is conceived and computer-modeled. It has a fixed geometric volume, form and primary-room sound absorption. Ray-tracing software simulates impulse responses, varying both aperture size and secondary-room sound absorption level, across a grid of receiver (listener) locations. The results are compared with statistical analysis that suggests a highly sensitive relationship between the double-sloped condition and the architecture of the space. This line of study aims to quantitatively and spatially correlate the double-sloped condition with (1) aperture size exposing the chamber, (2) sound absorptance in the coupled volume, and (3) listener location.

3:00

1pAA5. Relating double slope decay in coupled volumes with volume ratio, absorption ratio, and aperture size. David T. Bradley and Lily M. Wang (Architectural Eng., Univ. of Nebraska–Lincoln, 1110 S. 67th St., Omaha, NE 68182-0681, dbradley@mail.unomaha.edu)

Coupled volumes typically consist of two spaces connected through an opening known as a coupling aperture. If the absorption characteristics in the auxiliary space are such that its decay time is longer than that of the main space, late-arriving energy will be fed back into the main space by the secondary volume. This energy coupling leads to the phenomenon known as the double slope decay. This project uses computer modeling to study the effect of three architectural variables on the double slope profile from a simple coupled volume system. The three variables studied are the volume ratio between the main and secondary space, the absorption ratio between the two spaces, and the aperture size. The resulting energy decay profiles are categorized according to their degree of double slope behavior and cross-referenced with the independent variables. Quantitative results from this analysis are presented.

3:15

1pAA6. A comparison of background noise levels and reverberation times measured in unoccupied elementary classrooms. Richard D. Godfrey (Owens Corning, Sci. and Technol., 2790 Granville Rd., Granville, OH 43023)

The key performance criteria listed in ANSI S12.60-2002, Acoustical Performance Criteria, Design Requirements and Guidelines for Schools, are that the maximum background noise is limited 35 dBA, and that the maximum reverberation time is limited to 0.6 seconds in the most common classroom size. Limits on sound transmission properties of the room envelope are also made. If these performance criteria are met, each student, no matter where he or she sits in the classroom, will be in an environment that affords adequate speech intelligibility, i.e., an adequate opportunity to understand the teacher's words correctly. During the standard development process, the author had the opportunity to work with his colleges at The Ohio State University. The OSU group made measures in 34 classrooms in the Columbus area (inter-city, suburbs, and rural; public and private). The author analyzed these data, and compared the results with ANSI S12.60-2002 performance criteria. It was found that the background noise levels were around 50 dBA. This is 15 dBA above the standard requirement. A majority of the reverberation times were in line with the 0.6-second criteria, but a significant number had reverberation times around 1 second.

3:30–3:45 Break

3:45

1pAA7. The effects of stage configurations in a recital hall. Wonyoung Yang (School of Occupational and Environ. Hygiene, Univ. of British Columbia, 2206 East Mall, Vancouver, BC V6T 1Z3, Canada), Kathleen K. Hodgdon, and Jiry Tichy (Penn State Univ., State College, PA 16804)

The room acoustical parameters, the reverberation time (RT), the early decay time (EDT), the bass ratio (BR), the clarity factor (C80), the initial time delay gap (ITDG), and the interaural cross-correlation coefficient (IACC) were measured for acoustical evaluation both in the audience seats and on the stage, using six different stage configurations in a 450-seat unoccupied hall. The stage configurations consisted of combinations of varying the position of the drapes with and without the reflector at the back of the stage. The listening test was conducted in a laboratory using high fidelity headphones to verify the subjective preference for the seats and the presence of the reflector. Analysis showed that the draped area of the stage was related to the RT, EDT, C80, BR, while the ITDG was strongly dependent on the presence of the reflector. There was not a strong impact observed in the IACC for the varying configurations. In general,

the reflectors improved most of the acoustical parameters. The loudness of the musical sound was the primary parameter used to decide the subjective preference, which correlated with the C80 and the ITDG with the reflector.

4:00

1pAA8. Comparison of echo criteria for a large fan-shaped auditorium. Lance L. Locey, Timothy W. Leishman, Scott D. Sommerfeldt, and Brian E. Anderson (Dept. of Phys. and Astron., Brigham Young Univ., N281 Eyring Sci. Ctr., Provo, UT 84602, LLL6@email.byu.edu)

Complaints of perceived echoes at specific locations in a 21,000 seat fan-shaped auditorium have prompted the measurement and analysis of numerous impulse responses. The responses were first processed using the echo criterion of Niese, then using the criterion of Dietsch and Kraak. This paper compares the two criteria and explores their abilities to assess whether peaks and anomalies of the measured responses were likely to produce audible echoes in the hall. The Niese criterion was found to better predict the perception of echoes produced by broad irregular decay trends. The Dietsch and Kraak criterion was able to better predict echoes produced by sufficiently strong specular reflections. Neither criterion alone was able to fully characterize these perceptions. Results obtained for both criteria at various seat locations will be presented and compared with subjective findings.

4:15

1pAA9. Measurement of the acoustic characteristics of the concert hall at the Sydney Opera House. John Bassett and Densil Cabrera (School of Architecture, Design Sci. and Planning, The Univ. of Sydney, Sydney NSW 2006, Australia)

The Sydney Opera House Trust is considering making changes to its Concert Hall. Prior to any alteration, the Acoustics group at the University of Sydney has sought to document the hall. The main measurements were made for 48 receiver locations and 6 source locations, using omnidirectional measurement microphones, B-format (Soundfield) microphones and dummy head microphones in every receiver position. The measurements included impulse responses, anechoic music recordings, and recordings of a calibrated sound power source. Results documented by previous practitioners and researchers are described and comparison is made with the recent results.

4:30

1pAA10. A full simulation of the Quetzal echo at the Mayan pyramid of Kukulkan at Chichen Itza in Mexico. Nico F. Declercq, Joris Degrieck (Soete Lab., Dept. of Mech. Constr. & Prod., Ghent Univ., St Pietersnieuwstr 41, 9000 Gent, Belgium), Rudy Briers (KATHO, Sint Jozefstraat 1, 8820 Torhout, Belgium), and Oswald Leroy (IRC-KULAK Univ., E. Sabbelaan 53, B-8500 Kortrijk, Belgium)

It is well known that a handclap in front of the staircase of the pyramid produces an echo that sounds similar to the chirp of the Quetzal bird. This phenomenon occurs due to diffraction. There exist some publications concerning this phenomenon and even some first attempts are reported to simulate it. However, no full simulation (amplitude, frequency, time) has ever been reported before. The present work presents a simulation which is based on the theory of the diffraction of plane waves and which takes into account continuity conditions. The latter theory is the building block for an extended theory that tackles the diffraction of a spherical sound pulse. By means of these principles it is possible to entirely simulate the echo following a handclap in front of the staircase. [Work supported by The Flemish Institute for the Encouragement of the Scientific and Technological Research in Industry (I.W.T.)]

MONDAY AFTERNOON, 28 APRIL 2003 ROOMS 103/104, 1:00 TO 5:20 P.M.

Session 1pAO

Acoustical Oceanography, Underwater Acoustics and Signal Processing in Acoustics: Geoacoustic Inversion I

Peter Gerstoft, Chair
Scripps Institution of Oceanography, Marine Physical Laboratory, University of California, San Diego, 9500 Gillman Drive, La Jolla, California 92093-0238

Chair's Introduction—1:00

Invited Papers

1:05

1pAO1. Probabilistic geoacoustic inversion. Stan Dosso (SACLANT Undersea Res. Ctr., Viale S. Bartolomeo, 400 19138 La Spezia, Italy and Univ. of Victoria, Victoria, BC, Canada, sdosso@uvic.ca)

The problem of estimating seabed geoacoustic parameters from ocean acoustic measurements has received considerable attention in recent years. Geoacoustic inversion represents a convenient alternative to direct measurements (e.g., coring) and provides sensitivity relevant to acoustic source localization applications; however, it requires solving a strongly nonlinear inverse problem. A variety of approaches have been developed (by a number of researchers) based on seeking geoacoustic parameters that provide the optimal match to measured acoustic fields using global search techniques. Other approaches include inversion of bottom-loss or seabed-reflectivity data and ambient noise. Topics of current interest include range-dependent inversion, coherent spatial/temporal processing, and uncertainty estimation. This paper reviews the above approaches in terms of a general probabilistic formulation for geoacoustic inversion. The goals of the probabilistic approach are to fit the acoustic data and available prior information to within their uncer-

tainties, and to estimate geoacoustic parameters, their uncertainties, and inter-relationships. This is accomplished using a Bayesian formulation and Markov chain Monte Carlo approach (Gibbs sampling) to extract features of the posterior probability density such as the maximum *a posteriori* estimate, marginal probability distributions, and correlations. The approach is illustrated for matched-field inversion, inversion of seabed reflectivity, and source localization with environmental uncertainty.

2:05

1pAO2. Using measured acoustic data to obtain estimates of seabed physical parameters. D. P. Knobles (Appl. Res. Labs., Univ. of Texas, P.O. Box 80, Austin, TX 78713-8029, knobles@arlut.utexas.edu)

Inversion for seabed parameters is part of a broader analysis that seeks to characterize the nature of and to quantify the predictability of sound propagation in complex shallow seas. This talk examines results obtained from numerous inversions using real data and attempts to place inversion techniques in perspective as a tool for understanding the physics of the interaction of sound with various types of seabeds. Data sets include those obtained in the Gulf of Mexico, off the eastern coast of the United States, and off the Korean Peninsula. Acoustic data include multifrequency measurements of transmission loss versus range and time series generated by both impulsive sources and surface ships received on both HLA and VLA. The issue of uniqueness is addressed by a combination of methods that include the comparison of inverted results to geophysical measurements, cost function sensitivity, and a comparison of modeled results to independent data not used in the inversion and localization. Of particular interest is the attenuation structure in both hard and soft sediments and the acoustic data required to extract this information. [Work supported by ONR.]

2:25

1pAO3. Measures of uncertainty in geoacoustic inversion. N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3055, Victoria, BC V8W 3P6, Canada, chapman@uvic.ca)

Inversion methods for estimating geoacoustic model parameters from acoustic field data have been applied with considerable success over the past decade. The most effective methods that have been developed generally fall into two categories, nonlinear methods that are posed as optimization problems, and linearized methods that invert quantities such as horizontal wave numbers that are derived from the acoustic field data. For either case, the complete solution of the geoacoustic inverse problem requires a measure of the error for the estimated parameter, as well as the estimate itself. This paper describes an approach for specifying an error measure in nonlinear inversion processes based on matched field processing. The inversion uses an optimization algorithm that combines global and local search processes to sample the model parameter space. An effective error measure is obtained from information in scatter plots of the cost function versus parameter values for models that were tested in the search process. Examples are presented to demonstrate the application of the method to test cases from the recent Geoacoustic Inversion Benchmark Workshop, and from experimental data from vertical line arrays and seafloor horizontal arrays. [Work supported by ONR.]

2:45

1pAO4. Geoacoustic inversion of data collected in two "typical" shallow-water environments. Peter Nielsen (SACLANT Undersea Res. Ctr., Viale San Bartolomeo 400, 19138 La Spezia, Italy) and Martin Siderius (SAIC, San Diego, CA 92121)

Two distinct types of shallow-water experiments were recently conducted by SACLANTCEN in different shallow-water areas (<150 m) to perform geoacoustic inversion of received acoustic signals. A towed sound source–horizontal line array was used in one of the experiments, and in the other the source and vertical array were moored on the bottom. The two experimental configurations were deployed in two areas with soft and hard bottom properties in order to investigate (1) the performance of geoacoustic inversion using horizontal and vertical line arrays, and (2) the robustness of the inversion method to different bottom characteristics. The transmitted signals were 1-s linear frequency modulated sweeps in the band from 200–800 Hz. The source–horizontal array separation was 300 m and the fixed path between the source and the vertical array was 2 km. This short source–receiver separation minimizes the impact of the range-dependent water-column and seabed properties on the acoustic propagation. The applied geoacoustic inversion technique of the data is the traditional model-based matched-field processing utilizing a global search algorithm. Results from successful geoacoustic inversion in the regions with soft-bottom properties are presented for the two experimental configurations. Severe difficulties appeared when inverting data from the hard-bottom regions and possible causes are given.

3:05–3:20 Break

Contributed Papers

3:20

1pAO5. Optimal search strategies for nonlinear, multi-parameter geoacoustic inverse problems. Gopu R. Potty, James H. Miller, Colin J. Lazauski, and Chuen-Song Chen (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882)

A mode travel time based technique for geoacoustic inversions has been developed using broadband explosive sources [Potty *et al.*, J. Acoust. Soc. Am. **108**, 973–986 (2000)]. In this inversion scheme we search for various parameters including water depth, sound speed in the water column represented by Empirical Orthogonal Function (EOFs), compressional wave speed in the sediment and sub-base, thickness of the sediment and range. The sensitivity of the objective function to perturbations in these parameters is different, i.e., different parameters have different sensitivities. The most sensitive parameters can be estimated with higher accuracy compared to parameters with low sensitivities. In this study we aim to link the search process to parameter sensitivities by matching the fineness of search to sensitivities. The highly sensitive parameters will be searched with a fine scale sampling of the model space with a coarser sampling adopted for less sensitive model parameters. This condition will be incorporated into the search tool (Genetic Algorithm) used for the inversion scheme. The objective of this approach is to utilize the computational resources in an optimum manner by giving more emphasis to parameters which are more sensitive as opposed to a uniform weighting for all parameters. This approach will be applied to data from the ASIAEX-2001 and Shelf Break Primer data sets. [Work supported by ONR.]

3:35

1pAO6. Geoacoustic inversion techniques (GAIT) Version 1.0 global search (GS). Peter Neumann (Planning Systems, Inc., 408 Alleghany St., Blacksburg, VA 24060-5007) and Gregory Muncill (Planning Systems, Inc., Reston, VA 20191-3453)

Geoacoustic Inversion Techniques (GAIT) Version 1.0 is a PEO (C4I and Space) PMW 155 funded product that accepts measured acoustic data and produces an optimized estimate of the bottom environment that produced the observed acoustic data. The Global Search (GS) segment of GAIT pairs the Adaptive Simulated Annealing (ASA) algorithm with a variety of Navy standard propagation loss models (PE, ASTRAL and Nautilus) and an active sonar performance prediction model (ASPM). The goal of the GS segment of GAIT is to provide a best estimate of the geoacoustic properties of the ocean bottom that, when paired with a selected model, result in the observed acoustic data. An overview of the GS segment of GAIT 1.0 will be presented with details on the ASA algorithm, component models, cost functions and geoacoustic parametrizations. Inversion results will be shown for synthetic test cases from the Inversion Technique Workshop (ITW) held in May 2001 and from both narrowband and broadband measured data test cases. [Work supported by PEO (C4I and Space) PMW 155 and uses the products of a Phase I and II SBIR from the ONR (Code 321US).]

3:50

1pAO7. Active rapid geo-acoustic characterization. Kevin D. Heaney (Orincon Corp., 4350 N. Fairfax Dr., Ste. 470, Arlington, VA 22203)

There is a growing body of evidence that the use of archival database values to estimate acoustic propagation and reverberation leads to substantial errors. To address this issue, ideas have been proposed to use mono-static reverberation to perform simultaneous geo-acoustic and scattering strength inversions. The difficulty with this technique has to do with the underlying physics of propagation and reverberation. The current techniques of using mono-static reverberation to invert for scattering strength and the geo-acoustic parameters are fundamentally limited by the fact that all reverberant energy must propagate from the source to the bottom and then back to the receiver. Thus all reverberant energy contains twice the transmission loss (TL). This ambiguity is un-resolvable from only a direct mono-static reverberation measurement. This ambiguity represents itself as a 1 dB error in estimation of the TL yields a 2 dB error in the scattering strength (SS). Further, an accurate sediment characterization and scattering strength determination, handling the range-dependence of these variables, is required. To address this problem, and to break the inherent degeneracy, a combined reverb direct blast inversion is proposed. This approach is applied to several sets of data from the Harsh Environmental Program (HEP).

4:05

1pAO8. Frechet derivatives for shallow water ocean acoustic inverse problems. Robert I. Odom (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

For any inverse problem, finding a model fitting the data is only half the problem. Most inverse problems of interest in ocean acoustics yield nonunique model solutions, and involve inevitable trade-offs between model and data resolution and variance. Problems of uniqueness and resolution and variance trade-offs can be addressed by examining the Frechet derivatives of the model-data functional with respect to the model variables. Tarantola [*Inverse Problem Theory* (Elsevier, Amsterdam, 1987), p. 613] published analytical formulas for the basic derivatives, e.g., derivatives of pressure with respect to elastic moduli and density. Other derivatives of interest, such as the derivative of transmission loss with respect to attenuation, can be easily constructed using the chain rule. For a range independent medium the analytical formulas involve only the Green's function and the vertical derivative of the Green's function for the medium. A crucial advantage of the analytical formulas for the Frechet derivatives over numerical differencing is that they can be computed with a single pass of any program which supplies the Green's function. Various derivatives of interest in shallow water ocean acoustics are presented and illustrated by an application to the sensitivity of measured pressure to shallow water sediment properties. [Work supported by ONR.]

4:20

1pAO9. Estimation of error bounds in geoacoustic inversions. William Sanders (Naval Res. Lab., Code 7185, Stennis Space Center, MS 39529)

Geoacoustic inversion has been shown to yield accurate estimates of effective ocean bottom parameters for simple environments when high fidelity models are used for both environmental deconstruction and signal propagation. But as the environment becomes more complex, some parameters are estimated with less certainty than others. Uncertainty, as represented by errors in inverse problems stem from measurement inaccuracies, model imperfections and prior assumptions. Whereas the propagation of errors in inverse problems is not generally possible, the problem becomes tractable if the forward equation can be linearized about some (preferably the true) set of values and, if further, all errors are assumed Gaussian. Under these assumptions, the covariances of the *a posteriori* errors can be formulated, thus providing bounds on the uncertainty resulting from the inverse process. These are ultimately expressed in terms of bounds on measurement errors, modeling errors and the linearization of the forward model. This effort analyzes the errors involved in some well understood and benchmarked cases and compares results to other published analyses. This is done by utilizing a parabolic equation (PE) propagation model. Derivatives of the field with respect to the environmental variables are derived in order to calculate the error bounds.

4:35

1pAO10. Two methods for solving a 3D acoustic inverse scattering problem. Fadoulourahmane Seydou, Nail A. Gumerov, and Ramani Duraiswami (Perceptual Interfaces and Reality Lab., Inst. for Adv. Computer Studies, Univ. of Maryland, College Park, MD 20742)

We consider the problem of finding the refractive index of a buried object by using far-field measurements in an inhomogeneous medium. We describe two methods for solving the inverse problem. Both methods are implemented in two steps in order to better deal with the ill-posedness of the problem. In the first method an integral equation of the first kind is derived for the far-field operator which is solved via least-squares and Tikhonov regularization. We then use the solution of the integral equation to derive an over-posed boundary value problem, i.e., the Helmholtz equation in a bounded domain with Cauchy data on the boundary. The index that satisfies this over-posed problem most closely is obtained via the Levenberg–Marquardt algorithm. The second method is an iterative method and is based on the Lippmann–Schwinger equation. It is implemented via the Newton method. The first step consists, as for the other method. Here we use a Fourier integral approach and regularization via discretization. The second step is to obtain the index by iterating the Lippmann–Schwinger equation starting with the Born approximation.

4:50

1pAO11. Ray-based geoacoustic inversion for high frequency broad band data. Martin Siderius, Paul Hursky, and Michael Porter (SAIC, 10260 Campus Point Dr., San Diego, CA 92121)

One of the difficulties in making reliable acoustic propagation predictions in shallow water is the lack of good information about the seabed type. In recent years, matched field processing- (MFP-) based geoacoustic inversion has been shown as a practical technique for estimating properties of the seabed. The MFP inversion method compares measured acoustic fields to those generated using an acoustic propagation model. Often, thousands of forward model calculations are required to find a set of seabed parameters that correlate well with the measured data. The large number of forward model calculations is computationally demanding and this is made worse when matching at higher frequencies or over broad band data. Ray-based propagation modeling relieves some of the computational burden since the calculation time is fairly insensitive to frequency and is inherently broad band. Further, the ray arrival amplitudes and delays are well suited for interpolation and this allows the seabed parameter search space to be explored using just a few ray trace calculations. The broad band nature of the modeled data provides flexibility in choosing correlation functions and this allows for more robust inversions. In this presentation, techniques using ray-based propagation modeling will be applied to the geoacoustic inversion problem.

5:05

1pAO12. Geoacoustic inversion of range-dependent shallow-water data using a complex plane-wave reflection coefficient approach. Steven A. Stotts, David P. Knobles, and Robert A. Koch (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029, stotts@arlut.utexas.edu)

The inversion of range-dependent broadband data in shallow water is presented using a previously developed approach with rays and complex bottom plane-wave reflection coefficients [Stotts, J. Acoust. Soc. Am. **109**, 2334 (2001)], allowing separation of seabed physics from water column contributions. Using geometrical optics, waterborne eigenray characteristics are calculated once and stored. Model fields are calculated by including reflection coefficient contributions from each bottom interaction and are evaluated for each perturbation of the seabed properties using simulated annealing to minimize a Bartlett-type cost function. Inversion results from range-independent and -dependent data taken during recent experiments off the coast of Florida are presented using a two-layer model. Implosive source broadband data and XBTs were recorded out to 10 km from a bottom-mounted, 52-element array 229-m deep. Grab samples obtained during the experiment show a thin, hard, crusted surface sediment. Inversions reveal an underlying softer sediment. Model time series using one inverted parameter set are compared to the data obtained from sources at different ranges. Data arrivals are compared to eigenray arrivals providing further propagation insight. Consistent inversion results from other forward models are discussed.

MONDAY AFTERNOON, 28 APRIL 2003 ROOM 208, 1:15 TO 3:50 P.M.

Session 1pBB

Biomedical Ultrasound/Bioresponse to Vibration, Physical Acoustics and ASA Committee on Standards: Acoustic Cavitation

E. Carr Everbach, Chair

Department of Engineering, Swarthmore College, 500 College Avenue, Swarthmore, Pennsylvania 19081

Chair's Introduction—1:15

Invited Papers

1:25

1pBB1. Acoustic cavitation movies. Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

Acoustic cavitation is a phenomenon that occurs on microsecond time scales and micron length scales, yet, it has many macroscopic manifestations. Accordingly, it is often difficult, at least for the author, to form realistic physical descriptions of the specific mechanisms through which it expresses itself in our macroscopic world. For example, there are still many who believe that cavitation erosion is due to the shock wave that is emitted by bubble implosion, rather than the liquid jet created on asymmetric collapse . . . and they may be right. Over the years, the author has accumulated a number of movies and high-speed photographs of cavitation activity, which he uses to form his own visual references. In the time allotted, he will show a number of these movies and photographs and discuss their relevance to existing technological problems. A limited number of CDs containing the presented materials will be available to interested individuals. [Work supported in part by the NIH, USAMRMC, and the ONR.]

1:45

1pBB2. Microbubble dynamics in the ultrasound field. Yoichiro Matsumoto (Dept. of Mech. Eng., The Univ. of Tokyo, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, ymats@tkd.att.ne.jp)

Ultrasonic biomedical applications such as ultrasound imaging with microbubble contrast agents, sonodynamic therapy, and so on, have attracted much attention in recent years. These phenomena have a close relationship to the behavior of microbubbles, so that it is essential to understanding the microbubble dynamics. A microbubble's motion in ultrasound field is simulated in detail by the full equations for mass, momentum, and energy in both phases. It is revealed that the bubble motion is much influenced by internal phenomena such as thermal diffusion, mist formation, mass diffusion, heat and mass transfer through the bubble wall. As the driving acoustic amplitude is increased, steep increases of the oscillation amplitude occur with the largest radius observed around the resonant bubble radius. Subharmonic and higher harmonic oscillations become obvious. The acoustic and thermal radiation from the microbubbles are analyzed and it is found that considerably stronger higher harmonic emissions are obtained from smaller bubbles than for the resonant bubbles.

2:05

1pBB3. Quantitative theoretical explanation of Apfel's experimental phase diagrams for sonoluminescing bubbles. Detlef Lohse (Dept. of Phys., Univ. of Twente, P.O. Box 217, 7500 AE Enschede, The Netherlands)

Robert Apfel had an enormous impact on the research of single bubble sonoluminescence [M. P. Brenner, S. Hilgenfeldt, and D. Lohse, Rev. Mod. Phys. **74**, 425 (2002)], the light emission of a single sound-driven bubble. In 1996, at the ASA Meeting in Hawaii, he posed a challenge to the theoreticians in the field: make experimentally testable predictions on single bubble sonoluminescence. Apfel collected the predictions and gave a wonderful review talk on the state of the field. Later, he came back several times to that list, comparing the predictions with latest experimental results. Our own predictions in those days referred to the phase diagrams of single bubble sonoluminescence. Later Apfel himself, together with Ketterling, measured those phase diagrams experimentally [J. A. Ketterling and R. E. Apfel, Phys. Rev. Lett. **81**, 4991 (1998); Phys. Rev. E **61**, 3832 (2000)]. Though qualitatively our 1996 predictions turned out to be correct, a full quantitative model could only be developed recently [R. Toegel and D. Lohse, J. Chem. Phys. (in press); B. D. Storey and A. J. Szeri, Phys. Rev. Lett. **88**, 074301 (2002)]. In the presentation we will compare the model predictions with Apfel's data.

2:25

1pBB4. The oscillations of vapor bubbles. Andrea Prosperetti, Z. Yin, and B. Yang (Dept. of Mech. Eng., Johns Hopkins Univ., 119 Latrobe Hall, 3400 N. Charles St., Baltimore, MD 21218, prosperetti@jhu.edu)

Bob Apfel had so many interests that it is impossible—however fitting and desirable—to pay homage to his work as a whole. Some of his early studies were devoted to bubble nucleation at high superheats. In the first part of this paper a recent application of this phenomenon is described. Once a vapor bubble is generated, its subsequent oscillations (free and forced) present analogies and differences with those of a gas bubble: the second part of the paper focuses on this topic. [Work supported by NSF and NASA.]

2:45

1pBB5. What are the limits of energy focusing in sonoluminescence? Seth Putterman, C. Camara, B. Kappus, C. K. Su (Phys. Dept., UCLA, Los Angeles CA 90095, puherman@ritva.physics.ucla.edu), and E. Kirilov

Sonoluminescence [SL] is amazing for the extraordinary degree by which ultrasonic energy can be focused by a cavitating bubble. Local energy dissipation exceeds Kirkhoff's law by 1E15 and the acoustic energy density concentrates by 12 orders of magnitude to create picosecond flashes of broadband ultraviolet light. At the minimum bubble radius, the acceleration exceeds 1E11 g and a megabar level shock wave is emitted into the surrounding fluid. For single bubbles driven at 30 KHz, SL is nature's smallest blackbody. This implies that the bubble's interior is such a dense plasma that the photon-matter mean free path is shorter than the wavelength of light, and suggests that SL originates in an unusual state of matter. Excitation of a vertical column of fluid [$\sim$10 Hz] so as to create a water hammer leads to the upscaling of SL and generation of flashes of light with 3E8 photons and peak powers approaching 1 W. At 1 MHz, the spectrum resembles bremsstrahlung from a transparent plasma with a temperature $\sim$1 MK. At 10 MHz the collapsed size of the SL bubble approaches 10 nm, which raises the possibility that the SL parameter space may extend to the domain of quantum mechanics. [Research supported by DARPA and DOE.]

Contributed Papers

3:05

1pBB6. Light-scattering measurements of cavitation from ultrasound contrast agent bubbles. Jingfeng Guan, Wen-Shiang Chen, and Thomas Matula (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@apl.washington.edu)

Ultrasound contrast agents are typically micron-sized gas bodies with stabilizing shell coatings. The shell prevents the gas bubble from dissolving. The shell also changes a bubble's scattering properties and affects a bubble's destruction. Optimization of the shell can be beneficial for diagnostic imaging and therapeutic ultrasound applications. Thus, understanding agent response to pulsed ultrasound is important. However, because the agents are so small, their response to pulsed ultrasound is difficult to measure. Until recently, the only tool available for measuring contrast agent dynamics was a very expensive high-speed camera. We have been investigating the use of light-scattering to measure aspects of cavitation from contrast agents in which a small HeNe laser beam scatters off a bubble and is focused onto a photomultiplier tube detector. Previous experiments have involved clusters of agents, where collective bubble oscillations and destruction were observed. Our current apparatus is designed to observe single-bubble oscillations from a diagnostic ultrasound system. The advantages for using light-scattering include small cost and the ability to monitor bubble oscillations for extended times. The technique will be described, and representative data samples will be shown.

3:20

1pBB7. Use of intensity interferometry to determine the size of a cavitation hot-spot. Carlos Camara, Seth Putterman (Phys. Dept., Univ. of California, Los Angeles, CA 90095-1547, camara@physics.ucla.edu), Keith Weninger (Stanford Univ., Stanford, CA 94305-4060), and Paul Evans (Dept. of Mater. Sci. Eng., Wisconsin Univ., Madison, WI 53706)

Although it is generally accepted that a sonoluminescing bubble collapses to a volume determined by the van Der Waals hard core of its contents, the size of the light-emitting region remains undetermined. Photos of the hot-spot indicate that the resolution of its size is diffraction limited. Intensity interferometry can resolve smaller sizes as well as the flash width if the light is incoherent. An experiment aimed at exploiting this is underway.

3:35

1pBB8. Nonlinear bubble dynamics in viscoelastic media. Stanislav Y. Emelianov (Dept. of Biomed. Eng., Univ. of Texas, Austin, TX 78712-1084), Mark F. Hamilton, Yurii A. Ilinskii, and Evgenia A. Zabolotskaya (Univ. of Texas, Austin, TX 78712-1063)

Microbubbles are already widely used for biomedical and clinical purposes, and are further being investigated for innovative medical and industrial applications. In most cases, the Rayleigh–Plesset equation for a gas bubble in liquid is used. In this paper we present a more general equation for gas bubble dynamics in viscoelastic media such as soft tissue.

The equation, describing oscillations of a spherical bubble in an isotropic, incompressible medium, contains an integral over an elastic energy density function that depends on two invariants of the strain tensor. For moderate bubble oscillations one may use Landau's cubic expansion of energy density, which contains five elastic moduli—shear and bulk moduli corresponding to quadratic terms, and three moduli (A, B, and C) corresponding to cubic terms. With energy density expressed through invariants of the strain tensor in spherical coordinates, the total elastic energy for an incompressible medium becomes an explicit function of bubble radius and contains only two of the five moduli, the shear modulus and modulus A. Compared to the Rayleigh–Plesset equation, the resulting equation has additional terms associated with shear force and elastic nonlinearity. Further modifications of this equation account for weak compressibility, viscosity, surface tension, radiation loss, and elastic shells.

MONDAY AFTERNOON, 28 APRIL 2003 ROOMS 209/210, 1:00 TO 4:35 P.M.

Session 1pNS

Noise, ASA Committee on Standards, Signal Processing in Acoustics and Architectural Acoustics: Dan Johnson Retrospective

Mary M. Prince, Cochair
National Institute for Occupational Safety and Health, 4676 Columbia Parkway, M.S. R-16, Cincinnati, Ohio 45226

John P. Seiler, Cochair
Mine Safety and Health Administration, Cochran Mill Road, Building 038, P.O. Box 1823, Pittsburgh, Pennsylvania 15236

Chair's Introduction—1:00

Invited Papers

1:05

1pNS1. Dan Johnson—Mr. Standards. Paul Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

Dan Johnson had a long history of involvement in the standards program of ASA. He has chaired and/or participated in many working groups, primarily in the areas of hearing conservation and instrumentation. He was one of a rare set that was an individual expert to all four S-committees, being very familiar with instrument design from his days at Larson-Davis, and being familiar with physiological and psychological effects, noise, and vibratory effects from his work with the Air Force at Wright–Patterson AFB. Dan was the chair of S-12, the vice-chair of ASACOS, and, most recently, the chair of ASACOS and Standards Director. As Standards Director, Dan guided us through many turbulent times including establishment of the Standards Store, the change of the standards manager, the budgetary trials of 2000 and 2001, and the appeal of ANSI S12.60-2002 (Classroom Acoustics) by ARI before Accredited Standards Committee S12. He is very missed.

1:25

1pNS2. Monitoring audiometry for occupational hearing loss: A case for eliminating 500 Hz. Mark Stephenson (Natl. Inst. for Occupational Safety and Health, 4676 Columbia Pkwy., Cincinnati, OH 45226)

Audiometric monitoring is an important element in hearing conservation programs. Nearly every existing hearing conservation standard dictate that hearing thresholds should be measured at specific frequencies, and that 500 Hz be among those frequencies tested. Actual and estimated noise-induced permanent threshold shifts were evaluated as a function of exposure duration and exposure level. The results demonstrate 500 Hz to be of little value in assessing noise-induced hearing loss for typical industrial noise exposures of up to 40 years, at least for time-weighted average exposures of up to 100 dB A. Furthermore, few hearing conservation programs currently require audiometric monitoring to be performed in an environment that meets ANSI standards for maximum permissible background noise levels. This is particularly likely to compromise hearing testing at 500 Hz. As a result, this paper argues against the need for testing at 500 Hz, and recommends it be eliminated as a required test frequency in audiometric monitoring for noise-induced occupational hearing loss.

1:45

1pNS3. Predicting noise-induced hearing loss in human populations. The contribution of the ISO-1999 and ANSI S3.44 standards. Mary M. Prince (Centers for Disease Control, Natl. Inst. for Occupational Safety and Health, 4676 Columbia Pkwy., R-16, Cincinnati, OH 45226)

This paper reviews the ISO-1999 and ANSI S3.44 standards and the literature related to prediction of noise-induced hearing loss (NIHL) in humans. The dearth of contemporary population-based studies of hearing loss among industrial, low noise-exposed individuals have made the ISO-1999 and ANSI S3.44 standards a key source of comparative data for assessing the risk of NIHL. This paper will highlight how these standards have been used in several published papers to examine the magnitude of NIHL risk, predictions of noise-induced thresholds in hearing conservation data, and to generate hypotheses regarding the biologic plausibility of hearing loss due to chemicals or other ototoxic agents.

2:05

1pNS4. Dan Johnson's impact on hearing research. Lawrence I. Shotland (James H. Quillen VA Medical Ctr., Mountain Home, TN and Dept. of Communicative Disord., East Tennessee State Univ., Johnson City, TN)

Daniel L. Johnson is well known for his many technical contributions to noise research. Throughout a long and distinguished career at Wright–Patterson Air Force Base, Dan published the results of several significant experiments, including his landmark experiments on asymptotic threshold shift and exposure to impulse noise. His work in the area of noise exposure laid much of the groundwork for a greater understanding of the physiologic response to hazardous noise, much of which has since been incorporated in national and international standards. Dan is highly regarded for his tireless work on technical and advisory committees in noise, and most recently, ototoxicity. Throughout his career, Dan has adhered to a self-imposed standard of intellectual honesty and discovery. Dan's most recent endeavor, the development of a personal noise dosimeter designed for self-monitoring by the employee, is characteristic of his creativity and energy. Perhaps less well known are his contributions over the years to the success of his younger colleagues. He has accomplished this in an unselfish and egalitarian manner, oftentimes challenging and even contradicting his own research. The focus of this talk will elaborate on these facets of Dan's professional contributions.

2:25

1pNS5. A CHAT with Dan Johnson. Robert A. Bertrand (Bertrand Johnson Acoust., Inc., 5995 Gouin Blvd., Montreal, QC H4J 2P8, Canada)

Dan's preoccupation with preventing NIHL is well known. For those of us who had the occasion to CHAT with him, we often heard him say ''if ears would bleed when exposed to noise, people would pay more attention to the harmful effect of noise upon hearing.'' His objective, as he often stated, was to eradicate NIHL so that in a few decades, it would become a historical footnote. Among his many preoccupations in his illustrious career, approaches and techniques to prevent NIHL were of primordial importance. One approach he advocated is the use of TTS instead of the STS in HCP's. His intention was to use the identification of TTS as an easy approach to introduce appropriate measures to prevent NIHL at a stage of reversibility rather than waiting for a confirmed permanent hearing loss, as noted with a confirmed STS. One of his last projects was developing the CHAT (Change of Hearing Audio Test) for easy use both in industrial and environmental settings to identify subjects with a TTS. Several groups are interested in pursuing his aim of using the TTS in HCP's, hoping to fulfill his comment that in a few decades, NIHL will become a historical footnote.

2:45

1pNS6. Dan Johnson the mentor. Richard McKinley (Air Force Res. Lab., 2610 Seventh St., Wright–Patterson Air Force Base, OH 45433)

I first met Dan Johnson in early 1975 as I was interviewing for an engineering job with Henning von Gierke's bioengineering and bionics laboratory at Wright–Patterson Air Force Base. From the very beginning Dan was always direct and forthright. Over the ensuing next 27 years my knowledge and respect of Dan constantly grew. This presentation will review Dan's technical and personal contributions while at the laboratory at Wright–Patterson Air Force Base. He was instrumental in the development of a national noise exposure criteria with the equal-energy-rule, an accurate single number hearing protector attenuation measure based on ''C-A,'' an impulse noise exposure criteria, a longitudinal study of hearing loss in children, development of noise dosimeters, and description of hearing damage risk from nonoccupational noise exposures such as disco's, bowling alleys, lawn mowers, and school buses. Dan has had a significant effect on my career. I and the many people who knew him at the laboratory miss him greatly.

3:05–3:20 Break

Contributed Papers

3:20

1pNS7. Personal noise exposure assessment from small firearms. Chucri A. Kardous, William J. Murphy, and Robert D. Willson (NIOSH, Hearing Loss Prevention Section, 4676 Columbia Pkwy., M.S. C-27, Cincinnati, OH 45226-1998)

The National Institute for Occupational Safety and Health conducted noise exposure evaluations of law-enforcement personnel during firearms training at indoor and outdoor firing ranges. A representative cross section of weapons used by officers was measured. Shooters participated in live-fire exercise at an indoor firing range using three different weapons: a Beretta .400 caliber pistol, a Remington 12-gauge shotgun, and an M4 .223-caliber assault rifle. Indoor and outdoor measurements were obtained for the Smith and Wesson .357 pistol and Colt .450 and 9-mm pistols, the Glock .400 pistol, and the Heckler and Koch and Colt AR15 .223 rifles. Impulses were measured using a Bruel and Kjaer 4136 1/4-in. microphone and TASCAM digital audio tape recorder. Relevant impulse noise metrics were calculated. Peak levels ranged from 155 to 168 dB SPL. A-weighted equivalent levels ranged from 124 to 128 dBA. The contributions of the secondary weapon firings were approximately 1 to 9 dBA. Other parameters such as A/B durations, number and mixture of impulses, spectral content, energy, kurtosis, temporal spacing, and hearing protectors' effectiveness were examined. Comparisons of applicable damage risk criteria are presented. Further studies are needed to establish an occupational impulse noise damage risk criterion.

3:35

1pNS8. Attenuation measurements of passive linear and nonlinear hearing protectors for impulse noise. William J. Murphy and Chucri A. Kardous (NIOSH, Hearing Loss Prevention Section, 4676 Columbia Pkwy., M.S. C-27, Cincinnati, OH 45226-1998)

As a part of a NIOSH Health Hazard Evaluation of law-enforcement personnel, the attenuation of several types of earplugs were measured in response to impulse noise produced by small-arms gunfire. The earplugs were primarily flanged premolded plugs produced by EAR/Aearo and Bilsom/Baccou-Dalloz. Measurements for the North Sonic Ear Valve, EAR Classic earplugs, and EAR Ultra 9000 passive nonlinear ear muff were conducted. The EAR premolded earplugs were the Combat Arms passive linear and nonlinear, HiFi and Ultratech earplugs. The Bilsom devices were the 555, 655 NST, and 655 ISL earplugs. The Combat Arms and 655 ISL earplugs both utilize a cartridge developed by the French German Research Institute de Saint Louis that provides nonlinear attenuation. The peak reduction of these devices ranged between 10 and 28 dB.

The slope of peak reduction with peak level for the Ultra9000 device was about 0.5 dB/dB, while the slopes for most earplugs were about 0.1 to 0.3 dB/dB for weapons impulses between 159- and 170-dB peak level. The peak reductions ranged from 6 dB for the North Ear valve to 30 dB for the EAR Classic foam earplug.

3:50

1pNS9. Relation between peak pressure and spectrum for small caliber muzzle report and ballistic shock. Michael J. White and Larry L. Pater (U.S. Army ERDC/CERL, P.O. Box 9005, Champaign, IL 61826)

It might seem impossible to reconstruct the peak signal level of an impulsive sound from its spectrum if the phase information is not kept. Without such an estimate, sound propagation assessments made in the frequency domain are at a disadvantage when it is the peak level that is desired. From a set of experimental measurements of small caliber firearm noise at Camp Guernsey, muzzle report and ballistic shock signals were time-gated to obtain 1/3-octave sound exposure spectra and peak levels from each signal. The rms bandwidth of each signal was determined by summing the squared frequency, weighted by the relative 1/3-octave band exposures. Finally, a relation was presumed only such that the sound exposure was equal to a product of squared peak pressure, the reciprocal of the rms bandwidth, and an undetermined constant. Constants equal to 3 dB for muzzle blast and 3.5 dB for ballistic shock fit the data sets, with approximately 3 dB standard error. Greater error tended to coincide with smaller measured bandwidths, by overestimating the peak level.

4:05

1pNS10. Noise sampling issues for impact/impulse noise surveys. Mary M. Prince (NIOSH, IWSB/DSHEFS, 4676 Columbia Pkwy., R-16, Cincinnati, OH 45226) and Jeffrey S. Vipperman (Dept. of Mech. Eng., Univ. of Pittsburgh, 531 Benedum Hall, Pittsburgh, PA 15261)

Noise-induced hearing loss (NIHL) has been recognized as a serious health concern for decades. ISO Standard 1999:1990 provides a means to predict noise-induced hearing loss (NIHL) based on L_{Aeq} measurements in the working environments of workers. This standard seems to work well for predicting hearing loss in continuous noise fields. However, it is possible that ISO 1999 does not apply well to impact, impulsive, or other transient noise fields. NIOSH and University of Pittsburgh are currently developing noise-sampling strategies to measure impact and impulse noise in a manufacturing environment with the aim of developing new impulsive noise metrics. As part of the study, broadband impact/impulse pressure measurements will be made. Issues such as instrumentation, data quality, repeatability, spatial sampling, equipment portability, and calibration are addressed. Also, the annotation, digitization, and editing of the waveforms will be discussed. As part of the project, an archival database of manufacturing impulse/impact will be created to support the future algorithmic development. The ultimate goal of the project is to develop new metrics to characterize the hazards of impact/impulse noise that will complement ISO 1999 for predicting NIHL.

4:20

1pNS11. Analysis of impact/impulse noise for predicting noise induced hearing loss. Jeffrey S. Vipperman (Dept. of Mech. Eng., Univ. of Pittsburgh, 531 Benedum Hall, Pittsburgh, PA 15261), Mary M. Prince (NIOSH, IWSB/DSHEFS, Cincinnati, OH 45226), and Angela M. Flamm (Univ. of Pittsburgh, Pittsburgh, PA 15261)

Studies indicate that the statistical properties and temporal structure of the sound signal are important in determining the extent of hearing hazard. As part of a pilot study to examine hearing conservation program effectiveness, NIOSH collected noise samples of impact noise sources in an automobile stamping plant, focusing on jobs with peak sound levels (L_{pk}) of greater than 120 dB. Digital tape recordings of sounds were collected using a Type I Precision Sound Level Meter and microphone connected to a DAT tape recorder. The events were archived and processed as .wav files to extract single events of interest on CD-R media and CD audio media. A preliminary analysis of sample wavelet files was conducted to characterize each event using metrics such as the number of impulses per unit time, the repetition rate or temporal pattern of these impulses, index of peakedness, crest factor, kurtosis, coefficient of kurtosis, rise time, fall time, and peak time. The spectrum, duration, and inverse of duration for each waveform were also computed. Finally, the data were evaluated with the Auditory Hazard Assessment Algorithm (AHAAH). Improvements to data collection for a future study examining different strategies for evaluating industrial noise exposure will be discussed.

MONDAY AFTERNOON, 28 APRIL 2003 — ROOM 204, 2:00 TO 4:40 P.M.

Session 1pPP

Psychological and Physiological Acoustics: In Memory of Evan Relkin

Robert L. Smith, Chair

Institute for Sensory Research, Syracuse University, 621 Skytop Road, Syracuse, New York 13244-5290

Chair's Introduction—2:00

Invited Papers

2:10

1pPP1. Estimating the similarity of temporal discharge patterns of auditory neurons. William Shofner (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626, wshofne@luc.edu)

Behavioral data obtained from stimulus generalization experiments are often interpreted to imply how similar animal perceptions of test stimuli are to a training stimulus. An appropriate decision variable would be useful for evaluating physiological responses in the context of stimulus generalization tasks. The decision variable $P(A)$ obtained from ROC analysis of spike count distributions is useful for evaluating rate coding schemes [e.g., Relkin and Pelli, J. Acoust. Soc. Am. **82**, 1679–1691 (1987)], but it ignores temporal information which may exist in responses to periodic sounds. Neural autocorrelograms show temporal firing patterns following a spike

and were generated for cochlear nucleus unit responses to aperiodic, quasiperiodic, and periodic stimuli. Variance from average rate was computed as a function of time for each autocorrelogram; similarity between two autocorrelograms was estimated as the ratio of these variances. A value of 1 indicates that the autocorrelograms are identical, whereas larger values indicate less similarity. Ratios were around 1 if both autocorrelograms were from responses to wideband noise and were greater than 1 if one autocorrelogram was from responses to iterated rippled noises or harmonic complex tones. This ratio may be a useful decision variable for evaluating temporal coding schemes. [Work supported by NIDCD P01 DC00293.]

2:35

1pPP2. Can we learn anything new from the animal's perspective?: Bayesian stimulus estimation based on neural spike trains. B. Scott Jackson (Inst. for Sensory Res., Syracuse Univ., 621 Skytop Rd., Syracuse, NY 13244-5290, Scott_Jackson@isr.syr.edu)

The classical approach to understanding a neural coding scheme is to characterize spike trains elicited by multiple presentations of the same stimulus, an example being the rate-intensity function. However, the task for the organism is deciphering the stimulus content from spike trains elicited by single stimulus presentations, a process that is more analogous to Bayesian stimulus estimation. The usefulness of Bayesian stimulus estimation in the neural context, using the auditory periphery as a model system, was investigated. More specifically, the encoding of stimulus intensity in the spike rate of single primary auditory neurons was examined. It was found that the results of this method are heavily influenced by the *a priori* stimulus distribution and that apparent benefits of this approach, such as linearization and, in some instances, increased stimulus dynamic range, are offset by concomitant disadvantages, such as increased estimation error and decreased stimulus dynamic range in other instances. Hence, in this context, Bayesian stimulus estimation does not contribute meaningful additional knowledge to our understanding of neural coding, and these results suggest that related methodologies, such as stimulus reconstruction and information theoretic methods, be used and interpreted with caution. [Work supported by Syracuse University and NIH Grant 5-P01-DC000380.]

3:00

1pPP3. Monaural and diotic detection of tones in wideband and narrow-band reproducible noise maskers. Sean A. Davidson and Laurel H. Carney (Dept. of Bioengineering and Neurosci. and Inst. for Sensory Res., Syracuse Univ., 621 Skytop Rd., Syracuse, NY 13244, sadavids@syr.edu)

In order to test monaural and binaural processing components of binaural detection, diotic and monaural detection were studied using reproducible noises. Correlations across masker bandwidth and interaural configuration were computed. Preliminary results reveal high correlations for probability of detection across N_0S_0 and N_mS_m configurations for both wideband and narrow-band maskers, suggesting that the binaural system does not influence results across reproducible noises for diotic detection. As reported previously [Evilsizer *et al.*, J. Acoust. Soc. Am. **111**, 336–345 (2001)], significant differences in correlations between wideband and narrow-band results were observed for the diotic case, suggesting cross-frequency interactions. Preliminary findings indicate that the same result holds for the monaural case. These data will be used to test models that incorporate monaural processing based on the phase-opponency model [Carney *et al.*, Acta Acustica United with Acustica **88**, 334–347 (2002)], and binaural processing based on the Jeffress model and on the two-channel model [McAlpine *et al.*, Nature Neurosci. **4**, 396–401 (2001)]. Predictions will be made for N_mS_m, N_0S_0, and N_0S_π interaural configurations, and for wideband and narrow-band maskers. [Work supported by NIDCD.]

3:25–3:40 Break

Contributed Papers

3:40

1pPP4. The mid-duration hump in the intensity-difference limen as a function of frequency: Further evidence for the frequency–time listening window of the human ear. Lance Nizami and Walt Jesteadt (Ctr. for Hearing Res., Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

It has been suggested that the ear has a frequency–time listening window and that the stimulus that occupies the fewest frequency–time windows, i.e., that which has the smallest product of effective bandwidth and effective duration, should result in the largest intensity-difference limen (DL) [Van Schijndel *et al.*, J. Acoust. Soc. Am. **105** (1999)]. Van Schijndel *et al.* identified the Gaussian-shaped tone-pip as the optimal stimulus, obtaining DLs at 1 kHz and 4 kHz. The DLs were greatest for the tone-pips whose frequency–time window was one critical band wide and 4 cycles long in equivalent square duration D (envelope duration/ 3.19). Here, DLs for Gaussian-shaped 250-Hz tone-pips were obtained from 5 subjects for Ds of 1.25–22.57 ms at levels of 40, 50, 60, 70, 80, and 90 dB SPL. These data were combined with earlier results showing a mid-duration rise at 40–60 dB SPL for Gaussian-shaped tone-pips of 6.5 kHz, 2 kHz, and 500 Hz. The peak in the DL is sharp for 6.5 kHz but flattens as frequency drops, being replaced by a broad rise for 250 Hz. The highest DLs correspond to approximately 4 cycles in D, not far from the ringing duration calculated as the inverse of the critical bandwidth.

3:55

1pPP5. Growth of interleaved masking patterns for cochlear implant listeners at different stimulation rates. Bom Jun Kwon, Chris van den Honert, and Wendy Parkinson (Cochlear Americas, 400 Inverness Pkwy., Ste. 400, Englewood, CO 80112)

This study investigates the pattern of growth of masking (GOM) for interleaved masking with Nucleus cochlear implant users. For an interleaved masking paradigm, where the masker and probe overlap in a same time window, the masker may have contrasting effects: it may increase the threshold (as a masker normally does) or decrease it due to a neural summation effect, facilitating detection of the probe. Several stimulation rates and masker levels were tested to examine under what conditions what phenomenon would occur. The results indicated that, in most of the conditions, the amount of masking was positive, i.e., the facilitating effect was not consistently observed. However, the slope of the GOM appears to be dependent upon the stimulation rate: the higher the stimulation rate, the lower the slope, implying that the facilitating effect might be always present and make a bigger impact on overall masking as the stimulation rate becomes high. The amount of masking was also often nonzero (positive) even when the masker was below the threshold level. Overall, the present findings indicate that interleaved masking should be handled with care to understand cochlear implant users speech perception and improve speech coding, as it contains some nontraditional aspects of masking.

4:10

1pPP6. Biologically inspired robust onset detection. Leslie S. Smith (Dept. of Computing Sci. and Mathematics, Univ. of Stirling, Stirling FK9 4LA, Scotland, UK)

Onsets are rapid increases in signal strength. The common onset time in different frequency bands provides an important cue for dynamically grouping sound energy, and hence for sound streaming. Onsets are important for segmenting sounds [Smith, J. New Music Res. **23**, 11–23 (1994)] and for determining where to measure IIDs and ITDs for the sound direction finding [Smith, J. Acoust. Soc. Am. **111**, 2467 (2002)]. Effective onset detection requires low latency and the capacity to cope with wide variation in the dynamic range. Many neurons in the auditory brainstem are sensitive to onsets. The system filters sound into cochlea-like bands (using a Gammatone filterbank), then spike codes positive-going zero-crossings. A wide dynamic range is achieved by using multiple spike trains per filter band, each with different sensitivity. The spike trains from each band innervate a leaky integrate-and-fire neuron. The excitatory synapses from the spike trains are fast and depressing: the shunting inhibitory synapses are facilitating and slower. The combined effect is that the neuron produces a single spike for each onset over a wide dynamic range with very low latency. The use of both inhibitory and excitatory synapses improves onset detection over purely excitatory synapses, leading to a better sound direction finding than previously reported.

4:25

1pPP7. Otoacoustic emissions and the "set of the center:" Covariances linking ear, brainstem, and cortex, documented with the AXS Test Battery. Judith L. Lauter (Stephen F. Austin State Univ., Nacogdoches, TX 75962, jlauter@sfasu.edu)

Although otoacoustic emissions (OAEs) are believed to reflect afferent processes connecting the central auditory system with the ear, such relations have not been demonstrated in humans, for central components rostral to the lower pons. The Auditory Cross-Section (AXS) Test Battery (Lauter, 2000, 2002) provides a straightforward way to do this, with the result that one can observe covariances between any of a number of OAE variables (TEOAE amplitude, DPOAE stability, etc.), and measures of the physiological status of the auditory nerve, brainstem centers from caudal pons to midbrain, and auditory cortex. The testing methodologies employed are relatively inexpensive and should be within the means of many more clinics and laboratories than can afford access to brain-imaging technologies such as fMRI and PET. In addition, a new statistical means of testing covariation in multivariate, within-subject data sets (Ninness and Lauter, 2003) will be described that offers a means of mathematically testing the results. Data illustrating this approach will be presented, documenting covariances observed during spontaneous day-to-day systemic fluctuations, as well as responses to medications.

MONDAY AFTERNOON, 28 APRIL 2003 — ROOM 203, 1:15 TO 2:45 P.M.

Session 1pSP

Signal Processing in Acoustics and Speech Communication: Speech Processing

Douglas O'Shaughnessy, Cochair
INRS-Telecommunications, Place Bonaventure, Suite 6900, 900 de la Gauchetiere West, Montreal, Quebec H5A 1C6, Canada

Deborah M. Grove, Cochair
Zargis Medical Corporation, 755 College Road East, Princeton, New Jersey 08540

Contributed Papers

1:15

1pSP1. Robust automatic recognition of telephone speech. Douglas O'Shaughnessy and Selouani Sid-Ahmed (INRS-Telecommunications, 800 de la Gauchetiere West, Ste. 6900, Montreal, QC H5A 1K6, Canada)

A method is presented to compensate cepstral coefficients (MFCCs) in a speech recognition system for degraded telephone channel conditions. The technique proposed is based on a combination of the Karhonen–Loeve Transform (KLT) and Genetic Algorithms (GA). The idea consists of projecting the band-limited MFCCs onto a subspace generated by the genetically optimized KLT principal axes. Experiments show a clear improvement when the method was applied to the NTIMIT speech database. Word recognition results obtained on the HTK toolkit platform using N-mixture tri-phone models and a bigram language model are presented and discussed.

1:30

1pSP2. Speech enhancement using dynamic synapse neural networks. Hassan H. Namarvar and Theodore W. Berger (Dept. of BME, Univ. of Southern California, OHE-500, Los Angeles, CA 90089-1451, heidarin@usc.edu)

An idea of speech enhancement using a Dynamic Synapse Neural Network (DSNN) with an extended Kalman filtering (EKF) training method is described. The goal of this study is to introduce a new methodology in better speech enhancement in the presence of continuous environment background noise, such as fans and air-conditioning units. The efficiency of this method is shown by applying it to noisy speech signals to remove recorded laboratory noise from signals at different signal-to-noise ratio levels. The preliminary results have been encouraging enough to justify our idea. To provide more noise robustness, this could be used as a pre-processing level in automatic speech recognition (ASR) systems. The proposed method would have a profound impact on the performance of ASR systems. [Work supported by DARPA CBS, NASA, and ONR.]

1:45

1pSP3. Text-to-speech from concatenation of articulatory units derived from natural speech. Daniel J. Sinder and M. Mohan Sondhi (Avaya Labs, 233 Mt. Airy Rd., Basking Ridge, NJ 07920, sinder@avaya.com)

It has been conjectured that articulatory synthesis possesses the greatest potential for generating high quality synthetic speech. However, for text-to-speech (TTS), waveform concatenation techniques have proven more practical due in part to the challenge of generating appropriate trajectories of articulatory parameters. A waveform generation method for TTS that combines the practical success of concatenative methods with the quality potential of articulatory synthesis is under development. The system concatenates articulatory units derived from natural speech using an articulatory voice mimic. The mimic estimates articulatory parameters by minimizing a cost function that includes a spectral distance between natural and synthetic speech and a geometric distance that penalizes rapid or discontinuous changes in articulator positions. A database of articulatory trajectories representing phonetic units is constructed from the estimated parameters. For TTS, phonetic units generated by text analysis are used to select the corresponding articulatory units from the database. Duration modification, concatenation, and smoothing across units are performed in the articulatory domain resulting in a single articulatory trajectory for the complete utterance. Speech is synthesized from the trajectory using a two mass model for voicing, achieving a high degree of acoustic continuity across unit boundaries while also allowing for source–tract interaction.

2:00

1pSP4. A measure of aperiodicity content in a speech signal. Om D. Deshmukh and Carol Y. Espy-Wilson (ECE Dept., Univ. of Maryland, College Park, MD 20782)

Most of the current aperiodicity detectors measure aperiodicity indirectly, in which the absence of periodic component in a nonsilent region is termed as aperiodicity. Such indirect measurements are inadequate and can be misleading especially in cases like voiced fricatives or breathy vowels. This motivated us to develop a direct measure of aperiodicity which is independent of the periodicity measure. The speech signal is passed through a 60-channel gamma tone auditory filterbank. Average Magnitude Difference Function (AMDF) is computed on the envelope of each channel. The randomness in the distribution of the AMDF dips is the basis for the measure of the aperiodicity whereas the measure of the periodicity is based on the occurrence of the dips at multiple locations. The system was evaluated on the MOCHA database, which has simultaneous recording of the EGG data, and on the TIMIT database. Preliminary analysis shows that the aperiodicity and voicing accuracy on a per frame basis are 95% and 90.3%, respectively. In voiced fricatives and voiced stops, high aperiodicity and high periodicity were detected in 21.6% of the frames. Note that not all these sounds necessarily had both the sources prominent.

2:15

1pSP5. Spectral moments and alternative methods of characterizing fricatives. Oliver S. Blacklock and Christine H. Shadle (Dept. of Elec. and Computer Sci., Univ. of Southampton, Southampton SO17 1BJ, UK)

Parameters that both distinguish fricatives in a given language, and characterize normal and disordered productions would be useful. Various parametrizations have been tried since the 1950s with only partial success. Forrest *et al.* [J. Acoust. Soc. Am. **84**, 115–123 (1988)] defined spectral moments of speech power spectra, but found that these failed to distinguish the voiceless English fricatives. They have however proven useful for distinguishing sibilants, and for characterizing long-term place changes of sibilants in disordered speech. As discussed in this paper, other changes, and nonsibilants, are not so easily characterized. Relative amplitude differences have to date not been incorporated in spectral moment methodology. The quality of the spectral estimate (whether consistent, unbiased) affects the variability of the moments, and means of obtaining a good estimate usually involve assuming stationarity or ergodicity. Finally, moments capture little that is known about fricative production. Modern techniques for consistent PSD estimation (e.g., multitaper and wavelet analysis) have been examined; their suitability in analyzing fricatives is discussed. While these modern PSD estimation methods may improve the performance of spectral moments, they also lend themselves to new parametric methods which do not rely on assumptions of ergodicity or stationarity. [Research supported by EPSRC.]

2:30

1pSP6. Speaker identification by difference sum and correlation coefficients of narrow-band spectrum. Byunggon Yang (English Dept., Dongeui Univ., 24 Kayadong Pusanjingu, Pusan 614-714, Korea) and SunMee Kang (Seokyeong Univ., Jungneung-Dong Sungbuk-Ku, Seoul 136-704, Korea)

Some problems in speaker identification procedures were examined: transformation of acoustic parameters into auditory scales, invalid measurement values, and comparability of spectral energy values across the frequency range. To resolve those problems, the acoustic spectral energy of three Korean numbers produced by ten female students from narrow-band spectrograms at 19 proportional time points of each voiced segment were analyzed. Then, cells of the first five spectral matrices were averaged to form a matrix model for each speaker. The correlation coefficients and sum of the absolute amplitude difference in each pair of the spectral models of the ten subjects were obtained. Also, some individual matrix models were compared to those of the same subject or the other subject with a similar spectral model. Results showed that in numbers "2" and "9" subjects could not be clearly distinguished from the others but in number "4" it shed some possibility of setting threshold values for speaker identification if the coefficients and the sum of absolute difference were employed. Further studies would be desirable on various combinations of the range of long-term average spectra and the degree of signal pre-emphasis. [Work supported by grant No. R01-1999-000-00229-0 from the Korea Science & Engineering Foundation.]

NOTE: Separate registration required. See page A24.

MONDAY EVENING, 28 APRIL 2003 RENAISSANCE, TENNESSEE ROOM, 7:00 TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Hearing in Three Dimensions

Beverly A. Wright, Chair

Department of Communication Sciences and Disorders, Northwestern University, 2299 North Campus Drive, Evanston, Illinois 60208-3550

Chair's Introduction—7:00

7:05

1eID1. Hearing in three dimensions. Barbara Shinn-Cunningham (Boston Univ., Boston, MA 02215, shinn@bu.edu)

One of the key functions of hearing is to help us monitor and orient to events in our environment (including those outside the line of sight). The ability to compute the spatial location of a sound source is also important for detecting, identifying, and understanding the content of a sound source, especially in the presence of competing sources from other positions. Determining the spatial location of a sound source poses difficult computational challenges; however, we perform this complex task with proficiency, even in the presence of noise and reverberation. This tutorial will review the acoustic, psychoacoustic, and physiological processes underlying spatial auditory perception. First, the tutorial will examine how the many different features of the acoustic signals reaching a listener's ears provide cues for source direction and distance, both in anechoic and reverberant space. Then we will discuss psychophysical studies of three-dimensional sound localization in different environments and the basic neural mechanisms by which spatial auditory cues are extracted. Finally, ''virtual reality'' approaches for simulating sounds at different directions and distances under headphones will be reviewed. The tutorial will be structured to appeal to a diverse audience with interests in all fields of acoustics and will incorporate concepts from many areas, such as psychological and physiological acoustics, architectural acoustics, and signal processing.