Session 4aAA

Architectural Acoustics and Musical Acoustics: Archeological Acoustics I

David Lubman, Chair
David Lubman & Associates, 14301 Middletown Lane, Westminster, California 92683

Chair's Introduction—8:55

Invited Papers

9:00

4aAA1. Sense and nonsense concerning design and engineering of the theaters of classical antiquity and the writings of M. Vitruvius Pollio on the subject. George C. Izenour (16 Flying Point Rd., Stony Creek, CT 06405)

This paper is a concentrated encapsulation of the theater architecture of classical antiquity utilizing paired slides of theater restorations in plan and perspective section accompanied by selected photographs of archeological sites. All of this is occasioned by four decades of the author's travels, researches, writings, ruminations, and conclusions concerning the theaters of ancient Greece and Rome. Considered in the abstract and irrespective of historical epoch, the functional purpose underlying all theater architecture is to provide the physical means by which an audience, preferably comfortably seated, is enabled to both see and hear a staged performance. It is the purpose of this paper to equate the archeologically revealed ''SENSE'' with the romantically derived fictious ''NONSENSE'' of both the outdoor and the roofed theaters of classical antiquity. A contemporary perspective concerning the ''SENSE'' and the ''NONSENSE'' of the writings of M. Vitruvius Pollio on theater architecture is also provided. O tempora O mores.

9:30

4aAA2. Acoustics of ancient theatrical buildings in China. Ji-qing Wang (Inst. of Acoust., Tongji Univ., Shanghai 200092, PROC)

The performing arts in China could be traced back to a long history and had well developed during the Song and Yuan Dynasties, 11th–14th centuries. A unique form of pavilion stage, opened on three sides and thrust into the audience area, was then the most popular for the open-air theatres, the courtyard theatres, and the indoor theatres. As the traditional Chinese opera, Beijing opera and a variety of local opera, was performed in an abstract sense, no stage sets were used. Therefore, it was possible that the low stage ceiling and back wall of a pavilion stage could function as the reflective shell, which increased the early reflections and also the intensity of the sound to the audience, and meanwhile provided sufficient support to the singer. The measured reverberation times of these halls were around 1.0 s at major frequency bands that were optimal for occupied halls of moderate size. Such a stage house provided the performers a desirable sense that the auditorium was responsive to their effort. Illustrations of acoustical features in different types of stage house will be provided during presentation.

10:00

4aAA3. The Rani Gumpha: A 2nd century B. C. acoustical theatre in India. C. Thomas Ault (Theater Dept., Indiana Univ. of Pennsylvania, Indiana, PA 15705) and Umashankar Manthravadi (New Delhi, India)

The Rani Gumpha is a bi-level theatre, built in ancient Khalinga (now 5 km from Bhubaneswar, Orissa) by King Kharavela circa the 2nd century B.C. It is the first or Vikirsta type of theatre discussed by Bharat Muni in the *Natyashastra*, and is remarkably similar to the Hellenistic theaters of Greece, except that it has a square, not round, ground-plan. It is unique for its resonating chambers on the upper and lower levels which surround the performance space. We shall present a brief history of the site and, using visual and acoustical materials created from our on-site testing, demonstrate the effects of gain, resonance, and duration in the performance and spectator spaces. We shall also address vocal and instrumental range, harmonics, function of specific chambers and their interaction with the other chambers. Please note that this is a work in progress, not a complete or definitive assessment of the site. We anticipate that our colleagues will enhance our research through their shared expertise.

10:30

4aAA4. Legends of echoes linked through acoustics to prehistoric art. Steven Waller (American Rock Art Res. Assn., Deer Valley Rock Art Ctr., P.O. Box 41998, Phoenix, AZ 85080-1998)

A new area for acoustic research has arisen from a previously unsuspected relationship between ancient legends of echoes, and prehistoric rock art. This art includes Ice Age deep cave paintings, and Native American petroglyphs typically found high on canyon walls. This acoustic connection is important because there has been no satisfactory explanation of the motivation for the production of rock art (neither the unusual locations nor the subject matter the artists chose). Yet producing rock art clearly was a major preoccupation for early Homo sapiens sapiens over a span of tens of thousands of years. The concept of sound wave propagation and

reflection is a modern paradigm, and is based on the abstraction of invisible pressure waves being diverted by boundaries between media of different densities. In ancient times, however, the causes for many natural phenomena were explained by personification or animism, including attributing echoes to be the responses of spirits. Could echoes have motivated the ancient artists? Acoustic studies at rock art sites may be starting to answer this important question.

11:00

4aAA5. Mayan acoustics: Of rainbows and resplendent quetzals. David Lubman (David Lubman & Assoc., 14301 Middletown Ln., Westminster, CA 92683)

Progress is reported in understanding physical and mythological elements of the recently reported discovery that handclaps at the Mayan Temple of Kukulkan at Chichen Itza, Mexico produce birdlike descending chirped echoes. Echoes bear strong cognitive resemblance to the sound of the resplendent quetzal, the Mayan sacred bird associated with the temple. ''Picket fence'' (periodic reflections) from temple staircases explain the presence, frequency, and trajectory of descending chirps. But the rich harmonics are better explained by modeling staircases as acoustical analogues of inclined optical Bragg diffraction gratings. The remarkable conversion of handclap into chirped echo is explained through the time-dispersive, pulse-stretching properties of these gratings. Chirped echoes can thus be termed ''acoustical rainbows'' because acoustical energy is selectively dispersed over time, much as optical gratings selectively disperse colors over space. New evidence strengthens mythological arguments for intentional Mayan use of acoustical features. Echoes may have been exploited at solstice ceremonies. Preliminary evidence further suggests substantial sound reinforcement from the top of the temple to the plaza below. This could explain how Mayan kings addressed large crowds. Impressive flutter echoes and whispering galleries found at the nearby Great Ballcourt suggest possible widespread exploitation of acoustical architecture at this Mayan ceremonial site.

THURSDAY MORNING, 4 NOVEMBER 1999 KNOX ROOM, 8:15 TO 11:55 A.M.

Session 4aBB

Biomedical Ultrasound/Bioresponse to Vibration: Therapeutic Applications of Ultrasound

Inder Raj S. Makin, Chair

Ethicon Endo-Surgery, 4545 Creek Road, Cincinnati, Ohio 45242

Invited Papers

8:15

4aBB1. Acoustic hemostasis. Lawrence A. Crum, Michael Bailey, Kirk Beach, Stephen Carter, Wayne Chandler, Peter Kaczkowski, Roy Martin, Pierre Mourad, Sandra Poliachik, Shahram Vaezy (Univ. of Washington, Seattle, WA 98105), George Keilman (Sonic Concepts, Woodinville, WA 98072), Thomas L Anderson, Lee Weng, and David M. Perozek (Therus Corp., Seattle, WA 98121)

When high-intensity focused ultrasound (HIFU) at megahertz frequencies and kilowatt intensities is applied to animal tissue, absorption can result in focal temperatures that exceed 70 °C within a few seconds. These high temperatures result in protein denaturization and coagulative necrosis, with the result that bleeding in the capillary bed and within small vessels is rapidly terminated. Additionally, if HIFU is applied to tears or cuts in large vessels, a combination of acoustic streaming and sound absorption can lead to acoustic hemostasis also in a few seconds. The role of acoustic cavitation in this entire process is not clear. We have examined the application of this phenomenon to trauma care and general intraoperative surgery and find that it has many promising attributes. We shall present a general review of our work in this area as well as our most recent results and their clinical implications. [Work supported in part by DARPA.]

8:35

4aBB2. Influence of beam geometry and tissue boundaries on thermal ultrasonic lesions. Frederic L. Lizzi (Riverside Res. Inst., New York, NY 10036, lizzi@rrinyc.org)

The size and shape of thermal lesions induced by high-intensity focused ultrasound (HIFU) are key factors in treating disease. Studies are being conducted to determine how HIFU beams and exposures can be designed to produce useful lesion attributes in specific target tissues. The studies include radially asymmetric beams and tissues with different acoustic absorption coefficients; sharp thermal gradients at such boundaries can significantly affect lesion geometry. Ocular tumor therapy uses asymmetric beams that produce lesions with ellipsoidal cross sections to facilitate treatments with closely packed lesion matrices. Three-dimensional simulations have analyzed heat patterns and lesions induced by these beams in irregular tumors whose geometry is derived from clinical serial-plane scans using 50-MHz diagnostic ultrasound. The simulations documented difficulties in treating boundary regions adjacent to the waterlike vitreous humor. Lesion results were consistent with *in vitro* and *in vivo* animal experiments. A related study involves an annular-array therapy system with a central rectangular aperture, housing a diagnostic array. The system (Spectrasonics, Inc.) is intended for treatments in various organs, including liver. Since the therapy beams are not radially symmetric, 3-D simulations and initial animal experiments have examined the asymmetry in induced temperature patterns and lesions. These investigations also addressed spurious grating peaks in axial intensity.

8:55

4aBB3. Experimental studies of the use of a split-beam transducer for prostate cancer therapy in comparison to a single-beam transducer. Jimmy S. J. Wu, Narendra T. Sanghvi, Michael Phillips, and M. Kuznetzov (Focus Surgery, Inc., 3940 Pendelton Way, Indianapolis, IN 46226, nsanghvi@focus-surgery.com)

The therapy of prostate cancer by high-intensity focused ultrasound (HIFU) requires the treatment of whole prostate tissue. Single-beam HIFU results in treatment time of about 3 to 4 h. However, the treatment time can be reduced by the use of multiple HIFU beams that are generated simultaneously. We have approached this problem by splitting the main ultrasound beam into one main beam that is surrounded by multiple sidelobes (coined ''split-beam transducer''). In our split-beam transducer by means of geometrical configuration of the transducer, the main beam is surrounded by four sidelobes, producing a focal area about three times larger compared to that of a single beam. The step size between two adjacent ablative exposures can be increased using SBT. Experiments were designed to understand the use of ''split-beam transducer (SBT)'' on the Sonablate 200 device (Focus Surgery Inc.). Test objects, (1) Mylar strip, (2) Plexiglass block, and (3) turkey breast tissue were used. Temperature measurements using multiple thermocouples were made in the tissue near the beam entrance and about 5 mm deep into the samples. Results show that the split beam creates a larger volume of lesion than the single beam in turkey breast tissue. For the single beam used in BPH treatment step size was set at 1.8 mm to create connected necrosis. In the current study a step size of 2.5 mm can be used to create a connected lesion in turkey breast tissue, leading to a 30% reduction in treatment time for the same volume of tissue.

9:15

4aBB4. Self-focusing ultrasound phased arrays for noninvasive surgery. Emad S. Ebbini (Dept. of ECE, Univ. of Minnesota, Minneapolis, MN 55455)

Modern transducer technology is providing phased arrays with high-power, relatively large fractional bandwidth, and low cross coupling between array elements. As a result, it is possible to use these arrays in a dual imaging/therapy mode. We have recently demonstrated this using a 64-element 1-MHz array with 1.5×50 mm^2 elements on a spherical shell with a 100-mm radius of curvature. In the therapy mode, the array was shown to be capable of producing focal intensities in excess of 10 000 W/cm^2 and cause well-defined lesions in tissue at depth. In the imaging mode, the array was shown to be capable of forming images of acceptable quality in a region covering ±35 mm axially and ±25 mm laterally around its geometric center. In this paper, we describe new algorithms for using this array in self-focusing mode based on acoustic feedback from the intended focal region. Two modes of acoustic feedback will be described and compared, specifically, direct hydrophone measurements at the desired focus and backscatter from the focal region (collected in imaging mode). Experimental data are given to demonstrate optimal focusing of phased array systems in the presence of strongly scattering objects, e.g., the ribcage.

9:35

4aBB5. Ultrasonic tissue cutting with silicon surgical tools. Amit Lal (Dept. of Elec. and Computer Eng., Univ. of Wisconsin, Madison, WI 53706)

Ultrasonic cutting is a well accepted way to cut tissues. However the exact nature of the cutting is complicated and not well understood. Cavitation, direct cutting, inertia stiffening, and thermal damage are just some of the effects believed to occur. Furthermore, we have been developing silicon-based ultrasonic surgical tools that hold promise for more functional surgical tools. In this paper we present results on the measurement of forces during ultrasonic cutting. Temperature sensors embedded in the tissue are used to measure any ultrasound-induced heating. We will present force and temperature data measured with actuators with different frequencies and intensities. Tradeoffs between the cutting tip sharpness and ultrasonic tip velocity effecting tissue damage will be presented in the context of Hertzian contact theory. We will use this data to formulate a phenomenological theory of ultrasonic cutting useful to design transducers for ultrasonic cutting. [Work supported by the Whitaker Foundation.]

9:55

4aBB6. *In vivo* ultrasound-induced occlusion of arteries and veins. Kullervo Hynynen and Vincent Colucci (Div. of MRI, Dept. of Radiol., Brigham and Women's Hospital, Harvard Med. School, 221 Longwood Ave., Boston, MA 02115)

In this study rabbit ear vessels were sonicated *in vivo* with focused spherically curved ultrasound transducer (diameter 100 mm, radius of curvature 80 mm, frequency 2 MHz). The sonication duration was varied from 0.2 to 5 s. The acoustic power was between about 50 and 200 acoustic watts. The vessels were visualized under microscope during and after the sonication to follow the blood flow and the vessel diameter. A video camera on the microscope was used to record the observations. The sonication caused blood vessels to contract temporarily at low powers and at high powers the vessels occluded and remained closed over the 2-h follow-up. However, the x-ray angiograms performed by injecting x-ray contrast agent in the ear vessels after the animal was sacrificed showed often that complete occlusion was not achieved. A further increase in the power caused the skin to coagulate and these effects had less effect on the blood flow. The vessels were completely sealed by performing a follow-up sonication at about 34 W for 10 s. This coagulated the tissue around the blood vessels and stopped the flow.

10:15–10:25 Break

10:25

4aBB7. A hemisphere transducer array for transskull ultrasound therapy and surgery. Jie Sun, Greg Clement, and Kullervo Hynynen (Div. of MRI, Dept. of Radiol., Brigham and Women's Hospital, Harvard Med. School, 221 Longwood Ave., Boston, MA 02115)

Recently, it has been shown that it is feasible to generate a focused ultrasound field through an intact human skull for therapeutic purposes. Also it is clear that, in order to minimize the skull heating, a large 2-D array should be adopted to maximize the penetration area on the skull. With this in mind, an unconventional hemisphere transducer array is studied and developed. The radius of the transducer is chosen to be 15 cm in order to accommodate most head sizes and mechanical shifting while the transducer can still be fit into an NMR magnet coil to monitor treatments. The transducer's operating frequency, total number of elements (and the associated element size), and element geometry are determined based on numerical studies. The numerical results are presented for the transskull ultrasound field when the transducer is divided into 228 elements. The transducer is constructed and tested in water bath. The preliminary experimental results using this transducer are also presented.

10:40

4aBB8. On the role of acoustic cavitation in enhancing hyperthermia from high-intensity focused ultrasound. Patrick Edson, R. Glynn Holt, and Ronald A. Roy (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

There are several physical mechanisms that lead to localized heating of tissue and tissue-like media with high-intensity focused ultrasound (HIFU). Experimental results obtained *in vivo* [Hynynen, Ultrasound Med. Biol. **17**, 157 (1991)] and *in vitro* [Edson *et al.*, J. Acoust. Soc. Am. **104**, 1844 (1998)] clearly indicate the existence of an insonation pressure threshold above which cavitation activity can profoundly enhance heating rates at MHz insonation frequencies. The dominant mechanisms through which bubbles facilitate heating have yet to be determined. Candidate mechanisms include viscous dissipation in the flow near the bubble surface, absorption of radiated acoustic waves, attenuation of multiply scattered waves, heat transfer across the bubble wall, etc. Results of numerical simulations designed to investigate the relative impact of these mechanisms on heat deposition from single bubbles and bubbly assemblages are reported. The critical roles of equilibrium bubble size and rectified diffusion are addressed. Our goal is to understand the physics of bubble-mediated heating in order to affect cavitation-enhanced hyperthermia in a reproducible and controllable manner. [Work supported by DARPA.]

10:55

4aBB9. Sonoporation of monolayer cells by diagnostic ultrasound activation of contrast-agent gas bodies. Douglas L. Miller and Jawaid Quddus (Dept. of Radiol., Univ. of Michigan, Ann Arbor, MI 48109-0553, douglm@umich.edu)

Fluorescent dextran (500 kD) uptake by sonoporation was observed after 1-min ultrasound exposure of human (A431 epidermoid carcinoma) and mouse (strain L connective tissue) cell monolayers in the presence of 1% Optison ultrasound contrast agent (Mallinckrodt, Inc.). Ultrasound exposure was provided by a 3.5-MHz array (Acoustic Imaging model 5200B, Dornier Medical Systems) operated in the spectral Doppler mode (5-s pulses, 4.4-kHz PRF). The array was mounted in a 37 °C water bath and aimed upward 7 cm away at a 1-mm-thick chamber with the monolayer grown on the upper 5-m-thick mylar window. For a water path, up to about 10% of the cells in a rectangular 0.37-mm^2 observation area (smaller than the -3 dB beam width) exhibited fluorescent dextran uptake for the high-power setting. For exposure with a 6-cm tissue-mimicking phantom, which approximated a -0.3 dB/cm/MHz derating factor, some fluorescent cells were noted even for low power (0.25 MPa p-, 0.13 MI) increasing to about 4%–8% at high power. These results indicate that activation of contrast agent gas bodies by diagnostic ultrasound can be biologically effective at the cellular level. [Work supported by NIH CA42947.]

11:10

4aBB10. Ultrasound-induced cell lysis and sonoporation enhanced by contrast agents. Mark Ward, Jr. and Junru Wu (Dept. of Phys., Univ. of Vermont, Burlington, VT 05405)

The enhancement of ultrasound-induced cell destruction, lysis, and sonoporation in low cell concentration suspensions (2×10^5/mL) by the presence of contrast agents (gas bubble to cell ratio =3D 230) was demonstrated using cervical cancer cells (HeLa S3) suspensions containing micron-size denatured albumin microspheres filled with air (Albunex =AE) or octafluoropropane (OptisonTM). The suspensions were insonificated by 2-MHz continuous or toneburst ultrasound in the near field. The spatial peak pressure amplitude was 0.2 MPa. The enhancement of cell destruction due to Optison was shown to be much higher than that due to Albunex for similar bubble concentration and ultrasound conditions. For toneburst exposures, significant lysis and sonoporation only occurred in the presence of a contrast agent. The majority of the bioeffects observed occurred in the first 5 min of exposure. The relationship between the enhancement of bioeffects and duty cycle of toneburst ultrasound appears to indicate that both stable gas spheres of contrast agents and cavitation nuclei created by the disruption of the gas spheres play a significant role in causing the bioeffects.

11:25

4aBB11. Ultrasound-induced enhancement of skin permeability to Octa-L-Lysine. Ludwig J. Weimann, Jr. and Junru Wu (Dept. of Phys., Univ. of Vermont, Burlington, VT 05405)

The efficiency of ultrasound in transdermal delivery of high molecular weight protein drugs has been demonstrated recently [S. Mitragotri *et al.*, Science **269**, 850–853 (1995)]. It has been shown in our previous paper [J. Wu *et al.*, Ultrasound Med. Biol. **24**, 1–6 (1998)] that ultrasound-induced structural disorder in human stratum corneum may be responsible for the enhancement of skin permeability. In this presentation, the effect of 20-kHz ultrasound on the permeation of Octa-L-Lysine tagged with Fluorescein (Octa-L-Lysine-4-FITC, effective molecular weight 2.5 kDa) through the skin was shown by optical reflection images and confocal microscopic cross-sectional images of skin. It has been indicated that both sonophoresis and sonoporation may be involved in the procedure. The possible mechanisms and bioeffects introduced by the procedure will also be discussed.

11:40

4aBB12. A study of the effect of ultrasound on mandibular osteodistraction. Tarek H. A. El-Bialy, Thomas J. Royston, Richard L. Magin, and Carlotta A. Evans (Univ. of Illinois at Chicago, Chicago, IL 60607, troyston@uic.edu)

Previous studies have shown that low-frequency ultrasound can enhance bone fracture healing. Other craniofacial research studies have indicated that distraction osteogenesis (bone lengthening) can be performed with some degree of success on the mandible. However, a complication typically reported is the bending of the anterior portions of the osteotomized mandibles which is believed to be caused by the stretching of the circumoral muscles on the newly formed bone callus. It is hypothesized that application of a low-frequency ultrasound treatment to osteotomized mandibles may enhance bone formation and minimize the healing period, thus minimizing the stretching of newly formed callus. Preliminary results are reported here which test this hypothesis on rabbit models. The ultrasound treatment regimen and the acoustic techniques used for assessing healing are discussed.

Session 4aEA

Engineering Acoustics: Acoustic Properties and Characterization of Materials

Stephen E. Forsythe, Chair
Naval Undersea Warfare Center, 1176 Howell Road, Newport, Rhode Island 02841

Contributed Papers

8:15

4aEA1. Analysis of ultrasonic nondestructive evaluation data using singular value decomposition of the Hankel data matrix. Timothy C. Hanshaw, Chin S. Hsu (School of Elec. Eng. and Computer Sci., Washington State Univ., Pullman, WA 99164-2752), and Michael J. Anderson (Univ. of Idaho, Moscow, ID 83844-2752)

A method is presented for processing data from ultrasonic nondestructive evaluation (NDE) tests of material specimens by performing a singular value decomposition of the Hankel data matrix. The singular vectors and singular values of the Hankel data matrix correspond roughly to modal shapes and amplitudes in the ultrasonic waveform. A great deal of information about the specimen being tested can thus be conveyed with relatively little data; one approach which has been explored is to generate a colormap in which the intensity of the RGB components are determined by selected singular values. This approach was applied to simulated test data for both nominal and flawed specimens and the resulting colormap has been compared to a gray scale based on the rms value of the response data. The color map provides more information about the flaw than the gray scale in two ways: the intensity of the color components increases monotonically with increasing flaw severity, and the color map provides information about the flaw location.

8:30

4aEA2. Recursive application of cross-correlation for extraction of sparse impulse trains from ultrasonic nondestructive evaluation data. Timothy C. Hanshaw, Chin S. Hsu (School of Elec. Eng. and Computer Sci., Washington State Univ., Pullman, WA 99164-2752), Jeffrey A. Daniels, and Michael J. Anderson (Univ. of Idaho, Moscow, ID 83844-2752)

The goal of Quantitative Nondestructive Evaluation is to determine numerical values for a specimen's properties. In many cases, especially when the specimen is composed of thin layers, the material properties manifest themselves in a sparse train of impulses. This is the case for many state of the art hybrid materials. Signal processing techniques can be used to extract the desired impulse train from the ultrasonic data. In this paper, a cancellation method is used to recursively subtract a reference wavelet from the acoustic response of a test specimen in order to determine the desired impulse train of the specimen. The method was applied to acoustic data for a thin aluminum sheet. The resulting impulse train displayed enhanced fidelity as compared to the original wave form. Wave travel times and transmission coefficients thus obtained agree with expected values to within 6%.

8:45

4aEA3. Ultrasonic spectroscopy inversion method for the determination of thicknesses and acoustical properties in a thin-layered medium. D. Lévesque, M. Choquet, M. Massabki, C. Néron, and J-P. Monchalin (Industrial Mater. Inst., Natl. Res. Council of Canada, 75 de Mortagne Blvd., Boucherville, QC J4B 6Y4, Canada, daniel.levesque@nrc.ca)

Ultrasonic waves reflected from the front and back surfaces of a thin-layered medium (<2 mm) overlap and interfere in the time domain. A spectroscopy approach has been proposed in the past, collecting and analyzing a large number of echoes together in the frequency domain to measure resonance frequencies and determine properties of the layered medium. With the presence of a fluid between layers, the resonance modes are mainly those of individual layers in the vacuum and an inversion algorithm has been successfully applied to recover some properties. For adhesively bonded layers, the resonance modes of the different layers are strongly coupled and important frequency shifts are observed. In this paper, a characteristic equation which includes an adhesion parameter has been derived and a numerical inversion technique has been used to simultaneously obtain thickness maps of the individual layers from measured resonance frequencies. One application is the laser-ultrasonic detection of hidden corrosion in aircraft lap joints where the paint layer has a significant impact on the resonance frequencies of the paint/metal–skin structure. Results will be presented for samples simulating aircraft lap joints with various levels of metal loss. [Work funded by the U.S. Department of Defense under Contract F33615-98-C-5200.]

9:00

4aEA4. Ultrasonics for process monitoring and control. Leonard Bond, Margaret Greenwood, and Judith Bamberger (Pacific Northwest Natl. Lab., P.O. Box 999, M.S. K5-25, Richland, WA 99352)

Ultrasonic waves are well suited for use in measurements that characterize multiphase fluids and flows. The waves can interrogate fluids and dense optically opaque suspensions. Transducers can give signals that penetrate vessels. Such transducers, together with supporting instrumentation, can provide on-line real-time measurements for use in process monitoring and control. Staff at the Pacific Northwest National Laboratory have developed a family of devices that measure fluid and slurry density, viscosity, flow rheology, particle size distribution, concentration, and velocity profiles. The transducers have been deployed on model systems to provide data that can be integrated to provide a more complete characterization of a process streams, including to detect time-dependent changing interfaces caused by fouling or phase changes, to track process conditions during mixing, sedimentation, stratification, and slurry transport. A summary of the fundamental physical acoustics for wave-process stream interactions will be provided. Examples that illustrate the capabilities of each type of measurement applied to slurry systems at Hanford site and for wood pulp process streams will be presented. The potential for enhanced process monitoring and control will be illustrated using wood pulp data, where it is required to measure the pulp consistency and degree of refining.

9:15

4aEA5. Inexpensive technique for accurate determination of elastic properties by free decay in circular plates. Thomas B. Gabrielson (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804, tbg3@psu.edu)

Rapid determinations of elastic modulus are often made in the lab by measuring the resonances of regular bars of the material. When this is done, the most accurate results are obtained by suspending the specimen with as close to free conditions as possible. A circular plate supported by a taut wire through a central hole behaves accurately as an annular plate with both edges free. For low-loss materials, 10–20 modes are easily detected with an inexpensive microphone when such a plate is struck. Furthermore, the analytical solution for a thick, annular plate can be pro-

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grammed readily in computational tools such as MATHCAD, MATLAB, or MATHEMATICA. (Even for plates traditionally considered thin—thickness-to-diameter ratios less than 0.1—thin-plate theory can produce significant errors for higher-order modes.) The first two resonance frequencies can be used to determine the elastic modulus and Poisson's ratio and the higher modes can be used to assess the uncertainty or to uncover anisotropy. The resonance frequencies can be measured with a fast Fourier transform spectrum analyzer and an understanding of the response of such an analyzer to a slowly decaying sinusoid. [Work supported by the Naval Sea Systems Command.]

9:30

4aEA6. Estimation of model-parameter errors. M. Roman Serbyn (Phys. Dept., Morgan State Univ., 1700 E. Cold Spring Ln., Baltimore, MD 21251)

The linear time-invariant two-port, characterized by four complex parameters, has established itself as a reliable model for a variety of physical systems. It has been studied in two aspects: as the ''forward'' problem (analysis) and the ''inverse'' problem (synthesis). The focus of the research here reported is on the latter, that is, on inferring the values of the model parameters from the results of measurements at the input and output ports. In particular, the goal of this investigation has been to quantify the relationship between uncertainties in the measured values and the resulting errors in the parameter estimates. Two physical systems have been used in the present study: an electrical circuit with known component values and an uncalibrated electromechanical transducer. Mathematically, each device was modeled as a Moebius transformation, $w=(az+b)/(cz+d)$, whose parameters, (a,b,c,d), were estimated from least-squares fits to measured values of the variables (z,w). The *a priori* known electrical circuit provided a base line for checking the validity of the computation algorithms. MAPLE, MATLAB, and SPICE programs were utilized. [Work supported by HUD Special Projects Grant No. B98SPMD0074.]

9:45

4aEA7. Electromechanical material properties for new high-power sonar transduction materials. Elizabeth McLaughlin and James Powers (Code 2132, NAVSEA Newport, 1176 Howell St., Newport, RI 02841)

New high-power electroactive materials such as PMN-PT ceramic and PZN-PT single crystal are being considered for Naval sonar applications. But unlike conventional PZT ceramic, these new materials often have a large mechanical stress and temperature dependence. In this presentation we will describe a method which characterizes the stress and temperature dependence of these new materials under high-drive conditions and present data on several recently measured materials. The measurement generates the quasistatic polarization and strain versus field curves for different values of compressive stress and temperature, and the stress versus strain curve for different values of dc bias field. From these data, the large signal piezoelectric d constant, dielectric constant, short-circuit Youngs modulus, k33 coupling factor, and energy density can be determined. As an example, the large-signal piezoelectric constant for an 80/20/2 (PMN/PT/Lanthanum) material dropped 30% for an increase in temperature of 23 °C and dropped 14% for an increase of 34-MPa prestress. For each of the same conditions the coupling factor dropped 13% and the energy density dropped 22% and 13%, respectively. [Work supported by ONR, SPAWAR, and PEO-USW.]

10:00–10:15 Break

10:15

4aEA8. Sonar dome deflection measurements on the USS Radford. Joel F. Covey, Robert D. Corsaro (Naval Res. Lab., Washington, DC 20375), Diane B. Weaver, and Jonathan B. Walker (SFA, Inc., Largo, MD 20774)

Sonar dome deflection measurements were performed on the new monolithic sonar dome of the USS Radford in Oct.–Dec. 1997. Over 700 Mbytes of raw data from 22 environmental sensors and 128 displacement sensors were obtained. The at-sea data acquisition was highly successful with data collected over sea states 1 through 4, and for numerous ship speeds up to 31 knots. Analysis of this data has provided a more complete characterization of previously measured sonar dome deformations. Many new conclusions and observations on sonar dome behavior have been observed and correlated with ship motions.

10:30

4aEA9. A calculation of the flow resistivity in intermediate flow regime. M. A. Picard Lopez, P. E. Solana Quiros, and J. V. Arizo Serrulla (Dept. de Fisica Aplicada, Univ. Politecnica de Valencia, Camino de Vera, 14, 46022 Valencia, Spain)

Many of the acoustic properties of fibrous materials can be modeled and predicted with the aid of appropriate empirical formulas. Among the input parameters that are required by such types of macroscopic models, the flow resistivity is a very well-established quantity in the literature, as usually this parameter is measured in steady Poiseuille flow conditions, i.e., at comparatively low flow speeds. One major concern with respect to such a parameter is the question of its possible frequency dependence. From a theoretical point of view, we have examined to which extent the capilar pore approximation can be utilized in intermediate flow regimes. The tendency towards lower values of flow resistivity for increasing acoustic Reynolds number also appears in all samples studied. And only the greater density samples, for the tested frequencies, were found in the flow regime of Poiseuille. However, if we want to obtain the value of the flow resistivity, for any sample, the employed experimental procedure makes it possible by a relationship between the experimental values and those of the Biot function. The value of the flow resistivity in different dynamical flow regimes to that of Poiseuille can be obtained.

10:45

4aEA10. Normal incidence sound absorption coefficient measurement using sound intensity technique: An accuracy investigation. Mohamad N. Dimon, Ahmad K. Said (Elec. Eng. Faculty, UTM, 81310, Skudai, Johor, Malaysia), Md. N. Ibrahim (UTM, 81310, Skudai, Johor, Malaysia), and Md. Y. Jaafar (MARDI, Serdang, Selangor, Malaysia)

Sound absorption data are required in specifying and proposing the correct materials for room acoustic treatment. However, there are instances where noncommon materials such as direct piercing carved wood panels (dpcwps) are used without sound absorption data. Sound absorption measurements obtained using an impedance tube and reverberation chamber are not accurate for dpcwp. This is primarily because the wall backing the dpcwp during the measurements constitutes the carved surface, which does not exist in actual installation. Therefore, an alternative measurement technique using a sound intensity measurement technique is more appropriate for dpcwp. This paper focuses on experimental methodology where various aspects need to be taken into consideration to achieve accurate results. Various criteria affecting measurement instruments' capabilities and other factors affecting sound intensity measurement accuracy will be discussed. Sound absorption measurement was conducted on three different perforation panels which resemble dpcwp and it was found that the sound absorption measured with a deviation from an average value of −5.59% to 5.95% can be achieved, and is particularly accurate.

11:00

4aEA11. Rayleigh waves in anisotropic porous media. Alexander Kaptsov and Sergey Kuznetsov (Inst. for Problems in Mech., Prospect Vernadskogo, 101a, Moscow, 117526 Russia)

Rayleigh waves represent waves propagating in a half-space and attenuating exponentially with depth. During the last 30 years considerable progress has been achieved in developing both analytical and numerical methods for analysis of Rayleigh waves propagating in homogeneous media with arbitrary elastic anisotropy. The propagation of Rayleigh waves in heterogeneous media has not been studied nearly as often. Very few works are devoted to the analysis of speed and attenuation of Rayleigh waves in media (mainly isotropic) with uniformly distributed pores and/or

microcracks. The advantage of applying Rayleigh waves to the nondestructive determination of concentration and the preferred orientation of pores lies in its high sensitivity to variation of the material elastic parameters due to preexisting pores. The developed approach for analysis of Rayleigh waves in porous or cracky anisotropic media is based on a combination of six- and three-dimensional complex formalism and the two-scale asymptotic analysis. In its turn, the latter utilizes a newly developed spatially periodic boundary integral equation method. This is used for determination of the effective characteristics of heterogeneous media containing isolated uniformly distributed pores. Numerical data are discussed.

THURSDAY MORNING, 4 NOVEMBER 1999 HAYES ROOM, 10:00 A.M. TO 12:30 P.M.

Session 4aED

Education in Acoustics: Hands-On Demonstrations

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

Chair's Introduction—10:00

Approximately 20 experiments designed to introduce various acoustics principles will be set up in the room. After a brief introduction of each demonstration they will be available for experimentation by visiting high school students. Space permitting, conference participants are also invited to view the demonstrations and perform the hands-on experiments, provided such activity does not interfere with the purpose of the session, to introduce acoustics experiments to high school students. Experiments performed by the students will include many, ranging in sophistication from simple wave studies to mapping of normal modes in two-dimensional structures.

THURSDAY MORNING, 4 NOVEMBER 1999 REGENCY BALLROOM SOUTH, 9:25 TO 10:30 A.M.

Session 4aID

Interdisciplinary: Distinguished Lecture on Smart Structures and Microelectromechanical Systems (MEMS)

Jerry H. Ginsberg, Chair
School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, Georgia 30332

Chair's Introduction—9:25

Invited Papers

9:30

4aID1. Smart structures and microelectromechanical systems (MEMS). B. T. Khuri-Yakub (E. L. Ginzton Lab., Rm. 11, Stanford Univ., Stanford, CA 94305-4085)

Capacitor transducers have been around for as long as piezoelectric transducers. They have not presented much competition for piezoelectric transducers as transmitters and receivers because the method of their manufacture did not optimize and highlight their performance. With the advent of silicon micromachining, it is now possible to make capacitors with very thin gaps that sustain electric fields of the order of 109 V/m or more. At these levels of electric field, the transformer coupling between the electrical and mechanical parts of the capacitor transducer, and thus its performance, become comparable to that of piezoelectric transducers. Advantages such as practically infinite bandwidth, ease of manufacture, and the ability to integrate electronic circuitry make this type of transducer a very important candidate for many ultrasonic applications. We will review the design and performance of capacitor transducers for both immersion and airborne ultrasound applications. Single-element, 1-D, and 2-D arrays of transducers will be presented along with imaging results. The overall dynamic range of systems with these transducers will be shown to be over 140 dB/V/vHz. We will also present a design for making surface-wave and Lamb-wave transducers using the same capacitor concept. [This work was sponsored by the Office of Naval Research, the Defense Advanced Research Projects Agency, and the Air Force Office of Scientific Research.]

Session 4aMU

Musical Acoustics: Music, Rhythm and Development I

Caroline Palmer, Cochair
Department of Psychology, The Ohio State University, 1885 Neil Avenue, Columbus, Ohio 43210

Mari Riess Jones, Cochair
Department of Psychology, The Ohio State University, 1885 Neil Avenue, Columbus, Ohio 43210

Invited Papers

9:00

4aMU1. Music, rhythm, and development: Chairs' introduction. Caroline Palmer and Mari Riess Jones (Dept. of Psych., Ohio State Univ., 1885 Neil Ave., Columbus, OH 43210, palmer.1@osu.edu)

This special session is sponsored by the Caroline B. Monahan Fund and by the Center for Cognitive Science at Ohio State University. Caroline Monahan's contributions to the field of music cognition focus on auditory perception and memory, including topics of pitch and temporal parallels in music and speech perception, recognition memory for rhythmic and pitch patterns, and the role of timbral and loudness variables in discrimination of auditory patterns. Dr. Monahan's work has been concerned with applications of these topics in speech as well as in music, including language acquisition, hearing impairment, and second-language learning. Dr. Monahan received her Ph.D. at UCLA in 1984 in Psychology, and served as a Research Scientist at the Central Institute for the Deaf. She also served as a Research Associate in the Department of Communication Disorders at the University of Oklahoma. We acknowledge her contributions to and generous support of the field of music cognition.

9:05

4aMU2. Memory and the experience of time. W. Jay Dowling (Prog. in Cognit. Sci., Univ. of Texas at Dallas, Richardson, TX 75083-0688)

Researchers reflecting on the perception of music and speech, from William James a century ago to Eric Clarke very recently (and many in between), have been driven to the conclusion that we do not perceive auditory events instant by instant as they are received by the ear. Rather, we perceive the contents of our working memory where ''sound bites'' of the order of 5 s in length can be heard. This picture is complicated by the consideration that the contents of working memory are themselves in a state of flux over time. Recent research is reported showing ways in which memory for novel melodies changes systematically over periods of 5 min following presentation. Implications for the understanding of music and the experience of time are drawn.

9:30

4aMU3. Issues in music, rhythm, and development. Mari Riess Jones (Dept. of Psych., Townshend Hall, Ohio State Univ., Columbus, OH 43210, Jones.80@osu.edu)

Some general issues pertaining to the response of infants and children to auditory patterns will be discussed. One issue concerns the tempo of simple auditory sequences used as stimuli in perception and production experiments with subjects of different ages. Is there any evidence that children exhibit preferences for certain tempi? If so, do these preferences change with age? What implications can we draw from such findings? A second issue concerns children's sensitivity to relative time structure, including hierarchical time relationships. To what extent are children of different ages responsive to higher-order time relationships, e.g., as in metrical time relationships? A third issue also concerns relative timing. Are young children and/or infants preferentially sensitive to certain kinds of rhythmic relationships? If so, what might these be and what does this imply for understanding the development of rhythmic competencies? This talk will selectively review research addressed to these topics. The aim is to suggest a few general themes that are emerging in the field.

9:55

4aMU4. Children tap faster and within a smaller tempo window than adults. Carolyn Drake (Laboratoire de Psych. Exp., 28 rue Serpente, 75006, Paris, France)

Dynamic Attending Theory [Jones (1976)] proposes that, when listening to auditory sequences, listeners adapt their internal rhythms to those in the sequence (attunement), spontaneously focusing on events occurring at intermediate rates (referent level), and shifting attention to events occurring at faster or slower hierarchical levels (focal attending). The development of these abilities with age and musical training is examined by comparing motor tapping in 4-, 6-, 8-, and 10-year-old children and adults, with or without musical training. Seven motor tapping tasks (including spontaneous motor tempo and synchronization with simple sequences and music) revealed three changes with increased age and musical training: (1) a slowing of mean tapping rate (a reflection of referent period) and mean synchronization rate (a reflection of referent level); (2) an increase in range of tapping rates toward slower rates

(improved focal attending); and (3) a greater ability to adapt the taps to sequence structure, in particular incorporating a greater use of the hierarchical structure (improved attunement and focal attending). Interpreted in terms of attentional oscillators, these results suggest that increased interaction and experience with a particular type of sequence facilitate the passage from the initial use of a single oscillator toward the coupling of multiple oscillators.

10:20–10:30 Break

10:30

4aMU5. Mothers' speech and song for infants. Sandra E. Trehub and Tonya R. Bergeson (Univ. of Toronto at Mississauga, Mississauga, ON L5L 1C6, Canada)

The speech and singing interactions of mothers with their 4- to 7-month-old infants were recorded on two occasions separated by approximately one week. Repetitions of the same songs across sessions had similar tempo (mean difference of 3.73 beats per min) and pitch level (mean difference of 1.07 semitones). By contrast, repetitions of stereotyped verbal phrases had considerably greater differences in tempo (22.82 beats per min) and pitch level (4.96 semitones). Nevertheless, utterance repetitions were perceived as similar to the original because they retained its rhythmic structure. The characteristic focus of researchers on the pitch patterning of maternal speech may be obscuring important consistencies in the spoken rhythms of caregivers. Moreover, the demonstrable sensitivity of young infants to the rhythmic structure of auditory sequences makes it essential to document temporal regularities in the spoken and sung messages that are typically directed to infants. Rhythmic aspects of maternal speech may well emerge as critical components of the infant-directed speech register. [Work supported by the Social Sciences and Humanities Research Council of Canada.]

10:55

4aMU6. Recognition of prosodic cues in music performance. Caroline Palmer, Melissa K. Blakeslee (Psych. Dept., Ohio State Univ., 1885 Neil Ave., Columbus, OH 43210, palmer.1@osu.edu), and Peter W. Jusczyk (Johns Hopkins Univ., Baltimore, MD 21218)

As early as infancy, humans are sensitive to prosodic cues that can aid recognition of words embedded in sentence contexts. We addressed the question of whether similar cues formed by expressive nuances in music performances aid listeners in recognition of musical phrases embedded in melodic contexts. Utilizing a task similar to infant-research habituation paradigms, we report experiments with musically experienced and inexperienced listeners who are familiarized with performances of musical phrases that were identical in pitch/duration contents but differed in their intensity and articulation cues. Listeners then completed a recognition task for performances of the same musical excerpt whose cues either matched or did not match the performances at familiarization, and whose cues were either consistent or inconsistent with the rhythmic context in which the excerpts were embedded. Findings show that listeners can distinguish musical phrases that differ only in expressive nuances, and a mismatch of expressive nuances to the rhythmic context can facilitate recognition. These findings suggest that the prosodic cues that differentiate human performances are part of listeners' memory for melodies, and similar acoustic features may enable the recognition of auditory events in music as in speech. [Work supported by NIMH.]

11:20

4aMU7. Music cognition and Williams Syndrome. Daniel Levitin (Stanford Univ., Stanford, CA 94305) and Ursula Bellugi (Salk Inst. for Biological Studies, La Jolla, CA)

New studies of the musical abilities of individuals with Williams Syndrome (WMS) are presented. WMS is associated with poor spatial, quantitative, and reasoning abilities, coupled with excellent face processing and relatively preserved language abilities. WMS individuals tend to have richer (larger and more colorful) vocabularies than would be expected for their mental age and tend to be more emotionally expressive than normals. There have long been anecdotal reports of increased musicality in WMS (many of them are competent musicians who play from memory) and these are supported by the finding that the planum temporale (PT, a region adjacent to A1) is proportionately enlarged in WMS; in addition, the PT is unusually asymmetric in WMS, with the left side enlarged in a similar fashion to that found in professional musicians. The authors report new studies of rhythm production and memory for music in WMS. The results suggest that this domain of cognitive ability may also be relatively preserved. These findings provide additional evidence that music constitutes a domain-specific higher-level system of expertise in humans, perhaps independent of other mental abilities.

11:45–12:00

Panel Discussion

Session 4aNS

Noise: Progress Report and Discussion on the Continuing Activity of ASA's Role in Noise and Its Control

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

David Lubman, Cochair
David Lubman & Associates, 14301 Middletown Lane, Westminster, California 92683

The Technical Committees on Noise and Architectural Acoustics are holding a meeting to review current progress and invite further discussion on activity to increase the role of the ASA in noise, noise control and related architectural acoustics issues. This outreach effort has included activity on increasing public awareness about noise and noise control, on public hearing screening testing and development of self-testing techniques, on seminars on industrial noise, on meeting room acoustic environments and, currently of special interest, on classroom acoustics. Discussions about these and/or new related activity, including encouraging joint activity in these areas with other professional organizations, will be encouraged. Attendees will be particularly encouraged to discuss elements of a draft standard on classroom acoustics that is currently being developed.

THURSDAY MORNING, 4 NOVEMBER 1999 MCKINLEY ROOM, 10:45 TO 11:50 A.M.

Session 4aPA

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

Henry E. Bass, Chair
National Center for Physical Acoustics, University of Mississippi, University, Mississippi 38677

Chair's Introduction—10:45

Invited Papers

4aPA1. The history of physical acoustics: The worldwide scene. Robert T. Beyer (Dept. of Phys., Brown Univ., Providence, RI 02912)

The term physical acoustics, used by von Helmholtz in his book in 1862 to distinguish the subject from physiological acoustics and music, did not become popular until after World War II. The term covers the production, transmission, and reception of sound, as well as the interaction of sound with matter. With such a broad definition, we shall be able here to delineate only a few major threads. The production of sound moved in the early nineteenth century from musical instruments, explosions, and the human voice, to the ingenious devices of Helmholtz, into the field of ultrasonics through the work of the Curie brothers, and into underwater sound from the researches of Boyle and Langevin. The attenuation of sound includes the theoretical work of Stokes and Kirchhoff, the experiments of Neklepaev, the further theory of Einstein, and the experiments of Kneser. Nonlinear acoustics began with the theories of Poisson and Riemann, was pushed along by Rayleigh, and developed in this century by Fubini, Lighthill, and Khoklov. The optical effect, discovered by Debye and, independently, by Lucas and Biquard, received its theoretical basis in the work of Raman and Nath, and was confirmed in detail experimentally by Nomoto.

4aPA2. Selected topics in the history of atmospheric acoustics in North America, 1865–1940. David T. Blackstock (Mech. Eng. Dept. and Appl. Res. Labs., Univ. of Texas, Austin, TX 78712-1063)

Considered here are three topics in atmospheric acoustics, studied by North American scientists during the 75-year span between the Civil War and World War II. First, in 1865 Joseph Henry (1799–1878), as a member of the Light-House Board, began to investigate fog signaling. His experiments, carried on in friendly competition with John Tyndall in England, added a great deal to our practical knowledge of atmospheric refraction. Second, the first-ever experiments to measure the absorption of sound in the atmosphere were carried out by A. Wilbur Duff (1864–1951) in New Brunswick, Canada, in 1898 and in 1900. The failure of viscosity and heat conduction to account for the measured absorption prompted Lord Rayleigh to postulate that a relaxation mechanism in the air might be responsible for the difference. Finally, George Washington Pierce (1872–1956), who is perhaps best known for his work on piezoelectric and magnetostrictive devices, used an acoustic interferometer to measure dispersion in gases, particularly carbon dioxide. He also measured sound transmission over reflective surfaces. His results confirm the at-first surprising result that at grazing incidence even a very hard boundary seems to act as a pressure release surface.

THURSDAY MORNING, 4 NOVEMBER 1999 MARION ROOM, 9:00 TO 11:15 A.M.

Session 4aPP

Psychological and Physiological Acoustics: Binaural and Sound Field

Raymond H. Dye, Chair

Parmly Hearing Institute, Loyola University, 6525 North Sheridan Road, Chicago, Illinois 60626

Contributed Papers

9:00

4aPP1. Identification and localization of sound sources in the median sagittal plane. Brad Rakerd, William M. Hartmann, and Timothy L. McCaskey (Michigan State Univ., East Lansing, MI 48824)

The ability of human listeners to identify broadband noises having different spectral structures was studied for multiple sound-source locations in the median sagittal plane. The purpose of the study was to understand how sound identification is affected by spectral variations caused by directionally dependent head-related transfer functions. It was found that listeners could accurately identify noises with different spectral peaks and valleys when the source location was fixed. Listeners could also identify noises when the source location was roved in the median sagittal plane when the relevant spectral features were at low frequency. Listeners failed to identify noises with roved location when the spectral structure was at high frequency, presumably because the spectral structure was confused with the spectral variations caused by different locations. Parallel experiments on source localization showed that listeners can localize noises that they cannot identify. The combination of identification and localization experiments leads to the conclusion that listeners cannot compensate for directionally dependent filtering by their own heads when they try to identify sounds. [Work supported by the NIDCD.]

9:15

4aPP2. Contribution of spectra of input signals to two ears to sound localization in the sagittal plane. Kazuhiro Iida (AVC Res. Lab., Matsushita Communication Ind. Co. Ltd., 600 Saedo, Tsuzuki, Yokohama, 224-8539 Japan, kiida@adl.mci.mei.co.jp), Eigo Rin, Yasuko Kuroki, and Masayuki Morimoto (Kobe Univ., Nada, Kobe, 657-8501 Japan)

The previous studies show that the amplitude spectra of input signals to two ears contribute to sound localization in the sagittal plane [e.g., M. B. Gardner, J. Acoust. Soc. Am. **54**, 1489–1495 (1973)]. Furthermore, it is known that the ear on the source side has more influence than the ear on the opposite side on localization [M. Morimoto, Dissertation, Univ. of Tokyo (1982); R. A. Humanski and R. A. Butler, J. Acoust. Soc. Am. **83**, 2300–2310 (1988)]. It is, however, not clarified how each ear spectrum contributes to localization. The present paper builds up some hypotheses on this issue and examines them for their validity by some psychoacoustical experiments.

9:30

4aPP3. The influence of later arriving sounds on the ability of listeners to judge the lateral position of a source. Raymond H. Dye, Jr. (Parmly Hearing Inst. and Dept. of Psych., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626)

This investigation focuses on the effect that later sounds have on the ability of humans to report the spatial location of earlier sounds. Two dichotic pulses were presented (via headphones), separated by an echo delay between 4 and 64 ms. Listeners were asked to judge whether the first click appeared to the left or right of the intracranial midline. The interaural delay of each pulse was independently selected from a Gaussian distribution (μ, $\sigma=0$, 100 μs). The level of the echo relative to the source was 0, -6, -12, -18, -24, -30, or -36 dB. The effect of the echo was determined by measuring proportion correct and by deriving a normalized source weight. For echo delays of 16, 32, and 64 ms, the source and echo click were weighted equally when presented at the same level, and weights barely changed until the echo was attenuated by 18 dB. As the level of the second click was further reduced, the source weight approached 1.0 and the percentage correct approached that obtained in the ''no echo'' condition. For shorter echo delays, source weight and proportion correct increased more quickly as the echo was attenuated. [Work supported by NIDCD & AFOSR.]

9:45

4aPP4. Low-frequency ILD elevation cues. V. Ralph Algazi, Carlos Avendano (CIPIC, UC Davis, Davis, CA 95616, algazi@ece.ucdavis.edu), and Richard O. Duda (San Jose State Univ., San Jose, CA 95192)

It is well known that the binaural ITD (interaural time difference) and ILD (interaural level difference) are the primary cues for azimuth, while monaural spectral features due to pinna diffraction are the primary cues for elevation. Pinna cues appear above 3 kHz, where the wavelength becomes comparable to pinna size. However, it is shown that there are also important low-frequency ILD elevation cues primarily due to torso diffraction. In the experiments reported, random noise bursts were filtered by individualized head-related transfer functions, and four subjects were asked to report the elevation angle. Eight conditions were tested, depending on whether the source was in front or in back, in the median plane or on a 45-deg cone of confusion, and had wide bandwidth or was band limited to 3 kHz. For the band-limited signal, localization accuracy was at chance level in the median plane, and was poor in front. However, at 45-deg azimuth in the back, the accuracy was close to that for a wideband source, the average correlation coefficient being approximately 0.75 for the narrow-band source and 0.85 for the wideband source. A physical explanation for the cues is presented. [Work supported by NSF under Grant No. IRI-9619339.]

10:00

4aPP5. The effect of frequency modulation of the ability to judge dynamic changes in interaural level differences. William M. Whitmer and Raymond H. Dye, Jr. (Parmly Hearing Inst., Loyola Univ., Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626, wwhitme@luc.edu)

The cues for apparent auditory motion include dynamic interaural temporal differences, level changes, and frequency modulation (FM) [Rosenblum, Perception **16**, 175–186 (1987)]. The manner in which these cues interact was examined in a task in which listeners judged the point at which a stimulus having dynamic changes in interaural level difference (ILD) passed through the intracranial midline. In base-line conditions, stimuli were 2.0–2.5-s, 0.5- or 2-kHz sinusoids presented through headphones. In experimental conditions, a frequency-modulated tone was added. Frequency was modulated by a linear ramp or according to a Doppler-based algorithm. The midpoint of the frequency-sweep occurred either at the instant that the ILD passed through 0 dB, acting as a task cue, or at some other point. FM rate and magnitude, temporal position of frequency-sweep midpoint, and direction of apparent movement were randomized across trials. Results indicated that the presence of FM had a significant effect on the perception of intracranial midline. This effect was obtained across FM parameter values. Results were considered in terms of how dynamic cues for perceived motion perceptually interact. [Work supported by NIH and AFOSR.]

10:15

4aPP6. Does precedence prevail through sudden changes in selected partials of a complex natural spectrum? Miriam N. Valenzuela and Ervin R. Hafter (Dept. of Psychol., Univ. of California, Berkeley, CA 94720, miriam@ear.berkeley.edu)

Localization of a source followed by a delayed version of it from another direction is dominated by the direction of the first wavefront. It has been suggested that this ''precedence'' effect and its implication of echo suppression is a dynamic process, subject to listeners' expectations about what a ''plausible'' echo should be. The present study investigates what makes an echo ''plausible'' [Rakerd and Hartmann, J. Acoust. Soc. Am. **78**, 524–533 (1985); Clifton *et al.*, J. Acoust. Soc. Am. **95**, 1525–1533 (1994)]. Synthetic piano tones presented through pairs of speakers in an anechoic room were designed to simulate natural sounds heard in echoic environments. Trials consisted of conditioning stimuli followed by a test stimulus. Each stimulus was made up of a hypothetical source and echo. The spectra of the test echoes could be changed to reflect abrupt changes in the absorption qualities of simulated reflections. As in precedence, conditioning stimuli were localized in the direction of their sources. Attenuation of selected partials in the test-echo did not break down precedence, but when the echo-spectra were modified in less plausible ways by amplification of selected partials, precedence broke down.

10:30

4aPP7. The relative contribution of onset asynchrony, harmonic ratios and angular separation of sound sources to the cross-spectral grouping of complex tones in a free field. Martine Turgeon and Albert S. Bregman (Dept. of Psych., McGill Univ., Montreal, QC H3A 1B1, Canada, martine@hebb.psych.mcgill.ca)

The Rhythmic Masking Release paradigm (RMR) was used to study how onset asynchrony, harmonic ratios and speakers separation interact toward the cross-spectral grouping of concurrent complex tones. In RMR, a regular sequence is perceptually masked by some irregularly spaced masking sounds, which are acoustically identical to those of the rhythm. The identification of the rhythm becomes contingent upon the perceptual grouping of the masking sounds with some flanking sounds of different frequencies. The rhythm-detection accuracy in a 2-AFC procedure was used to estimate the degree of perceptual grouping of 48-ms long masking and flanking tones, each composed of four harmonics. For each of 18 listeners, synchrony of onset and offset between the masking and flanking tones yielded an almost perfect rhythm-detection performance, independent of whether or not they shared a common fundamental frequency and of the spatial separation of their sources. Rhythm-detection accuracy strongly diminished with onset asynchrony and weakly diminished with inharmonicity and speakers separation. Our results suggest that onset asynchrony is the determining factor for the perception of concurrent sounds of a short duration as separate events; inharmonicity and the spatial separation of sources are weak factors, although they appear to reinforce each other toward perceptual segregation.

10:45

4aPP8. Computational auditory scene analysis-constrained array processing for sound source separation. Laura A. Drake (Elec. & Computer Eng. Dept., Northwestern Univ., Evanston, IL 60208-3118), Janet C. Rutledge (Univ. of Maryland at Baltimore, Baltimore, MD 21201), and Aggelos Katsaggelos (Northwestern Univ., Evanston, IL 60208-3118)

In this work, techniques are developed and studied for the extraction of single-source acoustic signals out of multi-source signals. Such extracted signals can be used in a variety of applications including: automatic speech recognition, teleconferencing, and robot auditory systems. Most previous approaches fall into two categories: computational auditory scene analysis (CASA) and array signal processing. The approach taken here is to combine these complementary techniques into an integrated one: CASA-constrained array processing. In principle, this integrated approach should provide a performance gain since the information used by array processing (direction of propagation through a sound-field) is independent of other CASA features (fundamental frequency, on/offset, etc.). One difficulty encountered by CASA that can be overcome by array processing is the sequential grouping of spectrally dissimilar phonemes in a speech signal, such as a fricative followed by a vowel. The method presented here differs from standard array processing by the addition of CASA features for the signal separation decision. Compared to other CASA systems that use binaural cues, it: (1) is not limited to two microphones (since the goal is not auditory system modeling); and (2) makes complete use of source location and other CASA features—for simultaneous and sequential grouping.

11:00

4aPP9. Reducing hearing protector test time with a minimum audible pressure to field transfer function. William J. Murphy and John R. Franks (Bioacoust. and Occupational Vib. Section, NIOSH, MS C-27, 4676 Columbia Pkwy., Cincinnati, OH 45226-1998)

Effective hearing-loss prevention programs should train workers to properly fit hearing protection devices. In order to minimize the training time for protector fitting, unoccluded minimum audible field threshold (MAF) for narrow-band noise could be estimated from the worker's annual minimum audible pressure (MAP) audiogram. Murphy *et al.* previously reported a transfer function for 5- and 1-dB audiometry for 75 and 10 subjects, respectively [W. J. Murphy, J. R. Franks, S. L. Hall, and E. F. Krieg, J. Acoust. Soc. Am. **101**, 3126 (1997)]. Significant differences at several frequencies between the 1- and 5-dB groups prompted further study of the transfer function. This paper reports MAF and MAP results for 44 subjects tested with both 1- and 5-dB audiometry and 1-dB sound field. Pure-tone thresholds were significantly different, for 1- and 5-dB step sizes. The binaural reference equivalent threshold sound pressure level (RETSPL) sound-field thresholds were compared with the RETSPL thresholds for monaural listening in a sound-field listed in ANSI S3.6-1996, Table 9. The binaural thresholds were found to be significantly higher ($p<0.05$) than the standard values at all frequencies except 8000 Hz.

Session 4aSCa

Speech Communication: Physiology and Modeling of Voice

Jody E. Kreiman, Chair
Bureau of Glottal Affairs, Head and Neck Surgery, UCLA School of Medicine, Los Angeles, California 90095

Contributed Papers

8:00

4aSCa1. Measurement of vocal fold depth and thickness in frozen and thawed canine larynges. Niro Tayama (Natl. Ctr. for Voice and Speech, Dept. of Speech Pathol. and Audiol., The Univ. of Iowa and Dept. of Otolaryngol., Univ. of Tokyo, Tokyo, Japan, ntayama@dolce.shc.uiowa.edu), Roger Chan (The Univ. of Iowa), Kimitaka Kaga (Univ. of Tokyo, Tokyo, Japan), and Ingo Titze (The Univ. of Iowa, Iowa City, IA 52242)

Anatomical data on vocal fold dimensions are necessary for defining the vocal fold boundaries in biomechanical modeling of vocal fold vibration. In the mid-membranous coronal section, vocal fold depth can be defined as the distance from the vocal fold medial surface to the thyroid cartilage, whereas thickness can be defined as the distance from the inferior border of the thyroarytenoid muscle to the vocal fold superior surface. Unfortunately, reliable geometric data from histological sections can be obtained only if the effects of sample preparation are quantified. For instance, tissue deformations are often induced by fixation and dehydration, sometimes producing shrinkages around 30%. In this study, reliable geometric data of the canine vocal fold were obtained by comparing frozen and thawed larynges. Coronal sections of frozen larynges were thawed gradually in saline solution. Images of the mid-membranous coronal sections at various thawing stages were captured by a digital camera. Measurements of vocal fold depth and thickness were made using a graphic software package (NIH image). Results showed that geometric changes of the vocal fold induced by freezing were likely reversed by thawing, such that the vocal fold depth and thickness measured on thawed larynges were representative of the pre-freezing state. [Work supported by NIH Grant No. P60 DC00976.]

8:15

4aSCa2. Vocal rise time and perception of a hard glottal attack. Rahul Shrivastav (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405)

Three modes of voice onset are commonly recognized—breathy, normal and hard [Moore (1938)]. These differ in terms of the pulmonary airflow and the degree of medial compression force associated with vocal fold adduction [Koike *et al.* (1967)]. Perceptually, this is characterized by a ''transient rise and fall in pitch and loudness at the onset of voicing'' [Orlikoff and Kahane (1996)]. Perceptual identification of hard glottal attacks (HGA) is an important part of assessment and management of voice disorders. The relation between frequency and intensity rise times and perception of HGA was studied using a continuum of synthesized monosyllabic words varying in the frequency rise times. These tokens were recorded from a normal talker, modified and re-synthesized using STRAIGHT [Kawahara (1997)]. Five listeners were asked to rate the perceived strength of the HGA on a five-point scale. Preliminary results with tokens varying in frequency rise time suggest an increase in the perceived strength of HGA when the frequency rise time increases from 20 ms to 50 ms. The perceived strength of HGA appears to plateau beyond rise times of 50 ms.

8:30

4aSCa3. Pressure profiles on the walls of the glottis for oblique glottal ducts. Ronald C. Scherer (Dept. of Commun. Disord., Bowling Green State Univ., Bowling Green, OH 43403) and Daoud Shinwari (Univ. of Toledo, Toledo, OH 43607)

Oblique angles of the glottis occur during both normal and abnormal phonation. This study examined pressures on the glottal walls for a variety of oblique angle conditions. A Plexiglas model of the larynx was used to obtain wall pressures at 14 taps along the glottal surfaces. The model is 7.5 times larger than real life. A minimal diameter of 0.04 cm and duct axial length of 0.3 cm were used for all cases. Results indicated, for example, that for an included angle of 10 deg (divergence) and a transglottal pressure of 5-cm H_2O, an oblique glottal angle of 15 deg compared to the symmetric glottis: (1) displaced the minimal wall pressure location on the divergent side a short distance downstream and on the convergent side to mid-glottis; (2) increased the pressure drop on the divergent side by about 15% and decreased the pressure drop on the convergent side by about 13%; and (3) broadened the minimum pressure dip on both the divergent and convergent sides. These results suggest paradoxical (opposite) driving pressures on the inferior glottal walls, which would enhance the nontypical phase differences between the two vocal folds. [Research support: NIH grant 1R01DC03577.]

8:45

4aSCa4. Estimation of minimum glottal flow using optimal, low-pass filtering. Yingyong Qi (P.O. Box 210071, Tucson, AZ 85721, yqi@u.arizona.edu) and Robert E. Hillman (243 Charles St., Boston, MA 02114)

The amount of minimum glottal flow in each period of a sustained phonantion is an important parameter in voice research and clinic. In many cases, it is highly desireable to be able to measure the minimum glottal flow automatically. Here, we present a method for estimating the minimum glottal flow using an optimal, low-pass filter. The cutoff frequency of the low-pass filter is determined so that the sum of the variance within each ''closed'' phase of a recorded flow signal and the difference between the recorded and filtered flow signals is minimal. The minimum glottal flow is derived from this optimally, low-pass-filtered signal. This simple optimization procedure results in a complete automatic estimation of minimum glottal flow. Experiments using synthetic and real flow signals indicated that the method is highly accurate and robust.

9:00

4aSCa5. Modeling tremulous voices. Jody Kreiman, Brian Gabelman, and Bruce R. Gerratt (Bureau of Glottal Affairs, Div. of Head/Neck Surgery, UCLA School of Medicine, Los Angeles, CA 90095)

Vocal tremors prominently characterize many pathological voices, but acoustic–perceptual aspects of tremor are poorly understood. To investigate this relationship, two tremor models were implemented in a custom voice synthesizer. The first modulated $F0$ with a sine wave. The second provided ''random'' modulation based on white-noise low-pass filtered with cutoff frequency equal to the nominal modulation rate (usually 1–10 Hz). Control parameters in both models were the rate and extent of $F0$ modulation. (Amplitude modulation was not separately modeled.) Thirty

randomly selected 1-s samples of /a/ were modeled. Two synthetic versions of each vowel were created: one with tremor parameters derived from plots of $F0$ versus time, and one with parameters chosen to match the original stimulus auditorily, regardless of measured tremor values. Listeners judged the similarity of each synthetic voice to the original stimulus. Sine wave and random tremor models both provided excellent matches to subsets of the voices. Preliminary results suggest that tremor rates could be satisfactorily derived from $F0$ plots, but tremor deviations derived from these plots sounded too extreme. These results also demonstrate the advantage of this perceptual analysis-by-synthesis method in integrating acoustic and perceptual approaches to measuring voice quality.

9:15

4aSCa6. Vocal fundamental frequency drop during VCV productions in Japanese speakers. Makoto Kariyasu and John Michel (Dept. of Speech-Lang.-Hearing, Univ. of Kansas, Lawrence, KS 66045-2181, kariyasu@kuhub.cc.ukans.edu)

Vocal fundamental frequency (fo) drop in relation to changes in intraoral pressure (Po) was investigated for adult male and female speakers of Japanese producing /b/ and /v/ in VCV contexts. Three experiments were conducted to examine the following effects on fo drop: (1) vocal tract constriction, (2) vowel context, and (3) speech rate. The results showed that the fo drop from vowels to a consonant was greater for /aba/ than for /ava/; it was greater for high vowel contexts, /ibi/, /abi/, /iba/, than for a low vowel context, /aba/. Speech rate changes with metronome and self-paced had variable effects on the fo drop. Overall, the fo drop for /aba/ ranged from 4% to 13% or from 0.75 to 2.17 semitones. Comparisions with English speakers will be reported. Effects of gender, vocal pitch, and intensity levels, and accent may be discussed.

9:30

4aSCa7. Finite element analysis of vocal fold posturing. Eric J. Hunter (Dept. of Speech Pathol. and Audiol., Univ. of Iowa, Iowa City, IA 52246, eric-hunter@uiowa.edu)

Steady-state vocal fold posturing was simulated with finite-element analysis using three-dimensional elastic elements. The thyroarytenoid muscle was simulated by applying nodal forces along the fiber direction as measured in our laboratory. Other intrinsic laryngeal muscles and cartilage were defined as boundary conditions or applied loads. Because the arytenoid cartilage forms a complex boundary geometry, the arytenoid was modeled as part of the soft tissue, but with much stiffer, nearly rigid body, material constants. The finite-element model is based on a small displacement vibrational model of the vocal folds [Titze and Scherer, *Vocalfold Physiology, Biomechanics, Acoustics, and Phonatory Control,* pp. 183–190] and a large displacement model of the tongue [Wilhelms-Tricarico, J. Acoust. Soc. Am. **97**, 3085–3098 (1994)]. The medial shape of the modeled vocal folds was compared to published histological data.

9:45

4aSCa8. On the mechanism of lowering the voice fundamental frequency in the production of tones of Standard Chinese. Hiroya Fujisaki (Dept. of Appl. Electron., Sci. Univ. of Tokyo, 2641 Yamazaki, Noda, 278-8510 Japan, fujisaki@te.noda.sut.ac.jp)

While it is well known that the cricothyroid (CT) muscle is mainly responsible for lowering the voice fundamental frequency ($F0$) in many languages, the mechanism for $F0$ lowering in languages such as Standard Chinese (SC) has not been elucidated. Although several studies have shown that the sternohyoid (SH) muscle activity is strongly correlated with $F0$ lowering in SC, the mechanism itself is not clear since SH is not directly attached to the thyroid cartilage, whose movement is essential in changing the length and tension of the vocal chord. On the basis of an earlier finding on the production of Thai tones [D. Erickson, Annual Bulletin No. 27, RILP, Univ. Tokyo, pp. 135–149 (1993)], the present author has suggested the active role of the thyrohyoid (TH) muscle in $F0$ lowering in languages such as SC and Swedish [H. Fujisaki, Proc. XXIII World Congress of the International Association of Logopedics and Phoniatrics, pp. 156–159 (1995)]. The present study shows the detailed mechanism for $F0$ lowering involving both TH and SH in the production of the second, third, and fourth tones of SC based on new electromyographic observations.

10:00

4aSCa9. Using time-temperature superposition to estimate the elastic shear modulus and dynamic viscosity of human vocal fold tissues at frequencies of phonation. Roger Chan and Ingo Titze (Natl. Ctr. for Voice and Speech, Dept. of Speech Pathol. and Audiol., The Univ. of Iowa, Iowa City, IA 52242, roger-chan@uiowa.edu)

The principle of time-temperature superposition [J. D. Ferry, *Viscoelastic Properties of Polymers* (Wiley, New York, 1980), pp. 264–320] has been widely used by rheologists to estimate the viscoelastic properties of polymeric materials at time or frequency scales not readily accessible experimentally. This principle is based on the observation that for many polymeric systems molecular configurational changes that occur in a given time scale at a low temperature correspond to those that occur in a shorter time scale at a higher temperature. Thus the viscoelastic properties of these systems empirically measured at various temperatures at a certain frequency are indicative of measurements at different frequencies at a single reference temperature. Using a rotational rheometer, the elastic shear modulus and dynamic viscosity of human vocal fold mucosal tissues were measured at relatively low temperatures (5–25 °C) at 0.01–15 Hz. Data were empirically shifted based on this principle to yield a composite ''master curve'' which gives a prediction of the shear properties at higher frequencies at 37 °C. Results showed that the time-temperature superposition principle may be used to estimate the viscoelastic shear properties of vocal fold tissues at frequencies of vocal fold vibration, on the order of 100 Hz. [Work supported by NIH Grant No. P60 DC00976.]

10:15

4aSCa10. Effects of vocal tract on aerodynamics of hemilarynx. Fariborz Alipour, Douglas Montequin, and Niro Tayama (Dept. of Speech Pathol. and Audiol., The Univ. of Iowa, Iowa City, IA 52242)

Pressure-flow relationship was examined in the excised canine and human larynges with and without vocal tract. Canine and human larynges were prepared and cut in the midsaggital plane from the top to about 10 mm below the vocal folds. The right half was removed and replaced with a Plexiglas plate with imbedded pressure taps along the medial surface. The thyroid cartilage was glued to the plate and the arytenoid was pressed against the plate with a two-pronged probe for adduction control. The vocal tract was simulated with a 15-cm plastic tube of 25-mm diameter. Simultaneous recordings were made of the glottal pressure, mean subglottal pressure, and average airflow at various levels of adduction. Glottal adduction was controlled mechanically by inserting shims of various sizes. Oscillation was generated by the flow of heated and humidified air through the glottis. Preliminary data indicate that the pressure-flow relationships are similar to those of full larynx and are almost linear. The addition of the vocal tract increased the glottal resistance by moving these pressure-flow lines to the lower flow and higher-pressure region. The human larynx appears to phonate easier on the bench and has lower phonation threshold pressure. [Work supported by NIDCD grant DC03566.]

Session 4aSCb

Speech Communication: Potpourri (Poster Session)

Mary E. Beckman, Chair
Department of Linguistics, The Ohio State University, Columbus, Ohio 43210

Contributed Papers

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

4aSCb1. On estimating length and shape of the vocal tract. Edward P. Neuburg (IDA-CCRP, Thanet Rd., Princeton, NJ 08540, epn@idaccr.org)

Automatic vowel recognition is sometimes done by trying to estimate vocal-tract areas, usually derived using linear predictive coding. In the derivation there is a free parameter, namely, the length of the vocal tract; every choice of length leads to a different set of areas. This paper describes a revisit to experiments done in the early 1970s, in which the chosen length is the one that makes the vocal-tract shape most ''human.'' Using some improved criteria for optimality of shape, involving both smoothness and small variance, it is found that: (a) estimates of vocal-tract length are consistent, or at least highly correlated, with actual length; and (b) the tube shape (area function) derived using the optimal length seems to give a reliable characterization of the vowel, independent of the talker (and, in particular, independent of the talker's gender). Classification results are about the same as those using the currently popular cepstral coefficients, but areas have the possible advantage that they are physiological features that the talker can actually control.

4aSCb2. Principal components analysis of x-ray microbeam pellet positions. Robert E. Beaudoin and Richard S. McGowan (Sensimetrics Corp., 48 Grove St., Somerville, MA 02144)

Sets of pellet coordinates from the X-ray Microbeam Speech Production Database, each corresponding to a static articulatory configuration, are submitted to a principal components analysis. Several talkers are analyzed separately, and various criteria are used to select the times in the continuous speech data from which to extract pellet data. In some instances, all of a talker's speech data are submitted to an automatic procedure intended to select syllable nuclei, based on Mermelstein's algorithm [P. Mermelstein, J. Acoust. Soc. Am. **58**, 880–883 (1975)]. In other instances specific data are chosen by hand according to utterance type (e.g., vowel, glide, and consonant–vowel transition). These analyses are intended for use in a project to recover articulation from speech acoustics. In particular, we examine the relationship between pellet positions from data, the tongue shape reconstructed from the principal components analysis, and the parameters of a simple acoustic tube model, which is derived from the Stevens and House model [K. N. Stevens and A. S. House, J. Acoust. Soc. Am. **27**, 484–493 (1955)]. [Work supported by Grant NIDCD-01247 to Sensimetrics Corporation.]

4aSCb3. Articulatory and acoustic characteristics of emphasized and unemphasized vowels. Donna Erickson (Natl. Inst. of Multimedia Education, 2-12 Wakaba, Mihama, Chiba, 261-0014 Japan), Osamu Fujimura (The Ohio State Univ., Columbus, OH 43221), and Jianwu Dang (ATR Human Information Processing Res. Labs., 2-2 Hikaridai, Seika-cho, Soraku-gun, Kyoto, 619-0288 Japan)

This study examines the relation between formants, jaw $x-y$ position and tongue dorsum $x-y$ position for emphasized and unemphasized vowels (/ay/ as in ''high,'' /iy/ as in ''he,'' and /eh/ as in ''head''). Acoustic and articulatory measurements were made for target syllables at the moment of jaw height minimum, comparing emphasized and unemphasized conditions of the same vowel. The results can be interpreted as a hyperarticulation of both jaw and tongue movement for emphasis. When emphasized, mandibular position shows increased lowering with forward movement along the front–back axis of the occlusal plane, for all three vowels. The tongue dorsum shows increased raising and fronting for /iy/ and /eh/ but increased lowering and backing for /ay/. The associated $F1$-$F2$ show a corresponding spreading apart of $F1$-$F2$ for the higher vowels, and a bunching of $F1$-$F2$ for the low vowel consistent with the biomechanics of tongue/jaw articulation. These results are interpreted in light of the C/D model which assumes that prosody affects syllable magnitude, which in turn extrapolates the deviation from the neutral gesture along with more opening of the jaw. The C/D model also has the potential to explain timing discrepancies among the extrema as observed in different articulators.

4aSCb4. Respiratory system changes in relation to prosodic cues at the beginning of speech. Janet Slifka (50 Vassar St., Rm. 36-549 RLE, Cambridge, MA 02139, slifka@mit.edu)

The research presented here examines regional changes in respiratory drive and respiratory-influenced laryngeal adjustments in relation to acoustic signal changes associated with prosody. The respiratory system acts to create subglottal pressure (P_{SG}). When P_{SG} is changing rapidly, such as at the initiation of an utterance at the start of an exhalation, timing and amplitude constraints may be present. The actions of the articulators and the chest wall must be coordinated to rapidly reverse a negative P_{SG} during inhalation to a P_{SG} in the speech range. The current study has involved collection of simultaneous recordings of the acoustic signal and several physiologically related signals including subglottal pressure as estimated from esophageal pressure. Test utterances either begin with a stressed syllable or have one unstressed syllable prior to the first stressed syllable. Present results indicate that: (1) P_{SG} continues to rise from the initiation of an exhale until the first stressed syllable; (2) P_{SG} falls following the initial peak; and (3) fundamental frequency rise is not directly related to the rise in P_{SG}. Applications such as articulatory-related speech

synthesis may benefit from such information. [This research was supported in part by NIH grants #5T32DC00038 and #5R01-DC00266-14.]

4aSCb5. Kinematic evidence for the existence of gradient speech errors. Marianne Pouplier, Larissa Chen (Dept. of Linguist., Yale Univ., New Haven, CT 06511), Louis Goldstein (Haskins Labs., New Haven, CT 06511-6695), and Dani Byrd (Univ. of Southern California, Los Angeles, CA 90088-1693)

Speech errors have long been appealed to as evidence for segmental units in speech production. However, if speech errors can be shown to be potentially gradient, as opposed to always categorical, in nature, the view of segments such as ''swapping places'' must be called into question. Severe limitation of a transcription approach to evaluating speech errors is that gradient errors may go unrepresented in the transcriptional record if they are obscured by other articulatory events due to the coproduction of speech gestures. In a pioneering production study of errors, Mowrey and MacKay [J. Acoust. Soc. Am. **88**, 1299–1312 (1990)] present EMG data that suggest that gradient errors do exist, although they are not always audible. However, the anomalous activity they observe in single-motor-unit recordings does not preclude the possibility that a segmental unit was either still produced correctly via compensation by other muscle activity or omitted in its entirety in the articulatory kinematics. Using magnetometry, the present study for the first time is able to provide articulatory movement tracking data that exhibit gradient errors in phrases like ''cop top'' and ''Bligh Bay'' repeated rapidly. For example, during the [t] in ''cop top'' some tokens show a small (sometimes inaudible) raising of the tongue dorsum. [Work supported by NIH.]

4aSCb6. Articulatory and acoustic data for vowels in isolation and in an /sVd/ context: An x-ray microbeam study. Angela Slama, Gary Weismer, and Kate Bunton (Dept. of Communicative Disord. and Waisman Ctr., Goodnight Hall, Univ. of Wisconsin-Madison, Madison, WI 53706, weismer@waisman.wisc.edu)

Interspeaker variability in articulatory processes has been of great interest in the past decade, but studies addressing this issue have typically been limited to a relatively small group of speakers. In the present study we report on articulatory coordinates and formant frequencies for corner vowels spoken by 50 young adults. The vowels were produced in isolation, and in an /sVd/ context. Articulatory coordinates for the upper and lower lips and four tongue pellets were obtained at temporal midpoint of the vowels using the x-ray microbeam database. Formant frequencies, measured at the same temporal location, were derived from a well-known speech analysis package. Representations of both kinematic and acoustic interspeaker variability will be described and discussed, and relations between the kinematic and acoustic working space for vowels will be examined. A goal of this project is to develop some standards for kinematic/acoustic representations for vowels that can be applied to speakers with dysarthria. [Work supported by NIH DC00820.]

4aSCb7. Kinematics of compensatory vowel shortening: Intra and interarticulatory timing. Susan Shaiman (Dept. of Commun. Sci. and Disord., Univ. of Pittsburgh, 4033 Forbes Tower, Pittsburgh, PA 15260)

The acoustic shortening of vowels has been demonstrated to occur across a variety of contextual variations. The current study examined the kinematic adjustments involved in vowel shortening, as a function of speaking rate and coda composition (singleton consonants versus consonant clusters). Five normal speakers repeated the syllables /paep/, /paeps/, and /paepst/, embedded in a carrier phrase, across three distinct speaking rates (slow, normal, and fast). Changes in the timing of the jaw-closing gesture for post-vocalic bilabial production, and in the upper lip–jaw timing relationship (measured via phase angles), were examined. The onset of the jaw-closing gesture typically shifted earlier in the cycle of jaw movement for consonant cluster productions, across all speaking rates. Subject-specific modifications of interarticulator timing were observed, with some speakers adjusting upper lip movement in order to maintain constant timing with the jaw, while others tended to dissociate the upper lip and jaw. Both coda composition and speaking-rate manipulations resulted in substantial intersubject variability in the lip–jaw timing relationship. These findings suggest that, in order to achieve the intended acoustic-perceptual goals, articulatory coordination may not be absolutely invariant, but rather, systematically and individually organized across task manipulations. [Work supported by CRDF-University of Pittsburgh and NSERC.]

4aSCb8. Using tone similarity judgments in tests of intertranscriber reliability. Julia McGory (Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210), Rebecca Herman (Indiana Univ.), and Ann Syrdal (AT&T Labs.–Res.)

An important, albeit often neglected, aspect of intonational labeling is testing the consistency between labelers. When reliability among different labelers has been tested, two aspects in which transcribers differ from each other is in placement and choice of tone labels. Only identical labels count as ''agreement'' between transcriptions. This can be problematic given that some tone categories are perceived to be similar, making the choice between labels at times difficult. This work suggests a refinement of existing tests of intertranscriber reliability. Previous intertranscriber reliability tests did not take into account each labeler's perceived similarity between pitch accent categories. We examined a set of ToBI transcriptions of a single speaker corpus labeled independently by four experienced intonation labelers. The proposed method incorporates each transcriber's tone similarity judgments, which were elicited for the purposes of this experiment. The labelers judged the similarity of pitch accent pairs using a rating scale of 1–7. Multidimensional scaling (MDS) of similarity judgements resulted in quantifiable measures of degree of similarity among tones. Euclidean distances between pairs of tones were calculated from the MDS results. We used these values as the weights in calculating mean weighted agreements. [This work is supported by AT&T Labs.–Research.]

4aSCb9. Contextual influences on the internal structure of phonetic categories: A distinction between lexical status and speaking rate. J. Sean Allen and Joanne L. Miller (Dept. of Psych., 125 NI, Northeastern Univ., Boston, MA 02115)

A series of experiments examined the effects of an acoustic-phonetic contextual factor, speaking rate, and a higher-order linguistic contextual factor, lexical status, on the internal structure of a voicing category, specified by voice-onset-time (VOT). In keeping with previous results, speaking rate fundamentally altered the structure of the voiceless category, not only affecting the perception of stimuli in the voiced–voiceless category boundary region but also altering which tokens were rated as the best exemplars of the voiceless category. In contrast, the effect of lexical status was more limited. Although (as expected) lexical status also affected the perception of stimuli in the category boundary region, this effect disappeared in the region of the best-rated exemplars. This distinction between the effects of speaking rate and lexical status on the internal structure of the voiceless category mirrors the effects of these factors in speech production: It is well known that speaking rate alters the VOT values of voiceless consonants, whereas we confirmed in a speech production experiment that lexical status has no such effect. Based on these findings, we argue that higher-order contextual factors such as lexical status operate differentialy in processing from production-based, acoustic-phonetic factors such as speaking rate. [Work suppported by NIH/NIDCD.]

4aSCb10. Acoustic and articulatory differences across word position for American English /r/. Suzanne E. Boyce (Univ. of Cincinnati, Mail Location 379, Cincinnati, OH 45221) and Carol Y. Espy-Wilson (Dept. of Elec. and Computer Eng., Boston Univ., Boston, MA 02216)

Typically, American English /r/ is manifested with different formant frequency patterns in word-initial, word-final and word-medial position. In particular, it has been reported (Lehiste, 1962; Espy-Wilson, 1992) that initial /r/'s show lower third formants. This may be due to differences in position or configuration of the primary articulator for /r/—the tongue—or it may reflect an increased degree of lip-rounding. Differences in tongue shape across word position have been reported (Zawadaski and Kuehn, 1982), but it is not clear how reported differences may relate to formant lowering. In this study, eight native speakers of American English recorded words with initial, final and medial /r/'s. Articulatory movements of the tongue tip, tongue blade and tongue dorsum, lips and jaw were tracked simultaneously by Electro-Magnetic Midsagittal Articulometer (EMMA). For each speaker, we compare spatial position, time course, and trajectory for articulators across positional variants. Preliminary data suggest that tongue articulators show similar time course in all three positions. Results are discussed with regard to acoustical modeling studies, theories of articulatory strengthening and interarticulator timing. [Research supported by NIH and NSF.]

4aSCb11. Proportional and linear constancy in a repetitive speech production task. Kenneth de Jong (Dept. of Linguist., Indiana Univ., Bloomington, IN 47405)

Speech production studies have shown local temporal stabilities in intergestural timing and in its acoustic consequences [e.g., Kent and Moll, JSHR (1975)], such that the absolute duration of certain speech events is remarkably constant over differences in speech rate. Other production studies examining longer stretches of speech have found event durations to be proportionally constant with respect to a larger frame. Proportional constancy is especially apparent in repetitive production tasks [Cummins and Port, J. Phon. (1998)]. This paper examines how absolute constancy, such as would be specified by segmental contrasts, and proportional constancy interact in a repetitive speech task. Speakers repeated linguistically specified syllables in time to a metronome which specified repetition rate. Rates were varied by a factor of 2.7. Durations of speech events which indicate segmental contrasts, such as voice onset time in onset stops, remain fairly constant over these large changes in speech rate. Other durations, such as vowel durations in open syllables, often show proportional constancy. Vowel durations before coda stops, where duration acts as a secondary cue to consonant voicing, often exhibit linear constancy, suggesting a difference in the degree to which linear linguistic factors restrict how speakers perform a repetitive task.

4aSCb12. Commonality in the dorsal articulations of English vowels and liquids. Bryan Gick, A. Min Kang, and D. H. Whalen (Haskins Labs., 270 Crown St., New Haven, CT 06511)

Phonological studies have predicted that the dorsal articulations of English /r/ and /l/ correspond with those of schwa and open o, respectively [Gick, Phonology (in press)]. Specifically, /r/ and schwa are hypothesized to share pharyngeal configuration, while /l/ and open o share upper pharyngeal/uvular configuration. To test this prediction, midsagittal MRI images of the vocal tract of a male speaker of American English were collected and midsagittal distance (of airspace above the tongue surface) measured at 44 3-mm intervals along the vocal tract length. Regions of the vocal tract were defined as pharyngeal, uvular and oral, as in Whalen *et al.* [JSLHR (in press)], with the pharyngeal region divided into upper and lower halves. Midsagittal distances were collected for eleven sustained vowels plus /r/ and /l/. Distances for average vowels were subtracted point by point from /r/ and /l/ and a single rms calculated within each region of the vocal tract. As predicted, in the upper pharyngeal and upper pharyngeal/uvular regions, /l/ showed the greatest correspondence with open o, while /r/ was most similar to schwa throughout the pharynx. These results support the phonological interpretation of the dorsal gestures of English liquids as vocalic. [Work supported by NIH grant DC-02717.]

4aSCb13. Acoustic modification in English place assimilation. David W. Gow, Jr. (Neuropsychol. Lab., VBK 821, Massachusetts General Hospital, 55 Fruit St., Boston, MA 02114, gow@helix.mgh.harvard.edu) and Peter Hussami (MIT, Cambridge, MA 02139)

Recent behavioral research suggests that listeners hearing words containing segments that have undergone place assimilation are able to recover the underlying form of the modified segment and anticipate the place of the segment that triggers assimilation. The present study contrasts acoustic place cues in unmodified coronals, assimilated underlying coronals, and underlying noncoronals in connected read speech in an attempt to characterize the nature of acoustic modification produced by place assimilation, and to understand how a single segment might encode the places of articulation of two segments. Adult male and female speakers produced triplets of sentences showing this three-way contrast across a variety of consonants and vowel contexts. Acoustic measures examined formant transitions and relative amplitudes. The implications of the results of these analyses for the the structure of acoustic categories for place information and the abstract representation of place are discussed.

4aSCb14. An interactive atlas of English vowels: Design considerations. Robert Hagiwara, Sharon Hargus, Richard Wright (Dept. of Linguist., Univ. of Washington, Box 354340, Seattle, WA 98195), and Isaac Sterling (Univ. of Washington, Seattle, WA 98195)

Discussions of vowel quality differences in dialects of English are often based on imprecise, subjective descriptors (more open, slightly less back) and transcriptions rather than instrumental measurements. Available instrumental data focus on individual dialects rather than a broader picture of dialectal variation. In part to address these concerns, a database of vowel formant frequencies is currently being developed at the University of Washington, beginning with a sampling of static formant frequencies in the American West. The ultimate intent is to produce a web-based, expandable database, or atlas, of reference English vowel formant frequencies and audio examples. It is intended primarily as a pedagogical tool to accurately demonstrate vowel quality variation, and as a ''jumping off point'' for more intensive studies of acoustic variation. This poster discusses long- and short-term goals of the project, current data, the standardized methodology being used, web tools being developed, and the possibilities for future collaboration in data collection and analysis.

4aSCb15. Variations in temporal patterns of speech production among speakers of English. Bruce L. Smith (Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL 60208-3570)

Results from most studies concerning temporal properties of English are typically presented as group averages. Although various trends emerge from such analyses, relatively little is known about the extent to which individual speakers actually exhibit these tendencies. While some amount of variation among subjects is to be expected, it is also important to determine how regularly individual speakers manifest such temporal patterns. The present study examined several hundred productions by each of 10 native, adult speakers of English to assess the extent to which they showed various temporal characteristics. While a majority of subjects demonstrated, in varying degrees, most of the temporal patterns considered, some did not. For example, most speakers exhibited phrase-final vowel lengthening, averaging approximately a 20%–25% increase for the group across several different words; however, several speakers either did not show phrase-final vowel lengthening for any stimuli, or they showed it for certain words but not others. In addition, the range of performance among speakers who did manifest phrase-final vowel lengthening varied

from as little as 5% increases for some stimuli to more than double for others. Implications of these findings for development, second language acquisition, etc., will be discussed.

4aSCb16. Place of articulation for consonants: Comparing nasals and stops, and syllable position. Kenneth N. Stevens, Sharon Y. Manuel, and Melanie L. Matthies (Res. Lab. of Electron. and Dept. EECS, MIT, Cambridge, MA 02139, stevens@speech.mit.edu)

This study reports on measurements of transitions of the second formant ($F2$) for syllable-initial and syllable-final nasal and stop consonants in English, as cues to place of articulation. $F2$ values and the amount and direction of $F2$ changes in a 20 ms interval were determined in vowels adjacent to consonant ''implosion'' and release. The corpus included consonants in sentences and isolated nonsense syllables. The $F2$ transitions for a given place of articulation are roughly similar for nasals and stops for different syllable positions, as expected, but some interesting differences exist. Since the stop-consonant bursts influence the spectrum sampling point relative to release, there are some shifts in $F2$ onset frequencies for stops, as compared to nasals. Syllable-initial alveolars show additional manner differences, which can be attributed to differences in tongue configurations for alveolar nasals relative to stops in back-vowel contexts. These latter shifts are tentatively ascribed to differential coarticulatory effects for /n/ versus /d/ and /t/ differences which might be related to the lack of a distinctive syllable-initial velar nasal in English. There are also systematic shifts in $F2$ transitions as a function of syllable position, for both nasals and stops. [Supported in part by NIH Grants DC00075 and DC02525.]

4aSCb17. Effects of speech rhythm on timing accuracy of stressed syllable beats. Robert F. Port, Mafuyu Kitahara, Kenneth de Jong, David R. Collins, and Deborah Felkins Burleson (Linguistics Dept., Indiana Univ., Bloomington, IN 47405)

It has been shown that English speakers have a strong tendency to locate stressed syllable onsets (that is, ''beats'' or ''P centers'') near harmonic fractions (like 1/2, 1/3, 2/3) of the repetition cycle of a repeated phrase [Cummins and Port, J. Phonetics (1999)]. These results were interpreted as evidence of a role for frequencies at multiples of the repetition cycle creating temporal attractors for stressed syllable onsets at harmonic fractions of that cycle. If harmonic oscillators account for temporal attractors, then, having an attractor at 2/3 should necessarily imply the existence of an attractor at 1/3, whereas an attractor at 1/2 would not similarly imply an attractor at 1/4 (since a third oscillator would be required). Thus, if a phrase like *take a pack of cards* is repeated with *card* onset at phase 1/2, there should be a weaker attractor controlling the timing accuracy for *pack* onset than if *card* onset is at 2/3. Four speakers were tested using various text materials and speaking rates. Preliminary results support the hypothesis predicting greater temporal variability for *pack* with *card* beginning at 1/2 than at 2/3.

4aSCb18. Evidence from /k/ versus /t/ burst spectra for variable lingual contact precision in normal versus atypical phonological development. Jan Edwards, Marios Fourakis (Speech and Hearing Sci., Ohio State Univ., Columbus, OH 43210-1002, edwards.212@osu.edu), Mary E. Beckman, Pauline Welby, and Satoko Katagiri (Ohio State Univ., Columbus, OH 43210-1298)

Almost universally, phoneme inventories include at least one pair of lingual consonants contrasting an apical ''front'' contact to a dorsal ''back'' contact. Studies of infant productions across languages suggest that control of this contrast is acquired later than that between lingual and labial contact, and ''velar fronting'' (apparent substitution of /t/, /d/ for /k/, /g/) is a commonly observed pattern of functional misarticulation in English-acquiring children. This study uses spectral moments analysis of the burst to examine the patterns of emerging differentiation of alveolar and velar voiceless stops in eight pairs of preschool children with phonological disorder and normal age peers. Productions of four adults showed consistently higher skewness values and lower center of gravity for /k/ as compared to /t/. The children showed considerable intersubject variability, with some exaggerating the adult distinction and others showing substantial overlap. There was especially extreme overlap in the productions by children with ''velar fronting.'' However, the spectral moments values for these children did not suggest a categorical substitution. Instead, values were in the middle of the distribution on both dimensions, as if the child were making an ''undifferentiated lingual gesture'' [F. E. Gibbon, J. Speech. Hear. Language Res. **42**, 382–397 (1999)]. [Work funded by NIDCD No. DC02932.]

4aSCb19. Attention and laterality preferences in children who clutter. Gordon W. Blood, Ingrid M. Blood, and Glen W. Tellis (Dept. of Commun. Disord., Penn State Univ., University Park, PA 16802, F2X@psu.edu)

Children who clutter were compared to children who stutter and children who did not stutter on a synthetically generated dichotic consonant-vowel listening task and a timed response assessment task of attention. The Dichotic Listening Test consisted of 120 pairs of synthetically generated stop consonants. Subjects used a pointing response and tasks involved three listening conditions (free recall, directed left, directed right). The Conners' Continuous Performance Test was used to obtain an overall attentional index score, number of correct responses, percentage of omissions, percentage of commissions, hit reaction time, and beta and sensitivity scores. Results revealed that during the directed right and directed left listening tasks children who cluttered and control subjects showed right and left ear advantages. Children who stuttered demonstrated mixed laterality on the DLT. It appeared that children who cluttered benefitted directly from a task which directed their attention to a specific ear. Results of the Continuous Performance Test supported this hypothesis. Children who cluttered showed poorer performance on the overall attentional index as well as a number of the subscores than children who stuttered and children who did not stutter. Implications for future research will be discussed.

4aSCb20. Final consonant recognition in young children. Jan Edwards, Robert A. Fox, and Catherine L. Rogers (Dept. of Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, edwards.212@osu.edu)

Research has shown that typically developing children evidence gradual improvement with age on speech perception tasks. Studies have also shown that children with phonological disorder and specific language impairment perform more poorly than typically developing age peers on these same tasks. Previous research has mostly used synthetic speech stimuli, although it is known that additional attentional resources are required to process synthetic speech. Furthermore, most research has focused on identification or discrimination of word-initial consonants, rather than on consonants in other word positions. In our research, experimental paradigms were developed that use digitized natural speech, rather than synthetic speech, and focus on final rather than initial consonants, since final consonants tend to be acquired later in production. Results of a study with 15 typically developing 3- and 5-year olds will be presented. Familiar word pairs (cap versus cat and tap versus tack) were recorded, digitized, and presented at three different gates (up to 60 ms removed from final consonant closure). Values of d-prime were calculated for each gate, and an improvement in identification with age was found for the longer gates. Results will also be presented comparing 10 children with phonological disorder and age peers. [Research supported by NIDCD No. DC02932.]

4aSCb21. Children's and adults' perception of speech in noise. Marianne Fallon, Sandra E. Trehub, and Bruce A. Schneider (Dept. of Psych., Univ. of Toronto, 3359 Mississauga Rd. North, Mississauga, ON L5L 1C6, Canada, marianne@psych.utoronto.ca)

Age-related differences in identifying speech in noise were investigated by means of a computerized four-alternative forced-choice task. In experiment 1, children 5 to 11 and adults heard prerecorded instructions (e.g., ''Touch the X'') mixed with multitalker babble. Selections were made on a touch-sensitive screen and corrective feedback was provided. To control for age-related differences in sensitivity, signal-to-noise ratios yielding 85% correct were determined for each age group. The S/N difference between the youngest and oldest listeners was 5 dB, consistent with previous research. Equivalent additions of background noise had comparable effects for all age groups, which implies that previously reported age differences in noise were attributable to sensitivity differences rather than noise effects. Experiment 2 investigated the degree to which sentential context affected speech identification in similar noise. Children 5 and 9 years of age and adults identified the final word in high-context (e.g., ''I carried the water in a pail'') and low-context sentences (e.g., ''He looked at the pail''), with differences in auditory sensitivity controlled. Adults benefited ore from high sentential context than did younger children, indicating that limitations in cognitive processing adversely affect children's perception of speech in noise. [Work supported by the Medical Research Council of Canada.]

4aSCb22. Possible perceptual cues for accent distinction in whispered speech—A case of an aphonic child. Sawako Hirai, Noriko Kobayashi, and Hajime Hirose (School of Allied Health Sci., Ktasato Univ., 1-15-1, Kitasato, Sagamihara, Kanagawa, 228-8555 Japan)

Japanese word accent is associated with pitch change and $F0$ change is its major perceptual cue. In whispered speech, however, $F0$ is absent. We experienced a 10-year-old boy who had been using only whispered speech since birth, although he had no apparent physiological or psychological problems. After voice therapy, he started using voiced speech, but accent distinctions were poorly perceived in his voiced speech, while his whispered speech appeared to have better accent pattern variations. The purpose of this study was to find out the acoustic characteristics and perceptual cues produced by the child to make word accent distinctions in his voiced and whispered speech. The subject produced a vowel sequence /aa/ with different accent patterns. The results showed that accent distinctions were correctly perceived by listeners in both conditions, and that $F1$ change was significantly larger in whispered utterances than in voiced ones. The present result appeared to support the hypothesis of Higashikawa *et al.* (1997) that $F1$ change played a principal role in accent distinction in whispered speech. It was assumed that when the major cue ($F0$ change) was not available, another possible cut, $F1$ change, was exaggerated.

4aSCb23. Which dyslexic children have speech perception difficulties? Patricia A. Keating (Phonet. Lab, Linguist. Dept., UCLA, Los Angeles, CA 90095-1543, keating@humnet.ucla.edu), Marc F. Joanisse, Franklin A. Manis, and Mark S. Seidenberg (Univ. of Southern California)

The incidence of speech perception difficulties, and their co-incidence with other impairments, were examined in third-grade poor (bottom-quartile) readers. Several subgroups of dyslexics, plus age and reading-level controls, were compared in their identification of /d/ vs /t/ in a ''dug–tug'' continuum, and /p/ vs /k/ in a ''spy–sky'' continuum. Only those dyslexics who also did poorly on two standardized language tests of inflection and vocabulary—9 of 63 dyslexics—differed from controls in performance on these speech sound categorization tasks, with shallower categorizations. In contrast, dyslexics who did poorly on a test of phoneme deletion, but normally on the language tests, performed like controls in speech perception. Thus an association was found between speech perception and morphological/lexical knowledge, but not between speech perception and phonological knowledge—an unexpected result. A follow-up study tested AX discrimination using the ''spy–sky'' stimuli. The same dyslexic children who had done poorly on the identification tasks differed from controls in discrimination; the other dyslexics again performed like controls. However, the deficit of the poorer perceivers is not auditory—their across-category discrimination was normal, and their within-category discrimination better than normal. Thus their difficulty is specifically with speech sound categorization. [Work supported by NICHD, NIMH, and NSERC.]

4aSCb24. Building and exploring (speech) categories in early infancy. Francisco Lacerda (Dept. of Linguist., Stockholm Univ., S-106 91 Stockholm, Sweden, frasse@ling.su.se)

This paper will report the results from a speech perception study assessing the impact of correlated audio and visual information on category formation in infancy. The amount of speech and of visual information as well as the degree of correlation between the two modalities is systematically varied across subjects and will be subsequently related to the infants' success in picking up the categories implicit in the multisensory stimuli. Also the infants' active versus passive role during the conditioning phase is experimentally assessed in an attempt to study how attentional effects may influence the early stages of speech category formation. The study is designed to provide experimental evidence for the notion of emergent phonology and the exemplar-based approach presented by Lacerda (1998) at the 135th ASA meeting, Seattle.

4aSCb25. The relation between infant speech perception and early communicative development: A longitudinal study. Huei-Mei Liu, Feng-Ming Tsao, Erica B. Stevens, and Patricia K. Kuhl (CHDD, Box 357920, Univ. of Washington, Seattle, WA 98195)

This study investigated the correlation between speech discrimination in young infants and early communicative development. The MacArthur Communicative Development Inventory, a parent-report measure of words and gestures, was administered to 20 13-month-old infants who had participated in a study of speech discrimination when they were 6 months old. In the earlier study, the conditioned Head-Turn (HT) technique was used to assess infants' discrimination of the Finnish vowels /i/ and /y/. Infants were required to meet a criterion of seven out of eight correct responses prior to completing a 30-trial test phase. Results showed that the number of trials infants needed to meet the criterion at 6 months of age was significantly negatively correlated at 13 months with the number of phrases understood (-0.662) and words understood (-0.700). In a regression analysis, the number of criterion trials was shown to be moderately predictive of later communicative development, accounting for approximately 40% and 45% of the variance in phrases understood and words understood. The development of these children will be followed with a second administration of The MacArthur Communicative Development Inventory when they are 16 months of age, to further map the relationship between infant speech perception and early communicative development.

4aSCb26. Relationships between expressive vocabulary size and spoken word recognition in children. Benjamin Munson (Dept. of Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, Columbus, OH 43212)

Studies have shown that children require more acoustic information than adults to accurately identify spoken words [Munson *et al.*, J. Acoust. Soc. Am **99**, 2591 (1996)]. In this study, 61 children aged 3.0 to 7.11 were tested to assess whether spoken word recognition accuracy was predicted by scores on standardized tests of expressive and receptive vocabulary, articulation ability, preliteracy skills, and phonological awareness. Two word recognition tasks were used: a gated series [Elliot *et al.*, Percept. Psychophys. **42**, 150–157 (1987)] in which subjects identified CVC words

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with missing final consonants, and a noise-center series [Fox *et al.*, J. Speech Hear. Res. **35**, 892–902 (1992)], in which subjects identified CVC words with the medical vowel replaced by noise. Each task had four experimental conditions. Subjects' age predicted a significant proportion of variance in word recognition scores on the five conditions for which the majority of subjects performed above chance. Additionally, scores on a test of expressive vocabulary predicted a significant proportion of variance in two of these five conditions. No other test predicted relationships between expressive vocabulary growth and developmental changes in spoken word recognition. [Work supported by NIH.]

4aSCb27. Developmental changes in perceptual weighting strategies for speech are contrast specific. Susan Nittrouer and Marnie E. Miller (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

In the labeling of /s/-/ʃ/, compelling evidence exists that children pay more attention to formant transitions and less attention to noise spectra than adults do. The purpose of this work was to examine whether such age-related differences extend to other contrasts, specifically /f/-/θ/. This contrast was chosen because compelling evidence also exists that adults pay more attention to formant transitions and less attention to noise spectra for /f/-/θ/ decisions than they do for /s/-/ʃ/ decisions, undoubtedly because /f/ and /θ/ noise spectra provide little information. Thus adults' weighting strategies for this contrast should resemble those of children. To test that hypothesis, children (4, 6, and 8 years) and adults labeled fricative-vowel syllables consisting of natural /f/ and /θ/ noises combined with synthetic /ɑ/ and /u/ portions in which formant transitions varied from ones appropriate for /θ/ to ones appropriate for /f/. Results showed similar weighting strategies for adults and children for /ɑ/ syllables, but adults and 8-year-olds used the sparse information in the noise spectra for /u/ syllables to a surprising extent. Conclusions were that weighting strategies for some contrasts require no developmental adjustments, but that flexibility in perceptual weighting strategies is one general developmental change.

4aSCb28. The influence of pitch contour on vowel discrimination in infants. Laurel J. Trainor and Renée N. Desjardins (Dept. of Psych., McMaster Univ., Hamilton, ON L8S 4K1, Canada, ljt@mcmaster.ca)

Because the harmonics are farther apart in high- than in low-pitched vowels, it was predicted that it should be difficult to extract the frequencies of the formant resonances, and therefore, to discriminate high-pitched vowels. Infant-directed speech is very high in pitch, yet studies showing that young infants can discriminate vowels have used low-pitched male voices. The present study showed that 6-month olds were able to discriminate /i/ and /I/ when the pitch was 240 Hz, but were unable to do so when the pitch was 340 Hz, demonstrating that the high pitch of infant-directed speech indeed impedes vowel discrimination. How, then, do infants learn vowel discrimination? The answer may lie with the large pitch contours that are also characteristic of infant-directed speech. Because the harmonic structure follows the frequency of the fundamental, but the formant structure remains relatively constant across frequency shifts, a second prediction was made: discrimination should be better for vowels containing pitch contours than for those with steady pitch. This is exactly what was found. The addition of a downward pitch contour at both high (440–340 Hz) and low (240–140 Hz) pitches significantly improved infants discrimination of /i/ and /I/. [Research supported by NSERC.]

4aSCb29. Differences in child and adult loudness judgments of rock music. Donald Fucci (Lindley 219, School of Hearing and Speech Sci., Ohio Univ., Athens, OH 45701), Heather Kabler, Debbie Webster, and Doug McColl (Ohio Univ., Athens, OH 45701)

The present study was concerned with the perceptual processing of complex auditory stimuli in ten children ($\underline{M}$ age=8.1) as compared to ten young adults ($\underline{M}$ age=19.3) and ten older adult subjects ($\underline{M}$ age=54.2). The auditory stimulus used was 10 s of rock music [Led Zeppelin, CD Recording No. 19127−2 (Atlantic Recording Group, New York 1969)]. All three groups provided numerical responses to nine intensity levels of the rock music stimulus (10, 20, 30, 40, 50, 60, 70, 80, and 90 dB above threshold). Results showed that the children demonstrated a wider range of numerical responses than both adult groups. The range of mean numerical responses for the children was 0.54 to 54.24. For the young adults the range was 0.76 to 11.37, and for the older subjects it was 1.6 to 23.31. Results suggest that the children were not bound by the same set of rules as the adults in regard to magnitude estimation scaling of the loudness of the rock music stimulus. Their internal scaling mechanisms appeared to be more flexible and broader based than those of the adults who participated in this study.

4aSCb30. Speech intelligibility and vowel production in persons with dysphagia. Fredericka Bell-Berti, Diane M. Scott, Rita Kachmarchyk, and Nancy M. Colodny (Dept. of Speech, Commun. Sci. and Theatre, St. John's Univ., Jamaica, NY 11439, bellf@mail.stjohns.edu)

Neurologically based dysphagias are often accompanied by dysarthria or apraxia. Dysarthrias have traditionally been identified perceptually, rather than with more objective instrumental measures, and vowel articulation is examined only rarely in dysarthria evaluations. It is possible that some speakers with neurologically based dysphagia have sub-clinical disturbances of speech that affect vowel as well as consonant production. This preliminary report examines the speech intelligibility and acoustical measures of vowel productions in a group of dysphagic subjects. Our goal is a better understanding of the relations between the dysphagia and the dysarthria. [Work supported by St. John's University.]

4aSCb31. Intonation and speech timing: Association or dissociation? Samuel A. Seddoh (Dept. of Speech Pathol. & Audiol., Southern Univ., P.O. Box 9227, Baton Rouge, LA 70813, seddoh@cluster.engr.subr.edu)

A number of prosodic studies on brain damaged patients [e.g., Gandour *et al.*, Brain & Lang. **43**, 275–307 (1992)] have concluded that the production of fundamental frequency ($F0$) is tied to temporal control, and that abnormalities in $F0$ associated with intonation exhibited by aphasic patients is due to underlying disturbance in speech timing. The present study examined $F0$ and durational measures in syntactically simple English statements and interrogatives produced by fluent and nonfluent aphasic patients as well as a group of normal subjects. Measures conducted included initial $F0$ peak (P1), the initial frequency at the terminal region of the $F0$ contour (TIF), amount of terminal $F0$ fall/rise, utterance length, duration of terminal $F0$ fall/rise, and distance between P1 and TIF. Results showed that the nonfluent aphasic patients performed at comparable levels with the normal subjects on all $F0$ measures, but they were severely impaired across the board on temporal measures. The fluent aphasic patients' performance on both sets of measures were generally comparable to normal performance, although in absolute terms their durations were also longer than normal. These findings contradict the view that $F0$ production depends on speech timing. Rather they suggest that the two parameters may be dissociated.

4aSCb32. The effect of speaking rate on vowel production in Parkinson's disease. Kris Tjaden (Dept. of Communicative Disord. and Sci., The State Univ. of New York at Buffalo, 3435 Main St., Buffalo, NY 14214-3005)

Kinematic and acoustic descriptions of articulatory characteristics in Parkinson's disease (PD) generally are consistent with reports of hypokinesia and bradykinesia in the limbs. That is, articulatory movements tend to be underscaled and produced at slower than normal rates in comparison

to healthy speakers, although movement duration may be preserved. Slowed speaking rate is one therapy technique used for improving the adequacy of speech production in PD. The assumption is that the slowed rate allows for more extensive articulatory displacements, thereby increasing the acoustic-perceptual distinctiveness of phonetic events. Few studies have quantified articulatory changes associated with speaking rate variation in PD, however. The present study examined the effect of speaking rate on vowel production for individuals with PD and healthy controls. Participants read a passage at habitual, slow, and fast rates. Segment durations, $F1$, and $F2$ midpoint frequencies for the vowels /i/, /ɑ/, /u/, and /ae/ were measured. Vowel space area was calculated for each speaker's habitual, fast, and slow reading rate. The change in vowel space area was related to the magnitude of speaking rate change across rate conditions for both groups. Vowel space areas for PD and healthy speakers also were compared for each rate condition. [Work supported by NIH.]

4aSCb33. Effects of a computer-based voice modification device for a speaker with recurrent nerve paralysis. Noriko Kobayashi, Hajime Hirose (School of Allied Health Sci., Ktasato Univ., 1-15-1, Kitasato, Sagamihara, Kanagawa, 228-8555 Japan), Kenji Matsui, and Noriyo Hara (Matsushita Electric Co., Kyoto, 619-0237 Japan)

Patients with unilateral recurrent laryngeal nerve (RNL) paralysis may exhibit severe dysphonia depending on the resting position of the affected vocal fold. Major vocal symptoms of these patients are breathy voice, decreased vocal intensity, and physical fatigue due to great effort to approximate the vocal folds and to increase subglottal pressure. In some cases, phonosurgery is not applicable due to other medical problems, while a good prognosis is not expected by voice therapy. Technological assistance may be needed for such a patient. In this study, a real-time formant analysis-synthesis method was used to enhance speech intelligibility of a severely dysphonic patient with unilateral RNL paralysis. The voice source was modified by using inverse-filtered signals extracted from a normal speaker. A special hardware unit was designed to perform an analysis-synthesis process with temporal delay of 40 ms for signal processing. Vowels and sentences produced by the subject both with and without the device were analyzed. An approximately 20 dB increase of overall loudness level and increased harmonic components were observed for the processed speech. Perceptual judgment revealed improved speech intelligibility and less breathy voice quality for the processed speech. The subject reported less fatigue with the device as less effort was required for phonation.

4aSCb34. Structure of phonetic categories produced by general learning mechanisms. Andrew J. Lotto (Dept. of Psych. and Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626), Lori L. Holt (Carnegie Mellon Univ., Pittsburgh, PA 15213), and Keith R. Kluender (Univ. of Wisconsin, Madison, WI 53706)

The development of categories for complex auditory stimuli is an interest for both studies of general category learning and language acquisition. Previous work [Kluender *et al.*, J. Acoust. Soc. Am. **104**, 3568–3582 (1998)] demonstrated that avian species can learn to respond differentially to sounds from two vowel categories and the structure of their responses correlate well with human adult ratings of the vowels. In the current study, Japanese quail (*Coturnix japonica*) were trained to respond to either members of an /i/ or /ɛ/ distribution and to refrain, in both cases, from responding to members of an /I/ and /æ/ distribution. Birds responding to /ɛ/ (surrounded by /I/ and /æ/ in the vowel space) showed a prominent peak or ''prototype'' in their responses. Birds responding to /i/ (extreme in the vowel space) showed a weak or no ''prototype,'' but showed a strong gradient with response rate increasing for tokens further away from the other vowel distributions in the $F1$–$F2$ space. These data demonstrate that internal structure of (phonetic) categories is strongly influenced by relations to the competing stimulus set (vowel space). This is particularly important for theories of categorization or language acquisition that rely heavily on the existence of a ''prototype.''

4aSCb35. Influence of fundamental frequency on stop-consonant voicing perception: A case of learned covariation or auditory enhancement? Lori L. Holt (Dept. of Psych., Carnegie Mellon Univ., Pittsburgh, PA 15213), Andrew J. Lotto (Loyola Univ. Chicago, Chicago, IL 60626), and Keith R. Kluender (Univ. of Wisconsin-Madison, Madison, WI 53706)

Listeners labeling members of an acoustic series modeling VOT (e.g., /ba/-/pa/) are more likely to identify tokens with higher $f0$ as voiceless than they are for otherwise-identical tokens with lower $f0$s. This pattern of results may arise because a high $f0$ enhances perception of voicelessness, in line with auditory enhancement accounts of speech perception. Alternatively, because $f0$ and VOT covary in English production, it is possible that listeners respond in this manner due to experience with VOT/$f0$ covariation in the speech signal. The present investigation was designed to tease apart the relative contributions of these two potential mechanisms. Japanese quail (Coturnix coturnix japonica) were trained to ''label'' stimuli drawn from VOT series by pecking a key. During training, each quail experienced one of three styles of VOT/$f0$ covariation. For one group of quail, VOT and $f0$ covaried naturally with voiceless series members having higher $f0$s than voiced members. Another group of quail heard the inverse, ''unnatural'' covariation. A final group experienced stimuli for which there was no covariation between VOT and $f0$. Results indicate that experience with VOT/$f0$ covariation is the predominant force in shaping perception. Thus, general learning mechanisms may account for this symmetry between perception and production.

Session 4aSP

Signal Processing in Acoustics: Wavelets in Acoustics

Leon H. Sibul, Chair
Pennsylvania State University, P.O. Box 30, University Park, Pennsylvania 16804

Chair's Introduction—8:15

Invited Papers

8:20

4aSP1. Wavelet domain system/channel characterization. Randy Young (Appl. Res. Lab, Penn State Univ., P.O. Box 30, State College, PA 16804)

The primary operators in implementing wavelet transforms are the delay, scale, and correlation operators. The physical motion of sources and sensors causes signals to be compressed or dilated (scaled) depending upon their relative direction of motion. For many systems that use a signal modulation with a carrier frequency, this scaling operator leads to a Doppler shift of the carrier. For many acoustic signals (voice, machinery vibrations, structural vibrations, biomedical signals, etc.), they are not carrier-frequency modulated and they possess a large relative bandwidth. Additionally, the relatively slow propagation speeds compared to source/sensor motion leads to much higher compression/dilation ratios than in electromagnetics. Since scaling applies a different Doppler to each frequency, large bandwidth signals experiencing scaling are not well approximated by a single Doppler shift. Thus, the scaling operation, and, therefore, wavelet transforms are more applicable in acoustic analysis/system modeling than electromagnetic analysis. This discussion concentrates on the wavelet transform as an operator utilizing scaling, rather than a signal decomposition technique. Although wavelet transforms as signal decomposition tools have been extensively researched, their contribution as a system/channel analyzer for estimating motion of sources/sensors/environments is less developed. [Nancy Harned, ONR321US, is acknowledged for supporting this work (Contract N00039-97-D-0042).]

8:50

4aSP2. Wavelet processing for wideband spreading function estimation. Lora G. Weiss (Appl. Res. Lab, Penn State Univ., State College, PA 16804)

The statistic used to quantify the amount of environmental spreading a signal undergoes as it traverses through a channel is called a spreading function, and it includes the effects of moving, distributed scattering objects, multipath, boundary effects, etc., on the signal. Traditionally, narrow-band signals have been transmitted, and the spreading functions were estimated by calculating the outputs of narrow-band-matched filters. Now that wideband processing has become more accessible, we need a solid concept of estimating wideband spreading functions. This paper shows that a wideband spreading function can be estimated by computing the wavelet transform of the received signal while using the transmitted signal as the mother wavelet. The paper then computes the second-order statistic, called the wideband scattering function, associated with the wideband spreading function. To assess the total scattering, several scattering functions are convolved to yield an overall representation of the environment. This representation can then be incorporated into a detection processor. The payoff is that if any portion of the scattering environment is known *a priori*, this information can be exploited in the detector. As knowledge of the scattering process is acquired, it is combined via the cascaded scattering function formulation.

9:20

4aSP3. Application of wavelets to scattering for wave-packet synthesis. Charles F. Gaumond and David M. Drumheller (Code 7142, Acoust. Div. Naval Res. Lab., Washington, DC 20375-5320)

The wavelet transform can be used to analyze the time-scale domain structure of echoes derived from broadband, monostatic, acoustic scattering from ribbed shells. In the time-scale domain, the echoes can be seen to be composed primarily of a few individual components. Each component corresponds to a separate physical scattering mechanism in the shell. Yen has called these separated components wave packets, each having distinctive time-scale dimensions that derive from the physical process that generates it. Because each wave packet has distinctive time-scale domain properties and the wavelet transform is invertible, the wavelet transform can be used to isolate and synthesize a set of wave packets. The resulting wave packets are functions of aspect and time. Greater efficiency of representation can be achieved by representing these wave packets with a bilinear expansion. Wave packets generated with the wavelet transform can be used for matched filter detection. [Work supported by the Office of Naval Research.]

9:50–10:05 Break

10:05

4aSP4. A new wavelet basis for extracting periodic acoustic signals in noise. Antonio J. Miller and Karl M. Reichard (Grad. Prog. in Acoust., P.O. Box 30, State College, PA 16804-0030)

A new wavelet basis is constructed to extract periodic acoustic signals in noise. A wavelet based comb filter is constructed from the linear superposition of narrow-band, harmonically related wavelet components. The best choice of wavelet components has proven to be a hybrid design incorporating the flat passband of the Shannon wavelet with the transition band of the Morlet wavelet. There are several benefits to taking a wavelet approach to comb filtering when dealing with periodic acoustic signals. Good noise rejection using a finite impulse response filter requires a precise measurement of the signal period and an adaptive sample rate to capture an integral number of samples per period. The wavelet approach relaxes the perfect frequency localization requirement and gives the comb filter ''teeth'' a small, but finite passband. This passband allows for measurement error in the signal period and for the signal to be sampled at an arbitrary rate. The new approach has proved to be an attractive alternative to the time domain averaging techniques used in accessing mechanical fault conditions from vibration measurements of rotating machines.

10:20

4aSP5. Wavelet parameters for speech synthesis. Brian C. Tuttle and Claus P. Janota (Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

A standard method of analyzing human speech is to divide it into its constituent phonemes. Synthesizing speech from these phonemes without including the effects of coarticulation, however, can result in discontinuous and sometimes unintelligible sounds. A recently proposed model for the analysis of speech coarticulation uses wavelet system characterization where wavelet transforms describe the time-frequency behavior of a signal's transmission channel. The objective of this research is to verify the proposed model by synthesizing the original speech from its analysis. The coarticulated speech in question is the consonant–vowel–consonant combination that occurs in words such as ''deed'' or ''bib'' where the two consonants are the same. These words along with their vowels spoken in isolation are recorded digitally and processed using the wavelet system model. The result is an analysis of the coarticulated word with respect to its vowel. The analysis shows in time and frequency how the vowel changes when spoken in context. Synthesizing the speech using the inverse process serves to verify the model. Comparing the result to the original speech reveals that the accuracy is determined mainly by the scale resolution of the wavelet transforms.

10:35

4aSP6. Application of the discrete wavelet transform (DWT) to the enhancement of pulmonary breath sounds. J. F. Forren and G. L. Gibian (Planning Systems, Inc., 7923 Jones Branch Dr., McLean, VA 22102, ggibian@plansys.com)

Aural evaluation of pulmonary breath sounds, or auscultation, is a critical part of diagnosis and management in Pulmonary and Critical Care Medicine. However, reliability between examiners within an institution or across institutions is low. A study by Wilkins *et al.* of 277 North American physicians at the American College of Chest Physicians 1988 annual meeting revealed great inconsistencies in interpretation of breaths sounds. Normal breath sounds were identified as normal by only 80% of these physicians. In addition, variable interpretations were provided for patients with inspiratory and expiratory crackles as well as those with stridor. Clinicians had difficulty defining rhonchi and often misclassified them as wheezes. The algorithm, based on nonlinear operations on the DWT coefficients, makes adventitious features of lung sounds easier to hear. The algorithm makes both coarse and fine crackles much more clearly audible by suppressing the breath and other interfering sounds. While the processed stridor sounds less like stridor than before, the processing makes it easier to hear the difference between stridor and wheezes and thus aids in distinguishing vocal cord dysfunction from asthma. [Work supported by N.I.H.]

10:50

4aSP7. Application of adaptive impulsive noise separation to automotive squeak and rattle detection/quantification. Vy Tran, Sheau-Fang Lei, and Keng Hsueh (Ford Motor Co., vtran@ford.com)

Abnormal noises such as road-induced squeaks and rattles (S&R) in automobiles are very undesired and perceived as quality risks to the customers. Subjective evaluation by trained experts is the typical S&R assessment procedure for product development and quality control. S&R noises usually have impulsive characteristics (time varying, and transient), and are very difficult to be detected and quantified objectively using traditional signal processing techniques. Emerging wavelet signal processing technique and kurtosis impulsive criteria were used to develop an objective impulsive noise detection and separation technique (US Patent Filed). The method has been shown to work successfully with S&R samples in various frequency bands. The technique has been implemented in a portable PC to acquire or operate on any pre-recorded acoustic signal. The system then automatically detects the presence of any S&R, and separates them from the original composite signal. The results can be played back for listening or processed to characterize, and quantify the S&R noises. These new impulsive noise detection and separation tools have been applied to S&R assessment at the vehicle level and subsystem durability tests. Applications to vibration problems have been envisioned. Conceptually, this technique may also have potential signal enhancement applications in the telecommunication field.

11:05–11:35

Panel Discussion

Why Wavelet Transform Domain Processing?

Session 4aUW

Underwater Acoustics, Acoustical Oceanography and Animal Bioacoustics: The Effect of Man-Made Sound on Marine Mammals I

James F. Lynch, Chair
Woods Hole Oceanographic Institution, 203 Bigelow Building, Woods Hole, Massachusetts 02543

Chair's Introduction—8:00

Invited Papers

8:05

4aUW1. An acoustic integration model (AIM) for assessing the impact of underwater noise on marine wildlife. William T. Ellison (Marine Acoust., Inc., P.O. Box 340, Litchfield, CT 06759), Karen Weixel (Marine Acoust., Inc., Middletown, RI 02842), and Christopher W. Clark (Cornell Univ., Ithaca, NY 14850)

In recent years there has been a heightened awareness of the environmental impact of noise, especially man-made noise, on marine wildlife. The National Environmental Policy Act (NEPA), Executive Order 12114, The Endangered Species Act, The Marine Mammal Protection Act, and the Coastal Zone Management Act each provide for varying levels of regulation and control in protection of the environment and marine wildlife. In order to assess the environmental impact of a sound source, one must predict the sound levels that any given species will be exposed to over time in the locale of the source's radiated field. This is a three-part process involving (1) the ability to measure or predict an animal's location in space in time, (2) the ability to measure or predict the sound field at these times and locations, and finally, (3) integration of these two data sets so as to determine the net acoustic impact of the sound source on any specific animal. This paper describes a modeling methodology for accomplishing this task. Model inputs required to specify the acoustic environment, animal distribution and behavior, and sound source characteristics are discussed in detail. The AIM model output capabilities are described together with topical examples.

8:35

4aUW2. Masked temporary threshold shift (MTTS): A relevant measure for hearing shifts in open waters. Sam H. Ridgway, Donald A. Carder, James J. Finneran (SPAWARSYSCEN SAN DIEGO, Div. D35, 49620 Beluga Rd., San Diego, CA 92152-6266; ridgway@spawar.navy.mil), and Carolyn E. Schlundt (Sci. Applications Int'l. Corp., 3990 Old Town Ave., Ste. 105A, San Diego, CA 92110)

Cetaceans use sound in foraging, communication, and navigation. There is growing concern that human-made sound is a potentially serious auditory problem for cetaceans. Available data were inadequate to allow confident predictions regarding the levels of sound that should be of concern. An accepted method of determining the level of sound that produces a temporary reduction in the ears ability to respond fully is called temporary threshold shift (TTS). In the open waters of San Diego Bay, masked hearing thresholds were estimated for dolphins, *Tursiops truncatus*, and white whales, *Delphinapterus leucas*, before and after exposure to louder 1-s tones at five sonar frequencies. Masking noise was used to create a floor effect, thereby eliminating threshold variability in the ever-changing ambient noise environment in the natural waters where tests were conducted. The criterion for masked TTS (MTTS) was a 6-dB increase in threshold over baseline. Subjects exhibited alterations in their behavior starting at levels around 180 dB, and experienced MTTS at levels between 192-202 dB. Maximum levels of sound exposure without risk of hearing damage can be determined with the MTTS approach. More research employing different sound sources is needed. [Supported by Office of Naval Research.]

Contributed Paper

9:05

4aUW3. Characteristics of a set of noises used to test effects of underwater explosions on sea mammal hearing. Joseph A. Clark, Paul M. Moore (CDNSWC, Code 734, Bethesda, MD 20084), Jane A. Young (CDNSWC, Bethesda, MD 20084), and Joel B. Gaspin (IHNSWC, Indian Head, MD 20640)

As part of a study of temporary threshold shifts (TTS) expected to occur in sea mammals near underwater explosions, a set of ten explosion-like noises were synthesized [J. A. Clark, J. A. Young, and J. B. Gaspin, J. Acoust. Soc. Am. **105**, 1048 (1999)]. The ten transient noises differed principally in maximum peak-to-peak pressure and were used in the TTS study as an ascending series of levels presented to two beluga whales and two bottlenose dolphins. The set of levels were selected from a larger set of predicted signatures of underwater explosions corresponding to a variety of charge weights and ranges in an underwater environment of interest which was characterized by charge depth, receiver depth, water depth, and sound-speed profile. This talk describes the ten transients used for the TTS studies. Time histories, frequency spectra, time-frequency distributions, and a variety of characteristics derived from these graphical descriptions will be presented.

Invited Paper

9:20

4aUW4. Temporary threshold shift in pinnipeds induced by octave-band noise in water. David Kastak, Brandon L. Southall, Ronald J. Schusterman, and Colleen J. Reichmuth (Long Marine Lab., Univ. of Calif., 100 Shaffer Rd., Santa Cruz, CA 95060)

Low-frequency noise from manmade sources represents an increasing portion of the total noise in the ocean. Such noise may adversely impact diving pinnipeds, which hear relatively well at low frequencies. The effects of octave bands of moderately intense noise (under 4 kHz, 60–76 dB SL, 20–22 min duration) were examined in three species of pinniped, the California sea lion (*Zalophus californianus*), harbor seal (*Phoca vitulina*), and northern elephant seal (*Mirounga angustirostris*). Auditory sensitivity data were obtained behaviorally, using a go/no-go psychophysical procedure. Thresholds were determined before noise exposure, immediately after noise exposure, and again after 24 h. Mean threshold shifts ranged from 2.9–4.9 dB at the center frequency of the noise band. Recovery was complete after 24 h. These results show that noise levels as low as 60 dB SL can induce TTS. However, between 60 and 76 dB SL, there was no relationship between noise level and degree of TTS. Additionally, maximum hearing loss occurred at center frequency rather than one-half octave above the center frequency. Further testing at longer exposure durations and higher noise levels is needed to more completely assess the degree to which pinniped underwater hearing is affected by continuous noise exposure.

9:50–10:05 Break

Contributed Paper

10:05

4aUW5. A software package calculating zones of impact on marine mammals around industrial noise sources. Christine Erbe and David M. Farmer (IOS-Ocean Acoust., 9860 W. Saanich Rd., Sidney, BC V8L 4B2, Canada, erbec@dfo-mpo.gc.ca)

A software package is presented which estimates zones of interference around underwater noise sources affecting marine mammals. An ocean sound propagation model based on ray theory computes the spreading of complex underwater sound such as broadband animal vocalizations and manmade noise. On a grid of receiver locations (representing the affected marine mammal), the received signal and noise sound spectra are compared. Given a species-specific audiogram, the software package plots zones of audibility around the noise source. Given species-specific vocalizations, zones of masking are plotted based on results obtained during an earlier study which measured masked hearing thresholds of a beluga whale. Tools developed during this study (such as an artificial neural network model to predict the amount of masking) can be linked to the software package. Zones of behavioral disturbance are plotted based on received sound levels reported in the literature. Zones of discomfort, injury, and hearing loss could be plotted if thresholds were known or using current estimates. The software package is applicable to a variety of ocean environments requiring location-specific oceanographic input data. The case of icebreakers affecting beluga whales in the Beaufort Sea is demonstrated.

Invited Paper

10:20

4aUW6. Temporary threshold shift in hearing induced by an octave band of continuous noise in the bottlenose dolphin. Whitlow W. L. Au, Paul E. Nachtigall, and Jeffery L. Pawloski (Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734)

Temporary threshold shift in hearing at 7.5 kHz was studied with an Atlantic bottlenose dolphin. Immediately following a threshold measurement, the animal was required to station in a hoop and be exposed to an octave band of continuous noise from 5 to 10 kHz. Noise exposure sessions lasted about 50 min, with the requirement that the animal spend a total of 30 min in the hoop. The dolphin also had two preferred locations both about a meter to the side of the hoop, one at the surface, and the other at the hoop depth. The noise levels at the hoop and to the side were about the same but with different spectra. The noise at the surface was about 3-dB lower. After exposure to the fatiguing stimulus, the animal's hearing sensitivity was immediately measured. The animal's hearing was not affected when the noise was 171 dB at 1 μPa with a total energy flux density of 205 dB at 1 μPa^2 s. Temporary threshold shifts of 12–18 dB were obtained when the noise increased to 179 dB with an energy flux density of 213 dB or 1330 J/m^2. The fatiguing stimulus was about 96 dB above the animal's pure tone threshold of 84 dB. [Work supported by ONR.]

10:50

4aUW7. Masked temporary threshold shift for impulsive sounds in dolphins and white whales. Donald A. Carder, James J. Finneran, Sam H. Ridgway (SPAWARSYSCEN San Diego, Div. D35, 49620 Beluga Rd., San Diego, CA 92152-6266), and Carolyn E. Schlundt (Sci. Applications Intl. Corp., San Diego, CA 92110)

In an effort to develop acoustic safety criteria for marine mammals that may be exposed to impulsive sounds a masked temporary threshold shift (MTTS) experimental approach was used. Two dolphins, *Tursiops truncatus*, and two white whales, *Delphinapterus leucas*, were systematically exposed to increasing levels of impulsive sound (S-1) under water, then tested to threshold at three pure-tone frequencies (S-2) under water in the presence of controlled masking noise. One hearing frequency tested was at the pulse peak frequency, with the other two S-2 test tones at higher frequencies. Behavioral responses were monitored and hearing thresholds were ascertained to determine if criterion (6-dB increase over baseline) was reached. Thresholds were estimated using the free response paradigm method and an up–down staircase psychophysical procedure for stimulus presentation in open water in San Diego Bay. These data suggest that the rapid rise time of the leading edge in 1-s tones used in a previous study is not responsible for producing MTTS. Results of the present study add to the existing knowledge of MTTS for marine mammals and provide the first data on MTTS in cetaceans for impulsive sounds. [Work supported by the Office of Naval Research.]

11:05

4aUW8. The ASA standards program and underwater sound, are they incompatible? Daniel L. Johnson (Brüel Bertrand & Johnson Acoustics, 4719 Mile High Dr., Provo, UT 84604)

In the ASA National Standards Catalog, there are over 70 standards that concern the measurement and evaluation of sound in air. There are only a few standards, such as ANSI S1.11-1986 (filters) or ANSI S1.22-1992 (underwater transducer calibration) that concern underwater sound. This latter standard is the only standard of the 111 current standards that specifically relates to underwater acoustics. Standards such as ANSI S1.26-1995 (absorption of sound by the atmosphere) or ANSI S12.9-1996 (measurement of occupational noise exposure) just do not have a counterpart for water. Is this because there is no need? Is there no value in standardizing some common practices for underwater sound measurement and evaluation? Is it indeed true that underwater sound needs few standards? Comments from the audience are expected.

11:20–11:50

Panel Discussion

Meeting of the Standards Committee Plenary Group

ORGANIZATION OF STANDARDS COMMITTEE PLENARY GROUP MEETING

- **S1 ACOUSTICS**—U.S. Technical Advisory Group (TAG) for IEC/TC 29 Electroacoustics and ISO/TC 43 Acoustics
- **S2 MECHANICAL VIBRATION AND SHOCK—**U.S. Technical Advisory Group (TAG) for ISO/TC108 Mechanical Vibration and Shock
- **S3 BIOACOUSTICS—**U.S. Technical Advisory Group (TAG) for ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock
- **S12 NOISE—**U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC1 Noise

The meeting of the Standards Committee Plenary Group will precede the meetings of the Accredited Standards Committees S1, S3, and S12, to take place in the following sequence on the same day:

S12	9:45 a.m. to 11:00 a.m.	Union B Room
S1	2:00 p.m. to 3:30 p.m.	Union B Room
S3	3:45 p.m. to 4:45 p.m.	Union B Room

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees, plus a review of the international standardization (U.S. TAG) activities including reports on recent meetings and planning for forthcoming meetings.

Members of S2 on Mechanical Vibration and Shock (and U.S. TAG for ISO/TC 108 and five of its Subcommittees, SC1, SC2, SC3, SC5 and SC6) are also encouraged to attend the Standards Committee Plenary Group meeting, even though the S2 meeting will take place one day earlier, on Wednesday, 3 November 1999 at 9:00 a.m.

The U. S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3 and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. TAG
ISO		
P. D. Shomer, Chair H. E. von Gierke, Vice Chair	**ISO/TC 43** Acoustics	S1 and S3
P. D. Shomer, Chair H. E. von Gierke, Vice Chair	**ISO/TC 43/SC1** Noise	S12
D. Reynolds, Chair	**ISO/TC 108/SC4** Human Exposure to Mechanical Vibration and Shock	S3
D. J. Evans, Chair	**ISO/TC 108** Mechanical Vibration and Shock	S2
R. H. Mehta, Chair K. Won, Vice Chair	**ISO/TC 108/SC1** Balancing, including Balancing Machines	S2
A. F. Kilcullen, Chair	**ISO/TC 108/SC2** Measurement and Evaluation of Mechanical Vibration and Shock as Applied to Machines, Vehicles and Structures	S2
D. J. Evans, Chair	**ISO/TC 108/SC3** Use and Calibration of Vibration and Shock Measuring Instruments	S2
D. J. Vendittis, Chair R. F. Taddeo, Vice Chair	**ISO/TC 108/SC5** Condition Monitoring and Diagnostics of Machines	S2
G. Booth, Chair	**ISO/TC 108/SC6** Vibration and Shock Generating Systems	S2
IEC		
V. Nedzelnitsky, U.S. TA	**IEC/TC 29** Electroacoustics	S1 and S3

Meeting of Accredited Standards Committee (ASC) S12 on Noise

P. D. Schomer, Chair S12, and Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC1, Noise
2117 Robert Drive, Champaign, Illinois 61821

R. W. Hellweg, Vice Chair, S12
Compaq Computer Corporation, MS PK02-1/J60, Maynard, Massachusetts 01754

H. E. von Gierke, Vice Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC1, Noise
1325 Meadow Lane, Yellow Springs, Ohio 45387

Accredited Standards Committee S12 on Noise. Working group chairs will report on their progress for the production of noise standards.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control; including biological safety, tolerance and comfort and physical acoustics as related to environmental and occupational noise.

THURSDAY AFTERNOON, 4 NOVEMBER 1999 HAYES ROOM, 1:00 TO 3:00 P.M.

Session 4pAA

Architectural Acoustics and Musical Acoustics: Archeological Acoustics II

David Lubman, Chair
David Lubman & Associates, 14301 Middletown Lane, Westminster, California 92683

Invited Papers

1:00

4pAA1. Acoustics of ancient Chinese bells. Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115)

Many ancient Chinese bells, some more than 3000 years old, remain from the time of the Shang and Zhou dynasties. Most of these bells, being oval or almond shaped, sound two distinctly different musical tones, depending upon where they are struck. Studies of these bells have provided us with much knowledge about the musical culture of Bronze age China. Results are reported from our investigations of original bells in the Shanghai Museum and the Sackler Gallery at the Smithsonian Institution, as well as replica bells in our own laboratory. Modes of vibration in original bells, obtained by experimental modal testing in the museum, compare very well with holographic interferograms of replica bells. Our results are compared with our catalogue of data on ancient Chinese bells gathered by other investigators. Although acoustical studies have revealed a great deal about ancient Chinese music and casting practices, several questions still remain unanswered.

1:30

4pAA2. Acoustics of Karen bronze drums. Laura M. Nickerson and Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115)

Bronze drums are important to the culture of the Karen people, who live mainly in Burma and the mountainous region between Burma and Thailand. A Karen bronze drum, which is cast in one piece except for the animal adornments, consists of an overlapping tympanum that may range from 9–30 in. in diameter and a cylinder that is slightly longer than the diameter. Bronze drums have a wide variety of ritual use, both musical and nonmusical. The sound spectrum of a drum shows a dense collection of partials out to about 3 kHz. The corresponding modes of vibration, recorded using electronic TV holography, sometimes show the strongest vibration in the tympanum and sometimes in the cylindrical shell. The sound of the drum and the manner of playing will be illustrated by videotape.

4pAA3. The carnyx, the lur, and other ancient European horns. D. Murray Campbell (Dept. of Phys. and Astron., Univ. of Edinburgh, Edinburgh EH9 3JZ, UK, D.M.Campbell@ed.ac.uk)

Modern European brass instruments are the result of a long period of evolution. Relatively little is known about their ancestors, the lip-excited horns which were widely used in Europe several thousand years ago. Various attempts have been made to reconstruct playable versions of such instruments as the Scandinavian lur and the Celtic carnyx; although there is a considerable speculative element in all of these reconstructions, they offer the intriguing possibility of carrying out both musical and acoustical tests on these long-vanished instruments. In the last few years a project funded by the National Museums of Scotland has resulted in the reconstruction of two specimens of a carnyx, based on fragments found in Deskford, in Scotland, in the nineteenth century. Only the bell section, in the shape of a boar's head, had survived, but from other pictorial evidence it is known that the instrument had a tube about 2 m long, and was held vertically. This paper reports on a series of acoustical tests which have been carried out on the two reconstructions, and compares the acoustical and musical properties of the carnyx with those of reconstructed lurs and other ancient horns. [Work supported by EPSRC(UK).]

THURSDAY AFTERNOON, 4 NOVEMBER 1999 KNOX ROOM, 2:00 TO 3:30 P.M.

Session 4pBB

Biomedical Ultrasound/Bioresponse to Vibration: Nonlinear Acoustics in Medicine

Robin O. Cleveland, Chair

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

Invited Papers

2:00

4pBB1. Full wave simulations of pulsed focused fields in a nonlinear tissue mimic. Thomas L. Szabo, Frances Clougherty (Hewlett Packard, MS-095, 3000 Minuteman Rd., Andover, MA 01810), Gregory L. Wojcik, John C. Mould, and Laura M. Carcione (Weidlinger Assoc., Los Altos, CA 94022)

In order to simulate focused ultrasound fields in nonlinear tissue, a full wave pseudospectral finite difference model has been developed. Unlike models based on the KZK equation, this model can predict wide-angle propagation effects as well as tissue backscatter, essential for pulse-echo medical imaging. This simulation program is capable of including broadband pulse propagation, causal attenuation, nonlinearity (B/A), and inhomogeneities. Harmonics beyond the solver's Nyquist limit are removed numerically and quantified. To facilitate comparison to data and to expedite calculation, the model has been adapted for circularly symmetric transducers. As input to the program, waveforms are taken from hydrophone scans close to the faces of 2.25- and 5-MHz focusing transducers. Simulations are compared to data taken in water and in front of and behind a tissue mimicking cylinders of tofu, which are sliced repeatedly to give transmitted and reflected pressure data versus thickness. Tissue inhomogeneities are simulated by voids in the tofu. Acoustic properties of the mimic are measured separately. B/A is extracted numerically from observed harmonic generation versus input pressure and range.

Contributed Papers

2:30

4pBB2. The acoustic field of a clinical pulsed Doppler ultrasound system near lung. Christy K. Holland and Jeffrey T. Flament (Dept. of Radiol., M.L. 0742, Univ. of Cincinnati, 234 Goodman St., Cincinnati, OH 45219-2316, Christy.Holland@uc.edu)

Several animal models have exhibited thresholds for petechial hemorrhage in lung exposed to clinical levels of diagnostic ultrasound. Extravasation of red blood cells into the alveolar spaces has occurred consistently near the visceral pleura (at the outer surface of the lung). In order to quantify the ultrasound exposure conditions under which damage occurs, the pressure field of an ATL HDI 3000 in Doppler mode was measured near *ex vivo*, aerated rat lung in a water bath. The pressure output of the ATL L10-5 linear array in pulsed Doppler mode (6.0 MHz center frequency) was interrogated near the lung with a Sonic Technologies 0.4-mm bilaminar membrane-type PVDF hydrophone. A Velmex 3-axis translation stage was utilized for automated positioning of the hydrophone. Virtual instruments were developed using LABVIEW v.5.0.1 on a Macintosh 8100 to sample the pressure waveforms from a LeCroy 9350CL digital oscilloscope via an IEEE 488 interface and National Instruments data acquisition boards. Images of the pressure field in the axial, transverse, and elevational planes near the surface of the lung will be compared to the same measurements in the free field (without the lung present).

2:45

4pBB3. Path-integrated Goldberg number as a predictor of enhanced heating from finite-amplitude therapeutic ultrasound sources. Ibrahim M. Hallaj, Kullervo Hynynen (Brigham and Women's Hospital, Harvard Med. School, 221 Longwood Ave., Boston, MA 02115), and Robin O. Cleveland (Boston Univ., Boston, MA 02215)

Several theoretical and experimental studies have demonstrated that the higher-frequency harmonics generated by finite-amplitude propagation from therapeutic ultrasound sources result in enhanced heating of tissue. However, published results vary in their assessments of the significance of the nonlinearity on tissue heating. The variations in the results are due to the use of different sources and propagation paths. It would be useful to have an easily computed estimator of finite-amplitude effects for therapeutic ultrasound sources. The present study describes a dimensionless quantity, which takes the propagation path into account, to estimate the excess temperature rise due to nonlinearity. The dimensionless parameter is based on the Goldberg number which is a measure of the importance of nonlin-

ear effects to absorption effects. The relationship between the enhanced heating of tissue and the path-integrated Goldberg number for water-tissue paths is presented for typical therapeutic devices. [Work sponsored by National Cancer Institute.]

3:00

4pBB4. Holographic interferometric visualization of weak shock waves in castor oil packed with equal-diameter gelatine spheres. S. Hamid R. Hosseini (Shock Wave Res. Ctr., Inst. of Fluid Sci., Tohoku Univ., 2-1-1 Katahira, Aoba, Sendai, 980-8577 Japan, hosseini@ceres.ifs.tohoku.ac.jp), S. Moosavi Nejad (Tohoku Univ., Sendai, Japan), Tasneem B. F. R. Nahaboo, and Kazuyoshi Takayama (Tohoku Univ., Sendai, Japan)

The attenuation of shock waves in castor oil in which equal-diameter gelatine spheres were packed was measured with hydrophones at various stand-off distances. The motion of shock waves was quantitatively visualized by using double exposure holographic interferometry. Diameters of gelatine spheres under study were 3.0 mm $\pm1\%$, 4.5 mm $\pm1\%$, and 6.0 mm $\pm1\%$. Shock waves were produced at the center of a chamber in which gelatine spheres of total 4 L in volume were densely immersed in castor oil. In order to generate shock waves, irradiation of a pulsed YAG laser beam on silver azide pellets were used. The weight of silver azide pellets ranged from 3 μg to 10 mg, with their corresponding energy of 4.5 mJ to 15 J. The effect of the nonuniformity on the shock-wave attenuation created due to the multiple interaction of shock waves with the gelatine spheres was experimentally clarified. The result will be useful to establish a model of human tissue in numerically simulating the complex interaction of underwater shock waves with human tissue which takes place in shock-wave therapies.

3:15

4pBB5. Ultrasound field correction in trans-skull ultrasound surgery using a focused array. Gregory Clement, Jie Sun, Tonia Giesecke, and Kullervo Hynynen (Brigham & Women's Hospital and Harvard Med. School)

Correction of the distortion due to diffraction, reflection, and attenuation in trans-skull ultrasound propagation is investigated. Human skull fragments are positioned between an underwater focused transducer array and a PVDF needle hydrophone. A positioning system is used to scan the hydrophone along a plane normal to the transducer face to provide amplitude and phase delay information as a function of position. The hydrophone measurements are then used to manually correct the driving signal phase of each element in the array. The field is remeasured and the correction evaluated in terms of the obtained focus and the occurrence of grating lobes. The measurements are repeated at different orientations about each skull sample. Finally, skull information is obtained using MRI and CT scans. Variation of the measured ultrasound field as a function of skull thickness, density, and transducer location is examined.

THURSDAY AFTERNOON, 4 NOVEMBER 1999 HARRISON ROOM, 1:30 TO 5:00 P.M.

Session 4pEA

Engineering Acoustics: Transducer Design and Calibration

R. Daniel Costley, Jr., Chair
National Center for Physical Acoustics, MILTEC, P.O. Box 878, University, Mississippi 38677

Contributed Papers

1:30

4pEA1. Application of differential-capacitance accelerometers to underwater acoustic velocity sensing. Thomas B. Gabrielson (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804, tbg3@psu.edu)

The differential-capacitance accelerometer is relatively common in medium- to high-grade commercial accelerometers but has been largely ignored for application to underwater acoustic velocity or acceleration sensors. The differential-capacitance configuration is capable of good long-term stability and exceptional low-frequency noise performance. The effective sensor impedance is determined not by the signal frequency but by the ac bias frequency. The low sensor impedance associated with high-frequency ac bias relaxes the requirements on signal cables. In addition, the critical preamplifier characteristics are those at the ac bias frequency so low-frequency $(1/f)$ noise can be avoided. The differential-capacitance accelerometer does not have an inherent roll-off in response at low frequency so these devices can be used in self-orienting three-axis sensors for deployable systems. In machinery monitoring applications, synchronous detection of the ac-modulated signal yields a high degree of immunity to external interference and is amenable to straightforward multiplexing and optical or rf transmission. This paper summarizes the design principles associated with the differential-capacitance accelerometer through two illustrations: modification of a low-grade commercial product for improved signal-to-noise performance and multisensor multiplexing in the context of acoustic velocity sensing. [Work supported by the Office of Naval Research.]

1:45

4pEA2. Generalized state-variable model for the design of a closed-loop differential capacitive accelerometer. Barry J. Doust, Thomas B. Gabrielson (Grad. Prog. in Acoust. and the Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804), and Jeffrey L. Schiano (The Penn State Univ., University Park, PA 16802)

In an effort to develop a high-performance, miniature, directional acoustic sensor, the advantages of closed-loop operation are investigated. A nonlinear state-variable model of a generalized sensor system is presented. The sensor model is differential capacitive with electrostatic actuation for force rebalance of the proof mass. The physical model parameters are chosen to replicate scales and tolerances of typical MEMS devices with particular attention to nonlinearities associated with the electrostatic force and large deflections. Modeling and simulation will be presented for various control schemes and results will be compared in terms of important operational criteria such as dynamic range, bandwidth, linearity, and noise. [Work supported by the ONR AASERT program.]

2:00

4pEA3. Fiber-optic flexural-disk accelerometer. Andrew J. Doller, Karl M. Reichard, and Steven L. Garrett (Grad. Prog. in Acoust. and Appl. Res., The Penn State Univ., P.O. Box 30, University Park, PA 16804, adoller@psu.edu)

A fiber-optic, interferometric accelerometer incorporates a centrally supported flexural disk [U.S. Patent No. 5,368,485 (1994)] to convert vibration into differential strains on two spiral coils of optical fiber bonded to the disk's top and bottom surfaces. The sensor coils form two legs of a

Mach–Zender interferometer, terminated in a 3×3 coupler, whose outputs were used in a passive homodyne symmetric signal demodulator [U.S. Patent No. 5,313,266 (1994)]. A new digital implementation of this demodulation algorithm used fast fourier transforms to perform the necessary algebraic operations on time-windowed data segments, eliminating drifts. The sensor/demodulator system bandwidth was measured from 42 Hz to 5 kHz. A minimum detectable signal of 7 mrad was obtained with a 10-dB signal-to-noise ratio. At the lowest frequency, this corresponded to a dynamic range of 105 dB, which is 9 dB greater than the theoretical maximum for the 16-bit digitizer, due to interferometric conversion of amplitude modulation into frequency modulation. Thick-plate elastic models agreed with measured disk resonance frequencies. The measured sensitivity of 17 rad/g below resonance was less than the theoretical value of 23 rad/g, probably due to imperfect strain transfer through the epoxy adhesive used to bond the fibers to the disk's surface. [Work supported by a grant from the Office of Naval Research.]

2:15

4pEA4. Minimum point search method for improvements in absolute calibration of accelerometers at high frequencies using the fringe-disappearance method. Lixue Wu (Acoust. Standards, Inst. for Natl. Measurement Standards, Natl. Res. Council Canada, Ottawa, ON K1A 0R6, Canada)

The measurement uncertainty in the absolute calibration of accelerometers at high frequencies using the fringe-disappearance method was reduced by a minimum point search method. Outputs of a bandpass filtered photodetector and an accelerometer were simultaneously recorded by two true rms digital voltmeters. Displacement amplitudes were finely controlled by a precision attenuator. Minimum points of the output of the photodetector and the corresponding sensitivities of the accelerometer were found from the recorded data. During data processing, a simple model for the minimum points was proposed and a least mean squares algorithm was used to estimate parameters of the model. Results of calibrations were compared with these obtained using fringe-counting method at lower frequencies. It was found that the minimum point search method could compare with the fringe-counting method in measurement uncertainties. Good measurement reproducibility was also observed that resulted in the reduction of total measurement uncertainty.

2:30

4pEA5. Alternating-current calibration of solid-state gyroscopes by reciprocity. Thomas B. Gabrielson (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804, tbg3@psu.edu)

One of the hot topics in MEMS research is the development of miniature solid-state angular-rate sensors (gyroscopes). These devices have a number of potential applications including navigation systems for divers and unmanned vehicles, interpolators for periods of GPS loss of signal, and automobile skid sensors. For many of these applications, constant-rate (dc) calibrations are not representative of the expected excitations. Fortunately, high-quality ac-rate calibration can be done with an inexpensive apparatus by means of reciprocity. A suspended platform is driven in torsional oscillation by a pair of geophones acting as a force couple. Another pair of geophones is used as an angular velocity sensor. Since both of these transducer pairs are reciprocal, they can be used in conjunction with the device under test to perform a complete reciprocity calibration. The platform produces very smooth ac-rate motion from 1 to 20 Hz at rates of below 0.1 deg (rms)/h to 200 deg/h. The transfer impedance used in the reciprocity calibration is determined by the polar moment of inertia of the platform. This can be determined accurately by adjusting the separation of a symmetrically placed pair of masses and measuring the change in natural frequency. [Work supported by the Office of Naval Research.]

2:45

4pEA6. Thin, low-frequency sound projectors for use in shallow water. James F. Tressler and Thomas R. Howarth (Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375-5350, tressler@acoustics.nrl.navy.mil)

A thin, low-frequency acoustic projector is being developed at the Naval Research Laboratory for shallow water applications. The active panel consists of cymbal-type flextensional driver elements arranged individually behind an acoustic radiating plate. Variants of two panel designs have been calibrated underwater. The acoustic projector sandwich is nominally 6.35 mm thick and is intended to operate between 1 and 10 kHz. The calibration results that will be presented (TVR, SPL, directivity response) will focus primarily on a panel consisting of 12.7-mm-diam cymbal elements arranged in an 8×8 square configuration. [This research is being supported by Code 321SS of the Office of Naval Research.]

3:00–3:15 Break

3:15

4pEA7. Reduction of measurement variation: Small acoustic chamber screening of hard disk drives. Tami Ogle (Western Digital Corp., 822 Creek Dr., San Jose, CA 95125) and Lee Hornberger (Santa Clara Univ.)

Several small acoustic chambers on the manufacturing floor are used at Western Digital to screen out unacceptably loud disk drives. The screening test initially had large variation which made its effectiveness questionable. It was unclear whether this variation was coming from the drives or the chambers. In this study statistical methods were used to determine the source of the variation, and a design of experiments approach was used to determine the factors which significantly contributed to the variation. It was determined that the measurement chambers, not the drives, were the source of the large variation in testing. Additionally, the significant factors affecting the screening test were found to be the microphone calibration and microphone height inside the chamber.

3:30

4pEA8. A low-cost sound level meter based on personal computer. Guillermo de Arcas, Juan M. Lopez (Dept. of S.E.C., E.U.I.T. Telecomunicacion, Universidad Politecnica de Madrid, Crta. Valencia km 7, Madrid 28031, Spain, garcas@sec.upm.es), Manuel Recuero, and Alberto Martin (I.N.S.I.A., Madrid 28031, Spain)

In this abstract a low-cost sound level meter is developed with the help of digital signal processing techniques and an integrated programming environment. The instrument is based on the use of a general purpose data acquisition PC card to acquire the signal and a powerful digital signal processing algorithm based on digital filters to compute a one-third octave band analysis conforming to ANSI S1.11-1986 in real time. The objective is made possible thanks to the use of optimized signal processing routines from Intel Signal Processing Library. The use of this library and an instrumentation specific integrated programming environment such as LabWindows/CVI (National Instruments) makes it possible to design a low-cost instrument in a very short time. The algorithm is a multirate filter bank implementation of the standard. The discussion is completed with several simulations and benchmarks are presented for different design situations. Execution time is also compared to that obtained with a popular digital signal processor (TMS820C30 from Texas Instruments). [For Engineering Acoustics Best Student Paper Award.]

3:45

4pEA9. Transducer elctronic data sheet implementation within traditional precision microphones and array microphone systems. Michael J. Lally, Mike J. Dillon (The Modal Shop, Cincinnati, OH 45212), Gunnar Rasmussen, and Peter Wulf Andersen (The Modal Shop, Cincinnati, OH 45212 and GRAS Sound and Vib., Denmark)

Consistent with the IEEE1451 standard development of smart sensors, microphones have been developed that can digitally report critical transducer characteristics. This new, standardized communication ability automates communication of calibration scaling and eliminates potential bookkeeping errors in large channel testing. Discussion includes the framework of the evolving standard, implementation in preamps/array microphone systems and practical examples of error reduction and time savings.

4:00

4pEA10. Problems associated with driving an efficient Tonpilz transducer with broadband signals. Alan H. Lubell (Lubell Labs., Inc., 21 N. Stanwood Rd., Columbus, OH 43209)

A symmetrical piezoceramic-stack-driven Tonpilz makes a very efficient underwater loudspeaker for use in any body of water where human activity takes place. The large variation of impedance with frequency makes driving this type of transducer a unique problem. Transformer leakage inductance together with stack capacitance causes excessive current to be drawn at the series resonant frequency, which must be controlled with an appropriately sized series resistor. This paper presents measured and computed frequency responses and impedance magnitude versus frequency graphs for several of Lubell Labs.' underwater acoustic transducers. Included are recent calibrations made at the USN Dodge Pond facility. These are used to determine suitable values of leakage inductance and series resistance. A reasonable size series resistance is one that will not cause excessive loss at the fundamental transducer resonance. Excessive leakage inductance must be avoided because of its effect on high frequency response. An expression for the current drawn when the transducer is driven with bandlimited pink noise has been derived by the author and will be given herein.

4:15

4pEA11. Binaural equalization for loudspeaker systems. Peng Wang and Wee Ser (Ctr. for Signal Processing, S2-B4-08, School of EEE, Nanyang Technolog. Univ., Singapore 639798, p145255056@ntu.edu.sg)

On-axis equalization methods for loudspeaker systems can be found everywhere. In these schemes, the axial impulse responses are taken as the objects to be equalized. However, the generated equalizers will display unsatisfactory off-axis distortions and different equalization effects at left and right ears. In this paper, a new criterion, which combines the considerations of the distortions at left and right ears and the binaural difference, is presented. A binaural equalization method applying this criterion is introduced and the comparison with normal on-axis equalization methods is also given.

4:30

4pEA12. A novel dual-band equalizer design for loudspeaker systems. Peng Wang, Wee Ser, and Ming Zhang (Ctr. for Signal Processing, S2-B4-08, School of EEE, Nanyang Technolog. Univ., Singapore 639798, p145255056@ntu.edu.sg)

Many equalization methods based on DSP technology have been proposed during the past decade. However, nearly all the methods demand low-frequency components to be processed separately because of deficient resolution. A dual-band equalization method for loudspeaker systems is presented in this paper. In this method, the conventional equalization based on deconvolution of the system response and a recently introduced method based on warped filter design are combined, to achieve a refined performance in the whole audio frequency region with lower order filter. Computer simulations are given to demonstrate the effectiveness.

4:45

4pEA13. Calibration or conformance check. What is the difference. Ernst Schonthal (The Modal Shop, 1775 Mentor Ave., Cincinnati, OH 45212)

Both measurements are performed using certified equipment. A certificate is issued after both measurements. This paper will address the difference between the two certificates and why calibration is important. Also this paper will explain some instances where one has to do both.

THURSDAY AFTERNOON, 4 NOVEMBER 1999 HAYES ROOM, 3:25 TO 4:30 P.M.

Session 4pID

Interdisciplinary: Distinguished Lecture on Acoustics at the End of the 20th Century—An Overview of the State of the Art

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

Chair's Introduction—3:25

Invited Papers

3:30

4pID1. Acoustics at the end of the 20th Century—An overview of the state of the art. Malcolm J. Crocker[a] (Mech. Eng. Dept., Auburn Univ., Auburn, AL 36849)

Recent years have seen rapid of advances in digital computers, miniaturization of electronics, and the development of new materials. These advances have led to improved knowledge in many scientific fields including a large number of different areas of acoustics. In many cases the developments have been synergistic; new experimental knowledge has led to improved theoretical models and approaches and vice versa. This paper begins with a brief introduction reviewing some of the technological developments that are helping to accelerate the increase in knowledge in different areas of acoustics. It then continues with a general review of the state of the art in a number of specific fields such as: use of finite-element and boundary-element methods in acoustics, computational aeroacoustics, sonochemistry, thermoacoustic engines, active noise and vibration control, sound-intensity measurements, techniques of speech coding and speech recognition, ultrasonics in medical acoustics, and cochlear mechanics. Finally, some projections about future scientific and technological developments in different areas of acoustics are made. [a]The author is the editor-in-chief of the recently published *Encyclopedia of Acoustics*.

THURSDAY AFTERNOON, 4 NOVEMBER 1999 GRANT–HARDING ROOM, 1:30 TO 3:00 P.M.

Session 4pMUa

Musical Acoustics: Music, Rhythm and Development II

Caroline Palmer, Cochair
Department of Psychology, The Ohio State University, 1885 Neil Avenue, Columbus, Ohio 43210

Mari Riess Jones, Cochair
Department of Psychology, The Ohio State University, 1885 Neil Avenue, Columbus, Ohio 43210

Contributed Papers

1:30

4pMUa1. Perception of rhythm by children in multitimbral, multimetric contexts. Punita G. Singh (Sound Sense, 20-A Aurangzeb Rd., New Delhi 110011, India)

Perception of meter by children aged 6 to 12 was studied using sequences of 12 tones as stimuli. Either no accents, or physical accents at positions implying triple, quadruple, or both meters were provided by changing the spectral locus of four harmonics n, $n+1$, $n+2$, $n=3$, of tones L_n (where $n=2$ or 6) or the rise/decay times of their temporal envelopes (95+5 ms, vs 5+95 ms). Listeners reported if they perceived a triple, quadruple, or ambiguous meter, or no accents at all. Children were easily able to perceive the metrical structure of sequences with accents on triple or quadruple meter positions alone. Mixed meters were hard to parse in a single-timbre context. In mixed sequences with accents provided by different timbre features at quadruple and triple meter positions, listeners tended to follow the meter implied by tones L2 rather than L6. Temporal envelope variables were not effective in facilitating parsing of mixed meters. Results indicate that young listeners are similar to adults in using timbral information to parse complex meters. Lower harmonics seemed to be more effective carriers of accents than higher harmonics. Spectral resolving power may thus be an important aspect of not just pitch, but rhythm perception as well.

4p THU. PM

1:45

4pMUa2. Rhythm, expectancy, and time judgments: Some ''surprising'' results. J. Devin McAuley (Dept. of Psych., Ohio State Univ., Columbus, OH 43210, mcauley.11@osu.edu)

A series of time-judgment experiments were performed that manipulated the beginning and ending of a to-be-remembered time interval (the standard) relative to a preceding rhythmic context. Beginning and ending times of the tones marking out the standard were examined for conditions that were on-time, early, and late, relative to the implied beat of the context tones. Consistent with an entrainment model, time judgments about the standard time interval showed a ''shorter'' perceptual bias in early conditions and a ''longer'' perceptual bias in late conditions, and an overall expectancy profile, with better performance observed when the standard was on-time than when it was early or late. The most interesting feature of the data was an unexpected interaction between the beginnings and ending manipulations. Interpreted with respect to the entrainment model, the interaction suggests that a tracking oscillator's phase is more affected by unexpected beginnings, whereas its period is more affected by unexpected endings.

2:00

4pMUa3. Simulating meter perception in acoustic signals. Edward W. Large (Ctr. for Complex Systems, Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33431)

A model of meter perception is presented in which an acoustic signal provides parametric input to a pattern-forming dynamical system. Under rhythmic stimulation, the system undergoes bifurcations that give rise to patterns of self-sustained oscillations. The temporal structure of these patterns reflects the perceived temporal organization of acoustic signals that has been described by both linguists and music theorists [M. Liberman and A. M. Prince, Ling. Inq. **8**, 249–336 (1977); F. Lerdahl and R. Jackendoff, *Generative Theory of Tonal Music* (MIT, Cambridge, 1983)]. These patterns are stable, yet flexible: They can persist in the absence of input and in the face of conflict, yet they can reorganize given a strong indication of a new temporal structure. Both continuous and discrete time models will be discussed and their application to acoustic signals will be demonstrated.

2:15

4pMUa4. Motor learning, meter, and rhythm in music performance. Rosalee K. Meyer (Psych. Dept., Ohio State Univ., Columbus, OH 43210)

What do musicians learn when they perform music: motor-independent knowledge specifying abstract concepts or motor-specific knowledge designating motor movements? Whereas some theories of sequence production suggest that timing is motor independent, others have suggested a closer relationship between timing and motor movements. Previous research suggests that skilled pianists' knowledge of how to perform simple melodies is motor independent [C. Palmer and R. K. Meyer, Psychol. Sci. (in press)]. However, the relationship of motor movements to temporal structures such as meter (periodic accent structure) and rhythm (duration pattern) were not addressed in this study. Two experiments reported here investigate whether skilled pianists' knowledge of meter and rhythm is motor independent or motor specific. In a transfer of learning paradigm, pianists performed a short musical piece under speeded conditions and then transferred to another musical piece with identical pitches but the same or different motor movements. In experiment 1, the transfer melody contained the same or different meter, and in experiment 2, the transfer melody contained the same or different rhythm. Changes in total sequence duration indicate the importance of motor-specific knowledge in the context of music that differs in temporal dimensions such as meter and rhythm. [Work supported by NIMH.]

2:30

4pMUa5. Timing relationships in rhythmic tapping and music performance with delayed auditory feedback. Steven A. Finney (Dept. of Psych., Ohio State Univ., 142 Townsend Hall, 1885 Neil Ave., Columbus, OH 43210, finney.17@osu.edu)

Delayed auditory feedback (DAF) disrupts performance in speech, music, and rhythmic tapping, with a characteristic symptom in all domains being insertions or repetitions of material (resembling stuttering). One recent attempt to explain DAF impairment, the Node Structure Theory [D. G. MacKay, *The Organization of Perception and Action* (Springer, New York, 1987)] has proposed that such repetitions are due to the auditory feedback causing reactivation of the node which produced that feedback, i.e., the sound is a stimulus which precedes (and causes) the repetition. The current research investigated the effects of various auditory delays on tapping simple rhythms, and analyzed the precise temporal relationships between sounds and taps. For the delays causing the most impairment, sound onset was found to follow rather than precede the onset of inserted taps, counter to the requirements of the Node Structure Theory. DAF impairment in a previous experiment in musical performance [S. A. Finney, Music Perception **15**, 153–174 (1977)] was reanalyzed from this perspective, and alternative explanations of the repetition behavior are discussed.

2:45–3:00

Panel Discussion

Session 4pMUb

Musical Acoustics: The Ohio State University Marching Band Performance

James M. Pyne, Chair
The Ohio State University, 316 Weigel, 1866 College Road, Columbus, Ohio 43201

Chair's Introduction—4:25

Session 4pNSa

Noise: Status of Noise Regulations

Bennett Brooks, Chair
Brooks Acoustics Corporation, 27 Hartford Turnpike, Vernon, Connecticut 06066

Invited Paper

1:30

4pNSa1. Local ordinance targeted to low-frequency noise. Bennett Brooks (Brooks Acoustics Corp., 27 Hartford Turnpike, Vernon, CT 06066)

The local authorities of a suburban town wanted to control the low-frequency noise emissions due to musical entertainment at night clubs and taverns. Complaints from residents about booming bass guitar and drum sounds had to be balanced with the need for tavern owners to reasonably conduct business. It was recognized that the existing town code, which placed limits on A-weighted noise levels at property boundaries, was not an effective means to control low-frequency noise emissions. A revised ordinance was developed to address this issue, which now places additional limits on the allowable noise levels in specific octave bands. In particular, this includes the low-frequency (bass) octave bands. The octave band limits were selected such that the summation of band levels equals the A-weighted overall noise limit.

Contributed Paper

1:50

4pNSa2. Prediction and attenuation of noise resulting from construction activities in major cities: The case for modernizing construction noise codes. Daniel R. Raichel (Dept. of Mech. and Aero Eng. and the School of Architecture and Environ. Studies, City College of CUNY, New York, NY 10031) and Michael Dallal (Cooper Union, New York, NY 10003)

Noise arising from operation of construction machinery in major cities is generally most annoying at street and below-ground levels, where the loudest equipment is used. Both residential and business neighborhoods are adversely affected by the presence of excessive noise, particularly if construction activity occurs during early hours. In some cases, vehicular traffic patterns become altered with a resultant change in the traffic noise levels. A survey was conducted as to which type of machinery is most commonly used in New York City, the amount of noise pollution generated by specific equipment, and commercially available means of attenuating the noise, preferably through retrofitting of existing equipment. On-site sound-level measurements were taken and the results compared with published data. Methods of predicting noise levels through the use of computer models, based on the effect of surrounding topology, were also evaluated. Noise codes in several major cities in the United States and abroad were compared with respect to their compliance with U.S. federal regulations. This was done for the purpose of developing a proposal for a new set of construction noise regulations for the City of New York. [Work supported by the New York City Department of Design and Construction.]

2:05

4pNSa3. CNEL at two sites in Del Mar, due to a hundred helicopter flybys. Robert W. Young (1696 Los Altos Rd., San Diego, CA 92109)

In the Settlement Agreement in early 1999 of a lawsuit about helicopter noise, in which the City of Del Mar was one of the plaintiffs and the Marine Corps was one of the Federal defendants, there was a requirement that the community noise equivalent level (CNEL) be included as a standard for measuring noise significance. This is a preliminary report about CNEL measured continually at two sites in south Del Mar, California. In both cases the residents had filed official complaints about the noise of helicopters. These residences are distance from heavily traveled streets and highways. The automatic noise monitor consists of a Computer Engineering, Ltd., 493-2 integrating sound-level meter feeding a CEL-238 secondary processor, all in a weather-proof case. The outfit prints, and puts in memory, the one-hour average sound level during a preceding hour and at the end of a day the 24-hour average sound level (24 HL), the day–night average sound level (DNL), and the CNEL. At 22839 Via Grimaldi, Friday to Wednesday 5–10 May 99, the CNELs were 55.8, 55.5, 54.6, 55.9, 58.7, and 58.7 dB. At 421 Ocean Drive, Tuesday to Saturday 11–15 May 99, the CNELs were 58.4, 56.5, 59.6, 58.6, and 60.5 dB.

2:20

4pNSa4. A radiosity-based model for simulating sound propagation in urban streets. Jian Kang (The Martin Ctr., Cambridge Univ., 6 Chaucer Rd., Cambridge CB2 2EB, UK)

To study the fundamental characteristics of the sound field in urban streets resulting from diffusely reflecting boundaries, a theoretical/computer model has been developed using the radiosity technique. The model divides building facades and ground into a number of patches and then simulates the sound propagation in a street by energy exchange between the patches. Computations in a typical street with a single noise source show that with diffusely reflecting boundaries the sound attenuation along the length is significant, and the sound distribution in a cross section is generally even unless the cross section is very close to the source. The effectiveness of some architectural changes and urban design options on further increasing sound attenuation along the length, such as by strategically arranging buildings in a street or adding absorption patches on facades or ground has been analyzed. It has also been demonstrated that by replacing geometrically reflecting boundaries with diffusely reflecting boundaries, the sound attenuation along the length becomes greater, and the reverberation time becomes shorter. This suggests that from the viewpoint of urban noise reduction, it is better to design the street boundaries as diffusely reflective rather than acoustically smooth. [Work supported by the Lloyd Foundation.]

2:35

4pNSa5. Propagation of noise from petrochemical activities in the Taranaki region of New Zealand. Daryl Prasad (Acoust. Res. Ctr., Architecture Bldg., The Univ. of Auckland, Private Bag 92019, Auckland, NZ, d.prasad@auckland.ac.nz)

Noise is a significant environmental concern facing the petrochemical industry in New Zealand. Typical inland oil and gas exploration involves operations that can run continuously for 70 days. Exposure to the resulting steady and impulsive noise produces undue stress and loss of amenity to neighboring residents. This paper reports on an investigation into the propagation of noise in the Taranaki region of New Zealand. Of the many factors that influence the propagation from drilling, atmospheric conditions (especially wind, temperature, and humidity) create the most variability. This study aims to provide a better understanding into the way noise is propagated in this region. Results of wind and temperature profile measurements will be presented as well as the related frequency and seasonality of events such as temperature inversions. Actual noise propagation data combined with the meteorological data will be used to test a model developed for the prediction of noise levels based on the CONCAWE and ISO 9613-2 methods of calculation.

THURSDAY AFTERNOON, 4 NOVEMBER 1999 — UNION E ROOM, 3:05 TO 4:20 P.M.

Session 4pNSb

Noise: Hearing Protection

Elliott H. Berger, Chair

E-A-R/Aearo Company, 7911 Zionsville Road, Indianapolis, Indiana 46268-1657

Contributed Papers

3:05

4pNSb1. Occupational and leisure noise exposures as risk factors for adult hearing impairment: The Nord Trondelag (NT) Hearing Study. Howard J. Hoffman, Robert H. MacTurk, George W. Reed (Epidemiology, Statistics & Data Systems Branch, NIDCD, NIH, Bethesda, MD 20892), Kristian Tambs, Jostein Holmen (Natl. Inst. of Public Health (Folkehelsa), Oslo, Norway), and Hans M. Borchgrevink (The Natl. Hospital, Oslo, Norway)

In the NT Hearing Study (1995–1997), air-conduction thresholds were obtained using sound attenuation booths. This report is based on 27 997 subjects selected in a nested, case-control study to receive follow-up questionnaires on work and leisure noise exposures. Cases with low-frequency hearing loss (HL)—speech frequencies 500, 1000, and 2000 Hz—were defined by a pure-tone average (PTA) exceeding 30 dB in the better ear. A second case series with only high-frequency HL was defined by PTA exceeding 30 dB at 3000, 4000, and 6000 Hz. Controls were defined by PTAs less than 30 dB in both ears. Multivariate logistic regression analyses were stratified by sex and age. Family history of HL and recurrent ear infections in childhood were significant risk factors across all sex and age strata. Among men under 50, an odds ratio (OR) of 1.87 [95% confidence interval (CI): 1.53–2.28] for high-frequency HL was found for loud noise at work, an OR=1.35 (CI: 1.08–1.69) for use of noisy equipment, and OR=1.09 (CI: 1.02–1.17) for hunting/sport shooting. Low-frequency HL was related to occupational noise exposures in older (50+ years), but not younger men. Tinnitus was associated with occupational noise exposure at all ages.

3:20

4pNSb2. Noise dosimetry for telephone users. Philippe Moquin (Mitel Corp., 350 Legget Dr., Kanata, ON K2K 1X3, Canada, philippe_moquin@mitel.com)

Due to recent F.C.C. regulations, all telephones in North America will be required to have at least 12 dB of amplification. While this is a definite asset to the hard of hearing community, there are fears that normal hearing workers who use the telephone most of the day could suffer noise-induced hearing loss. Theoretical calculations validate this hypothesis. Measurements of actual levels in a call center were taken. Finally, a method of providing noise dosimetry within a digital telephone set is presented. Simultaneous measurements show good correlation.

3:35

4pNSb3. Progress on a rating system for hearing protector attenuation. William J. Murphy and John R. Franks (Bioacoustics and Occupational Vib. Section, NIOSH MS C-27, 4676 Columbia Pkwy., Cincinnati, OH 45226-1998)

The cumulative distribution of A-weighted noise reduction (CDNRA) has been proposed as an alternative to the noise reduction rating for hearing protectors [W. J. Murphy and J. R. Franks, J. Acoust. Soc. Am. **105**, 1131 (1999)]. Individuals' REAT data were A-weighted and summed across frequencies to determine the statistical distribution. The cumulative distribution of REAT data was fit to a combination of two logistic curves to capture the features of well-fit and poorly-fit hearing protectors. Previously, the CDNRA model was shown to yield Pearson's correlation coef-

ficients of 0.99 or better. In this paper, the CDNRA theory will be reviewed and the method will be applied to the subject-fit data from six HPDs. The Bilsom UF-1 earmuff was rated at 23-dB attenuation. The V-51R was rated at 6-dB attenuation. The EP100 was rated at 7-dB attenuation. And the EAR Classic was rated at 23-dB attenuation. Subject-fit data from the EAR Express and Howard Leight Max earplugs will be presented as well.

3:50

4pNSb4. Evaluation of a FitCheck hearing protector test system. William J. Murphy, John R. Franks, and Dave A. Harris (Bioacoust. and Occupational Vib., NIOSH, 4676 Columbia Pkwy., Cincinnati, OH 45226-1998)

The measurement of attenuation provided by a hearing protector is an important element in the development of an effective hearing-loss prevention program. Three systems were evaluated for accuracy compared with a sound-field test of real-ear attenuation at threshold (REAT) [J. R. Franks, J. Acoust. Soc. Am. **105**, 1129 (1999)]. The FitCheck method used with headphones was the most feasible system to use for field-test situations. However, the FitCheck method produced REAT values in the low frequency that were less than those produced by the sound-field method when used with the EAR Express earplug. A subsequent study has been conducted which evaluated the EAR Classic and the Howard Leight MAX earplugs with the FitCheck and sound-field methods. Twenty-four subjects were instructed to fit the devices without experimenter intervention (*re*: ANSI S12.6 Method B subject-fit protocol). Subjects were not allowed to adjust the earplugs between test methods. Mean REAT data from both methods will be compared. Preliminary data suggest that the FitCheck method yields lower REAT estimates for frequencies below 1 kHz, but the differences were not statistically significant.

4:05

4pNSb5. Changes in attenuation in personal hearing protector over time during use. Michael P. Valoski and George Durkt, Jr. (MSHA, Tech. Support, P.O. Box 18233, Pittsburgh, PA 15236)

Ideally, implementing engineering controls reduces workers noise exposures below the permissible exposure level (PEL). However, some occupations continue to have exposures above the PEL. To protect the hearing sensitivity of these workers, personal hearing protectors (PHP) must be worn. Manufacturers test PHPs under ideal conditions in a laboratory. For these tests, highly motivated and trained subjects remain stationary. Furthermore, a researcher helps the subject fit the PHP. Shortly after determining that the PHP was properly fitted, the tests are conducted. Under these conditions the optimum attenuation afforded by the PHP is achieved. Mining conditions differ substantially from those under which the laboratory attenuations are determined. Often, workers wear their PHPs for extended periods without removal. Moreover, a worker's constant movement may disrupt the seal of a PHP. Once the seal is broken, the PHP provides degraded performance. This project examines any change in earmuff attenuation over a 3-h wearing time in a laboratory. After donning an earmuff unaided, a subject wore the earmuff for 3 h while walking and stopping to lift containers from the floor to a table in a large reverberation chamber. It was found that the PHP's attenuation did not change substantially during the test.

THURSDAY AFTERNOON, 4 NOVEMBER 1999 — MADISON ROOM, 1:30 TO 5:45 P.M.

Session 4pPA

Physical Acoustics: Thermoacoustics

Robert A. Hiller, Cochair
Department of Physics and Astronomy, University of Mississippi, University, Mississippi 38677

Philip S. Spoor, Cochair
Los Alamos National Laboratory, MST-10, MS K764, Los Alamos, New Mexico 87545

Contributed Papers

1:30

4pPA1. Nonresonant referenced laser-induced thermal acoustics (LITA) for measurement of the speed of sound in gases. Allan J. Zuckerwar, Roger C. Hart, Jeffrey Balla, and Gregory C. Herring (NASA Langley Res. Ctr., M.S. 236, Hampton, VA 23681)

Nonresonant laser-induced thermal acoustics (LITA) is an effective tool for measuring the speed of sound in gases and is suitable for operation at elevated temperatures. Counterpropagating sound waves of known, fixed wavelength are generated electrostrictively by crossing two split laser beams which are generated from a short-pulse pump laser. The sound waves form a Bragg grating, which is illuminated by a second, long-pulse probe laser. A small fraction of the probe beam is diffracted to a detector, which permits accurate measurement of the sound frequency. Since the method comprises a free-filed measurement, the corrections required of a conventional acoustic interferometer or resonator are not necessary here. The presentation will include the principle of operation of LITA, comparison with other methods of measuring the speed of sound in gases, and sample measurements at temperatures up to 650 K.

1:45

4pPA2. Attenuation in tertiary gas mixtures with two relaxing components. Yefim Dain and Richard M. Lueptow (Dept. of Mech. Eng., Northwestern Univ., Evanston, IL 60208)

Vibrational relaxation accounts for absorption and dispersion of acoustic waves in gases that can be significantly greater than the classical absorption mechanisms related to shear viscosity and heat conduction. This vibrational relaxation results from retarded energy exchange between translation and intramolecular (vibration) degrees of freedom. Theoretical calculation of the vibrational relaxation time of gases based on the theory of Landau and Teller [Phys. Z. Sowjetunion **10**, 34 (1936)] and Schwartz *et al.* [J. Chem. Phys. **20**, 1591 (1952)] has been applied at room temperature to the polyatomic mixtures of nitrogen with water vapor and methane as relaxing components. The theory accounts for vibrational–vibrational coupling of all constituents together with vibrational-translational energy transfer. For negligible concentrations of methane and small vibrational–vibrational coupling the analytical expression matches previous theoretical and experimental results for the binary nitrogen–water mixture. For tertiary nitrogen–water–methane mixtures, the humidity and methane dependence of the nitrogen relaxation frequency differs from that of binary mixtures of either component with nitrogen due to coupling between the components. The significance of vibrational–vibrational coupling of binary pairs in the tertiary mixture has been analyzed.

2:00

4pPA3. Time-of-flight measurements on high-frequency surface acoustic phonons in silicon. D. M. Photiadis (Naval Res. Lab., Washington, DC) and J. Ding (ThermaWave, Inc., San Francisco, CA)

Results from time-of-flight measurements of high-frequency surface acoustic phonons (50–500 GHz) in (001) silicon wafers at low temperatures are reported. Pulsed laser excitation is used for the generation of the high-frequency surface acoustic phonons and aluminum edge bolometers are used as detectors. The times defined by the fast initial response of the edge bolometer correspond to surface acoustic delays in silicon. Late time responses consistant with echoes from the back face of the wafer were also observed. These results and the accompanying analysis will be discussed. [This research was supported by the Office of Naval Research.]

2:15

4pPA4. Acoustics on the planet Mars: A preview. Victor W. Sparrow (Grad. Prog. Acoust., Penn State Univ., 157 Hammond Bldg., University Park, PA 16802, sparrow@helmholtz.acs.psu.edu)

Assuming no operational difficulties, NASA's Mars Polar Lander will settle on the Martian surface in December 1999 and will transmit back to Earth sound samples from an audio frequency microphone designed and built by the University of California Space Sciences Laboratory and funded by The Planetary Society. Information on this MARS MICROPHONE is available on the World Wide Web at plasma2.ssl.berkeley.edu/marsmic/ and planetary.org. It is hoped that the microphone will detect natural environmental sounds on Mars as well as monitor operational noises made by the lander itself. This talk presents what little is known about the acoustic environment of Mars. Using available knowledge of the Martian surface atmosphere, the acoustics between Earth and Mars can be easily compared. Although the sound speeds on Earth (343 m/s) and Mars (228 m/s) are comparable, Mars' ambient density is only 1.4% of that on Earth. The resulting difference in characteristic impedances implies that a machine vibrating on Mars will radiate sound at a level approximately 20 dB lower than on Earth. Additional comparisons will be presented. [The helpfulness of Dr. Janet Luhmann and Dr. Greg Delory of the Berkeley SSL is appreciated.]

2:30

4pPA5. Design and construction of a small thermoacoustic refrigerator. Tamra S. Underwood, Dana M. Smith, and Ralph T. Muehleisen (Dept. of Civil, Environ., and Architectural Eng., Univ. of Colorado, Boulder, CO 80309)

While there has been much recent work on the development of medium to large scale (hundreds to thousands of watts) thermoacoustic devices, there has been relatively little work done on small thermoacoustic devices. Larger devices are often desired in order to get large cooling powers, but such devices can be expensive and difficult to construct. Small devices on the other hand, can be both inexpensive and easy to construct. To more carefully investigate small thermoacoustic devices, a refrigerator approximately 1 in. in diameter and 5 in. long has been constructed with a target of 10 W of cooling power and a temperature span of 25 °C. To achieve higher cooling powers, several of the devices can be operated in parallel. The performance of the actual device and the comparison to the predicted values will be presented. [Work supported by the Office of Naval Research.]

2:45–3:00 Break

3:00

4pPA6. Thermal performance of heat exchangers for thermoacoustic refrigerators. Yuwen Chen and Cila Herman (Dept. of Mech. Eng., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218, herman@titan.me.jhu.edu)

Heat exchangers in thermoacoustic refrigerators extract heat from the refrigerated volume and reject it to the surroundings. Eight essential heat-transfer processes coupled in the energy migration from the cold-side heat exchanger through the stack to the hot-side heat exchanger were identified, and a simplified computational model describing them was developed. Geometrical and operational parameters as well as thermophysical properties of the heat exchangners, the stack plate, and the working medium were organized into dimensionless groups. Heat transfer in the transverse direction, resulting from temperature differences between the inlet and outlet temperatures of the transport fluid in the heat exchangers, was accounted for in the simulations. Two types of boundary conditions between the thermoacoustic working fluid and the heat exchangers were considered: (1) constant temperature of the thermoacoustic working fluid over the heat exchangers with constant temperature difference along and across the stack and (2) constant heat flux from the thermoacoustic working fluid to the heat exchanger. Nonlinear temperature distributions and heat fluxes near the edge of the stack plate were observed. Effects of different parameters on the thermal performance of the heat exchangers were investigated. [Work supported by the Office of Naval Research.]

3:15

4pPA7. Use of electrodynamic drivers in thermoacoustic refrigerators. Ray Scott Wakeland (Grad. Prog. in Acoust., Penn State Univ., ARL, P.O. Box 30, State College, PA 16804, wakeland@psu.edu)

Some issues involved in matching electrodynamic drivers to thermoacoustic refrigerators are examined using an equivalent circuit model. Conclusions are that the driver should be chosen to have a large product $(Bl)^2/(R_e R_m)$. The suspension stiffness should be chosen to make the combined impedance of the mechanical and acoustical parts of the system entirely real at the operating frequency. The piston area should be selected to maximize electroacoustic efficiency or other desired parameters by matching the acoustical load to the optimum mechanical load for the particular driver. Alternately, if the piston area is fixed, the operating frequency can be adjusted to make this same match. Measurements made at this laboratory [R. W. M. Smith *et al.*, J. Acoust. Soc. Am. **105**, 1072(A) (1999)] provide experimental support for the conclusions of this paper concerning the attainable maximum electroacoustic efficiency. [Work supported by NSF and ONR.]

3:30

4pPA8. Thermoacoustic properties of nonuniform stacks. Gabriela Petculescu and Larry A. Wilen (Dept. of Phys. and Astron., Ohio Univ., Athens, OH 45701, wilen@helios.phy.ohiou.edu)

Thermoacoustic stacks of uniform cross section can be characterized completely in terms of the single thermoviscous function $F(\lambda)$ [W. P. Arnott, H. E. Bass, and R. Raspet, J. Acoust. Soc. Am. **90**, 3228–3237 (1991)]. For nonuniform stacks, one must treat thermal diffusion and viscosity separately, and it may be possible to ''tune'' the performance of the stack by taking advantage of the extra geometric freedom. Measurements have been performed to determine the thermoviscous characteristics of pin-like stacks for which the elements are oriented either along, or perpendicular to, the temperature gradient. The measurements characterize terms in the thermoacoustic equations responsible for thermal dissipation and thermoacoustic gain. The results for pins of varying size, shape, and spacing will be discussed. Time permitting, applications of novel designs to compact lumped element oscillators will be demonstrated. [Work supported by the Office of Naval Research.]

3:45

4pPA9. An acoustic impedance measurement of a parallel-pore stack used to determine the thermoviscous functions of the stack. Timothy G. Simmons, Richard Raspet (Dept. of Phys. and Astron., and Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677, tgsimmon@olemiss.edu), and Jeremy Brewer (Rhodes College, Memphis, TN 38112)

Recently, thermoviscous functions have been measured for single pores of various cross-sectional geometry [L. A. Wilen, J. Acoust. Soc. Am. **103**, 1406–1412 (1998)]. This investigation consists of measuring the acoustic impedance through a stack with parallel pores, inverting to yield the thermoviscous functions $F(\lambda)$ and $F(\lambda_T)$, and determining the accuracy with which the inversion can be accomplished. The measurement shall be performed using a standard impedance tube with the typical measurements taken, i.e., pressure maximum, pressure minimum, and the positions of two pressure minima. In order to be able to solve for the thermoviscous functions, a second measurement must be taken with the stack positioned away from the rigid termination, leaving a region of air. The impedance translation theorem (modified to account for the stack porosity) will have a functional form containing both $F(\lambda)$ and $F(\lambda_T)$ and shall be employed for both stack placements to provide two equations which can be solved for the two unknowns, $F(\lambda)$ and $F(\lambda_T)$, for a given frequency. If the method proves successful, modifications shall be made to the experimental setup so that it can accommodate a temperature gradient imposed across the stack. [Work supported by ONR.]

4:00

4pPA10. Thermoacoustics of moist air with wet stacks. William V. Slaton and Richard Raspet (Natl. Ctr. for Physical Acoust., University, MS 38677, wmslaton@meta3.net)

Recent work in porous media has led to an analytical treatment of sound in wet-walled tubes [Raspet *et al.*, J. Acoust. Soc. Am. **105**, 65–73 (1999)]. This work has been extended to examine thermoacoustic refrigerators which utilize moist air at atmospheric pressure as the working fluid as described by Hiller [Thermoacoustic Review Meeting 1999, National Center for Physical Acoustics]. In the described system the moisture in the air stream is cooled below its dew point and appears as either liquid on the stack surface or as a fog. This physical system can be adequately described by ignoring thermal diffusion of the water vapor (Soret effect) and neglecting temperature variations in the tube wall. The effects on sound propagation and efficiency due to the presence of liquid water in the system described above will be discussed. [Work supported by ONR.]

4:15–4:30 Break

4:30

4pPA11. Experimental investigation of the acoustical properties of narrow tubes filled with a mixture of gas and condensable water vapor. D. Felipe Gaitan, Richard Raspet, Craig J. Hickey, and William Slaton (Univ. of Mississippi, NCPA Coliseum Dr., University, MS 38677)

It is generally believed that using the latent heat of condensable vapors will increase the efficiency and energy density of thermoacoustic refrigerators. This phenomenon has been recently considered in the literature [Raspet *et al.*, J. Acoust. Soc. Am. **105**, 65 (1999)] in which the equations describing the acoustic properties of wet pores are described. In order to investigate some of the assumptions made in these equations (e.g., the boundary conditions at the walls of the wet pore), we have measured the acoustic damping in a small (1.3-mm-radius) tube filled with a mixture of gas and condensable water vapor. The explored parameter space included ambient pressure (0.1–2 bars), temperature (20–100 °C), and acoustic frequency (2–10 kHz). Results of these measurements will be reported. [Work supported by the Office of Naval Research.]

4:45

4pPA12. Heat transport by acoustic streaming in a resonance tube. George Mozurkewich (Ford Res. Lab., Maildrop 3028/SRL, P.O. Box 2053, Dearborn, MI 48121-2053)

The steady fluid flow known as acoustic streaming is capable of transporting heat advectively. This heat-transport mechanism has occasionally been applied for beneficially cooling a warm object, but in other contexts (such as thermoacoustic cooling) its effects can be detrimental. Gopinath and Mills have computed heat transport by streaming between two plane sections of a cylindrical resonator with insulating walls. In contrast, the experiments described here treat the interaction between streaming gas and thermally conductive cylinder walls. The walls of a horizontally oriented resonance tube were instrumented with a heater and more than 20 thermocouples to permit quantification of axial conduction through the solid walls and radially outward natural convection. By applying conservation of energy to the tube walls, the local, radially inward heat current, q(x), was deduced. The quantity q(x) is related to streaming transport within the tube. The measurements delineate regions where heat is added to and removed from the streaming gas. Although the magnitude of q initially increases with increasing acoustic amplitude, it quickly saturates. The fluid-dynamical implications of amplitude-independent q(x) will be discussed.

5:00

4pPA13. Streaming in thermoacoustic engines. Gordon Smith, Richard Raspet, and Henry Bass (Natl. Ctr. for Physical Acoust., Coliseum Dr., University, MS 38655)

Thermoacoustics utilizes an acoustic process to transport heat. Interestingly enough, sound sources that move essentially in a sinusoidal manner often do not yield flow fields of a sinusoidal nature. In addition to the oscillation of each fluid element contributing to the acoustic system, there can exist a pattern of steady vortices or other time-independent circulation regions known as acoustic streaming. Many different models for acoustic streaming have been developed, each stemming from a unique specific premise to the problem. Within a thermoacoustic device, acoustic streaming provides a means for a thermal short, in which a direct fluid flow would be established between two heat exchangers, permitting a much easier path for thermal transport than that established by a thermoacoustic process. However, little empirical data exist to provide a means of estimating the amount of thermal loss from such a process. Theoretical work to date will be presented, as well as experimental data from a single stage prime-mover thermoacoustic device. [Work supported by ONR.]

5:15

4pPA14. Characterization of the nuisance heat load in a thermoacoustic refrigeration demonstration device. Ray Scott Wakeland (Grad. Prog. in Acoust., Penn State Univ., ARL, P.O. Box 30, State College, PA 16804, wakeland@psu.edu)

The heat load in a thermoacoustic refrigeration demonstration device is studied. In the demo, which acts only to create a low temperature at the position of a thermocouple, the only heat load is nuisance heat, which is not well known. The method described here for estimating experimentally the size of this heat load assumes that heat transport associated with acoustics depends on acoustic amplitude p and on the magnitude of the deviation from ambient temperature $|\Delta T|$, but is independent of the sign of ΔT. That is, the heat load due to acoustics-induced transport is taken to have the form $\dot{Q}_{\text{transport}} = -F(p)\Delta T$, where $F(p)$ is some function of p only, and ΔT can be either positive (refrigerator) or negative (heater). Direct thermoviscous heating at the site of the thermocouple $\dot{Q}_{tv}(p)$ is also a function of p only. At steady state, $\dot{Q}_{\text{heater}} = F(p)\Delta T - \dot{Q}_{tv}(p)$. In the experiment, the refrigeration stack is replaced by an electric heater, the power of which is easily measured. The steady−state temperature rise above ambient ΔT is measured as a function of heater power and acoustic amplitude. The experiment generates a family of straight lines, the slopes of which determine the function $F(p)$, and the intercepts give $\dot{Q}_{tv}$. [Work supported by ONR and NSF.]

4p THU. PM

5:30

4pPA15. Separation of gas mixtures in the acoustic boundary layer. Philip S. Spoor and Gregory W. Swift (Condensed Matter and Thermal Phys. Group MST-10, Los Alamos Natl. Lab., Los Alamos, NM 87545)

Spatial separation of binary gas mixtures in the presence of a sound wave has been reported by various experimenters in the last half-century, and is usually attributed to static pressure or temperature gradients. The authors suggest an additional mechanism, a ''concentration pumping'' in the acoustic boundary layer analagous to thermoacoustic heat pumping. Data from a simple acoustic mixture-separation experiment will be compared with predictions of this new theory. Observation of mixture separation between two acoustically coupled thermoacoustic engines will also be discussed.

THURSDAY AFTERNOON, 4 NOVEMBER 1999 MARION ROOM, 2:00 TO 4:30 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Pitch and Complex Stimuli

Gary R. Kidd, Chair

Department of Speech and Hearing Science, Indiana University, Bloomington, Indiana 47405

Contributed Papers

2:00

4pPP1. Pitch of envelope-modulated rippled noise. Stanley Sheft and William A. Yost (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626, ssheft@luc.edu)

Both amplitude modulation (AM) and the delay-and-add process of rippled-noise (RN) generation can impart a pitch to wideband Gaussian noise. For AM, and possibly also RN, the basis of the pitch percept derives from temporal processing of the stimulus. In the present study, the pitch of envelope-modulated rippled noise was matched to filtered pulse trains. RN was generated with a delay of either 5 or 10 ms with either one or eight iterations and a gain of either plus or minus one. AM rates ranged from 50 to 200 Hz. Without rippling the carrier, accuracy in pitch matching to AM rate was fairly consistent over the range of rates studied with most deviations occurring as ''octave errors.'' With envelope modulation of RN carriers, pitch matches were distributed between pitches associated with the AM rate and fine-structure delay, in some cases with a noticeable increase in ''octave errors.'' Even with eight iterations in RN generation, an influence of AM rate on pitch matches was obtained. Pitch-match distributions can be roughly accounted for by consideration of the largest positive peak of both the fine-structure and envelope autocorrelation functions. [Work supported by NIH and AFOSR.]

2:15

4pPP2. Pitch strength of regular interval stimuli as a function of duration and filtering. William A. Yost (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626)

The ability of listeners to discriminate between a random interval noise and a regular interval stimulus, iterated rippled noise (IRN), was measured as a function of duration and bandpass filtering. IRN is generated by delaying (by d ms) a noise and adding it back to the undelayed noise after attenuation (g) in an iterative process (where n is the number of iterations). A two-alternative, forced-choice adaptive procedure was used in which stimulus duration was varied adaptively. Different values of g, d, n, and filter condition were used. In general, the duration required to discriminate IRN from regular interval noise increased as the filter condition provided less low-frequency energy. Duration thresholds were higher for high values of attenuation. The smallest duration thresholds were obtained for d in the range of 2–5 ms. The results will be discussed in terms of temporal models of the pitch of complex stimuli. [Work supported by NIDCD and AFOSR.]

2:30

4pPP3. Finding the missing fundamental: A connectionist model of harmonic detection. Clifford F. Lewis, Stephan B. Fountain (Dept. of Psych., Kent State Univ., P.O. Box 5190, Kent, OH 44242-0001, clewis@kent.edu), and Michael K. McBeath (Arizona State Univ., Tempe, AZ 85287-1104)

A major shortcoming of the place-coding principle of pitch perception has been that it does not account for the phenomenon of virtual pitch, or the missing fundamental, because there is no physical energy on the basilar membrane at the location corresponding to the fundamental frequency. A connectionist model provides an example of a simple mechanism for virtual pitch that is consistent with place-coded input. An autoassociator network was trained using place-coded amplitudes of harmonics that corresponded to tones produced by a variety of musical instruments. After training, the network exhibited the missing fundamental illusion. Specifically, activation was present in the output unit corresponding to the fundamental frequency, even when the input energy at the fundamental was removed. This activation was approximately an order of magnitude larger than the activation of other units receiving no input energy, and it was the same order of magnitude as the activation when the fundamental frequency was present. The model demonstrates how virtual pitch can result from learned associations between harmonics that typically occur simultaneously in nature. When one of the harmonics is missing, the network simply fills in the missing part of the pattern.

2:45

4pPP4. On loudness and pitch of ''complex noise'' in relation to the factors extracted from the autocorrelation function. Shin-ichi Sato, Hiroyuki Sakai, and Yoichi Ando (Grad. School of Sci. and Technol., Kobe Univ., Rokkodai, Nada, Kobe, 657-8501 Japan)

In a previous study, it was shown that the loudness of bandpass noise within the critical band changes increasing the effective duration of the envelope of the normalized autocorrelation function (ACF) τ_e changing the filter slope of bandpass noise and the subsequent reverberation time T_{sub} of sound field [Y. Ando, *Architectural Acoustics—Blending Sound Sources, Sound Fields, and Listeners* (AIP Press/Springer-Verlag, New York, 1998), Chap. 6]. In this study, the loudness of the complex signals including bandpass noises with different frequency (complex noise) is examined. The complex noises consist of the bandpass noises whose center frequencies are the harmonics of the fundamental frequency. The bandwidth of each component is changed within the critical band by use of a sharp filter of more than 1000 dB/octave. Subjects judge the loudness of complex noise by comparing the tone of its perceived pitch frequency, the ''missing fundamental.'' The results are discussed with the four factors extracted from the autocorrelation function, which are τ_e, the energy represented at the origin of the delay $\Phi(0)$, and the amplitude and the delay time of the first peak of ACF ϕ_1 and τ_1. [Work supported by ACT-JST and Grant-in-aid for Scientific Research, JSPS, Japan.]

3:00

4pPP5. Absolute pitch is demonstrated in speakers of tone languages. Diana Deutsch (Dept. of Psych., Univ. of California, San Diego, La Jolla, CA 92093), Trevor Henthorn (Univ. of California, San Diego, La Jolla, CA 92093), and Mark Dolson (E-mu Creative Technol. Ctr., Scotts Valley, CA 95067)

In two experiments, Vietnamese and Mandarin speakers manifested a precise form of absolute pitch in reading lists of words. In one experiment, seven Vietnamese speakers read out a list of ten words twice, on two separate days. The average pitch of each word was determined by computer analysis, and for each speaker comparisons were made between the average pitches produced by the same word on different days. Remarkable correspondences were obtained: For most speakers, the signed pitch difference, averaged across words, was well within a semitone. In another experiment, 15 Mandarin speakers read out a list of 12 words twice within a session, with the 2 readings separated by roughly 20 s; and in 2 sessions which were held on different days. Again, comparing across sessions, for most speakers the signed pitch difference, averaged across words, was well within a semitone. Furthermore, the differences were no greater when the pitches were compared across sessions than within sessions. These findings demonstrate that tone language speakers are able to refer to a remarkably precise and stable absolute pitch template in producing words. Relationships to other work on absolute pitch are discussed, and the findings are illustrated by sound recordings.

3:15

4pPP6. Infants' discrimination of spectral slope. Christine D. Tsang and Laurel J. Trainor (Dept. of Psych., McMaster Univ., Hamilton, ON L8S 4K1, Canada)

Spectral slope is a global property of the spectral envelope representing the change in energy across spectral frequency. Spectral slope is a dimension for discriminating speech, voices and musical instrument timbres. The average spectral slope of speech and music is about -6 dB/octave. If spectral slope is important in timbre perception, enhanced sensitivity for spectral slopes around -6 dB/octave would be expected, and poor discrimination would be expected for spectral slopes very different from this value. Eight-month-old infants were tested across a range of spectral slopes (-16 to $+16$ dB/octave) to determine whether infants' slope discrimination is best near -6 dB/octave. The stimuli were generated according to the following equation: $dB_i = b \log_2 i$, where i is the harmonic number, b is the spectral slope value, and dB_i is the relative amplitude of harmonic i. The results show that infants were only sensitive to differences in spectral slope in a limited range between -10 and -4 dB/octave, despite the fact that local intensity cues were smallest in this range. This suggests that infants are using spectral slope as a dimension for discriminating timbre and that the auditory system is tuned to sounds with real-world spectral slopes. [Research supported by NSERC.]

3:30

4pPP7. Sound quality judgments of everyday sounds. Gary R. Kidd and Charles S. Watson (Communication Disorders Technology, Inc., 501 N. Morton St., Ste. 215, Bloomington, IN 47404)

An experiment was conducted to determine the psychological dimensions that underlie listeners' judgments of the qualities of everyday sounds and to identify the stimulus properties on which such judgments are based, using Osgoods' semantic differential technique. A collection of 145 common sounds was presented to 32 listeners who rated the sounds on 20 seven-point rating scales. The sounds were of a variety of common objects and events (e.g., dishwasher, vacuum cleaner, door closing) as well as ambient sounds that sometimes involved several objects and events (e.g., traffic, cafeteria noise, rain). Rating scales included subjective adjective pairs (e.g., happy–sad, relaxed–tense) and objective pairs (e.g., hard–soft, large–small). A principal components analysis of the rating data indicated that average judgments of the listeners were associated with four dimensions, roughly corresponding to large–small, compact–scattered, harsh–mellow, and overall quality (high–low). However, the level of agreement across subjects varied considerably for the different sounds and rating scales. The relations between several static (averaged over the entire waveform) and dynamic (time-varying) stimulus properties and listeners' judgments will be discussed. [Work supported by NIH/NIMH.]

3:45

4pPP8. Observer weighting of monaural level information in a pair of tone pulses. Mark A. Stellmack and Neal F. Viemeister (Dept. of Psych., Univ. of Minnesota, 75 E. River Rd., Minneapolis, MN 55455)

In a cued single-interval task, listeners were presented a pair of standard pulses (each 10 ms, 1 kHz) of fixed level followed by a pair of similar tone pulses, the comparison, separated by a fixed interpulse interval (IPI of 2–256 ms. Listeners were instructed to attend to either the first or second pulse of the comparison and to indicate whether the target pulse was higher or lower in level than the corresponding pulse of the standard. The levels of the pulses in the comparison were selected randomly and independently in each trial ($\mu = 75$ dB SPL, $\sigma = 8$ dB). The relative weight given to each pulse was obtained by computing the correlation between the level of the first or second pulse in dB and the listener's responses. Listeners adjusted their weighting strategy most appropriately for the given task at the largest IPI. as IPI decreased, listeners gave increasingly similar weight to the two pulses for both target positions. This suggests that level information was integrated across the pulses. Thus, it appears that in certain situations integration of level information occurs over much larger time intervals than previously supposed. The data are not consistent, however, with simple temporal integration of the pulses. [Work supported by NIH DC 00683.]

4:00

4pPP9. Subjective evaluation of audio compression algorithms: Do conservatory students have a ''golden ear?'' Sam Carrier (Dept. of Psych., Oberlin College, Oberlin, OH 44074) and Schuyler Quackenbush (AT&T Labs-Res., 180 Park Ave., Bldg. 103, Florham Park, NJ 07932)

Digital audio compression algorithms are typically based upon psychoacoustic models of hearing; thus they are known as perceptual coders. For commercial applications such as providing music over the Internet, where bit-rate reduction is essential, these lossey codecs have customarily been evaluated through subjective testing using professional audio engineers as listeners. This study replicated the MPEG-2 AAC Stereo Verification Tests (ISO/IEC JTC1/SC29/WG11 N2006) with the 20 listeners drawn from undergraduates enrolled in the Oberlin Conservatory of Music. These tests used the triple-stimulus hidden-reference double-blind methodology with eight codecs and 10 test stimuli. In this procedure, listeners compare two test signals (one coded, the other not) to an uncoded reference, then rate the ''audio quality'' of both test stimuli on a 1–5 scale in 0.1-step increments. The test stimulus selected as the hidden reference is graded 5.0, the coded signal graded on the degree of impairment. Difference scores thus reflect the discriminability of the stimuli. Taking the magnitude of Studentized t-statistic values as a measure of acuity, MPEG listeners averaged 4.45, while Oberlin listeners averaged 9.86. While there were some methodological differences between the two experiments, subjects differed several ways Oberlin listeners were younger and were engaged in formal musical training.

4:15

4pPP10. The relations between auditory problems and cognitive, linguistic, and intellectual measures in first grade children. Gary R. Kidd, David A. Eddins, and Charles S. Watson (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405)

The relations between hearing ability and several sensory, perceptual, and cognitive abilities were determined for all children entering first grade in the Benton County, Indiana school system for three consecutive years. This research was part of a multi-disciplinary, epidemiological, longitudi-

nal study of the factors that influence the development of reading and language skills in the early grades. As a result of an auditory evaluation that included otoscopy, tympanometry, pure-tone screening, and speech audiometry, two groups of children with auditory problems were identified who also had lower scores on nonauditory measures. In the total sample of 459 children, 24 had hearing thresholds excess of 20 dB HL and 51 had either excessive ear wax or evidence of ear infection based on otoscopy. Scores for both groups on measures of reading, cognitive/intellectual abilities, and on the SCAN central auditory test battery were significantly lower than for children with normal auditory function. At entry into second grade, however, reading achievement for both groups was within normal limits. The association between these measures of auditory function and socio-economic status will be discussed. [Work supported by the Benton County School System and by a grant from Indiana University.]

THURSDAY AFTERNOON, 4 NOVEMBER 1999 MORROW ROOM, 1:30 TO 4:30 P.M.

Session 4pSA

Structural Acoustics and Vibration: Vibration and Radiation—Simple Structures

Richard P. Szwerc, Cochair
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Contributed Papers

1:30

4pSA1. Wave propagation experiments using point load excitation in a multi-layered structure. Chunnan Zhou, Jan D. Achenbach (Ctr. for Quality Eng. and Failure Prevention, Northwestern Univ., Evanston, IL 60208), and John S. Popovics (Drexel Univ., Philadelphia, PA 19104)

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

1:45

4pSA2. A review and critique of theories for piezoelectric laminates. V. G. Senthil, Vasundara V. Varadan, and Vijay K. Varadan (Ctr. for the Eng. of Electron. and Acoust. Mater., Penn State Univ., University Park, PA 16802, vvvesm@engr.psu.edu)

This paper presents a three-dimensional (3-D) complete field solution for active laminates based on a modal, Fourier series solution approach that is used to compute all the through thickness electromechanical fields near the dominant resonance frequency of a beam plate with two piezoelectric (sensor and actuator) and one structural layers. Then a detailed review of the extant laminate models used for piezoelectric laminates emphasizing the underlying assumptions in each case is presented. The nonzero, through thickness field components are computed under these assumptions. The results of the 3-D model and FSDT model are compared for two aspect ratios (AR thickness to width of layers). An AR of 20 is at the limit of FSDT and an AR of 50 well within the assumptions of FSDT. It is concluded that for moderate aspect ratios, several of the approximations of FSDT are questionable at resonance frequencies, resulting in large errors in the estimation of the elastic and electric field components.

2:00

4pSA3. A novel technique for the prediction of the displacement of a line-driven plate with discontinuities. Daniel DiPerna and David Feit (Carderock Div., Naval Surface Warfare Ctr., CDNSWC, 9500 MacArthur Blvd., West Bethesda, MD 20817)

A novel technique is presented for obtaining approximate analytic expressions for an inhomogeneous line-driven plate. The equation of motion for the inhomogeneous plate is transformed, and the transform of the total displacement is written as a sum of the solution for a homogeneous line-driven plate plus a term due to the inhomogeneity. This expression may in general be solved numerically. However, by introducing a small parameter into the problem, it may be solved approximately using perturbation techniques. This series may not be convergent, but its convergence may be improved using the Padé approximation. Results are presented for the case of an inhomogeneity concentrated at a single point, and a distribution of points along the line-driven plate. [Work sponsored by the CDSNWC ILIR program and ONR, Code 334.]

2:15

4pSA4. Structural intensity in a point-excited infinite elastic plate in the high-frequency range. Sabih I. Hayek and Jungyun Won (Active Vib. Control Lab., Dept. of Eng. Sci. and Mech., 227 Hammond Bldg., Penn State Univ., University Park, PA 16802)

An infinite elastic plate is excited by a mechanical point force, which generates a vector active structural intensity field in the plate. In an earlier paper [Hayek *et al.*, J. Acoust. Soc. Am. **105**, 1299 (1999)], the structural intensity was evaluated analytically for a plate using the Bernoulli–Euler theory, which is valid in the low-frequency range. The paper then employed the same theory for the active control of total structural intensity by use of co-located point forces and moments. In this paper, the evaluation of the vector structural intensity field due to point forces and moments is modeled by use of the more exact Mindlin plate theory, which is valid in a much higher frequency range. The main objective is to achieve active

control of the total structural intensity in infinite elastic plates using the Mindlin plate theory through a co-located point force and a point moment actuator at an arbitrary location on the plate.

2:30

4pSA5. The measurement of structural intensity on a plate with nonuniform thickness. Richard Szwerc, Henry Chang (Naval Surface Warfare Ctr., Code 725, 9500 MacArthur Blvd., West Bethesda, MD 20817-5700, szwercrp@nswccd.navy.mil), and Courtney Burroughs (Penn State Univ., State College, PA 16804)

A method to measure the structural intensity on a plate with nonuniform thickness is presented and demonstrated with experimental results. The method is an extension of the wave decomposition method, previously demonstrated on uniform thickness beams. It treats a plate as a superposition of beams, and relies only upon local rather than global vibration measurements. The average thickness in the local region of the measurement points is the plate thickness used in the calculations of the power levels.

2:45–3:00 Break

3:00

4pSA6. Estimating radiation modes using wave-number filtering. Scott D. Sommerfeldt and Dong Lin (Dept. of Phys., Brigham Young Univ., Provo, UT 84602, s_sommerfeldt@byu.edu)

Acoustic radiation modes have been investigated in recent years as a means of estimating the far-field radiated acoustic power from a structure. The authors have been investigating two possible methods of estimating the amplitudes of the radiation modes associated with radiation from a baffled beam. The first method, based on using distributed sensors shaped according to the radiation modes, has been reported previously [S. D. Sommerfeldt and D. Lin, J. Acoust. Soc. Am. **105**, 1088 (1999)]. This paper will report a second method, based on using distributed sensors designed to act as low-pass wave-number filters. The spatial filtering provided by the sensors allows the array to estimate the radiation mode amplitudes with only a small number of sensors, while still avoiding spatial aliasing problems. Numerical analysis indicates that the method is capable of providing good estimates of the radiated acoustic power for a normalized frequency range extending up to $kl=10$, using an array of only six sensors. Although the radiation mode shapes are frequency dependent, the method has demonstrated the ability to provide good modal estimates throughout this entire frequency range. Broadband estimates of the acoustic radiation are typically within 1 dB of the actual value.

3:15

4pSA7. A computational acoustic field reconstruction process based on an indirect variational boundary element formulation. Zhidong Zhang, Nickolas Vlahopoulos (Dept. of Naval Architecture and Marine Eng., Univ. of Michigan, 2600 Draper Rd., Ann Arbor, MI 48109-2145), S. T. Raveendra (Automated Analysis Corp., 2805 S. Industrial, Ann Arbor, MI 48104), T. Allen, and K. Y. Zhang (Ford Motor Co., 20901 Oakwood Blvd., MD299, Dearborn, MI 48121)

The objective of this work is to develop a computational capability based on the indirect variational boundary element method (IVBEM) to evaluate appropriate velocity boundary conditions on an assembly of piston-type sources such that they can recreate a prescribed acoustic field. Information for the acoustic pressure of the original acoustic field at certain field points constitutes the input to the developed process. Several new developments associated with the IVBEM are necessary for completing the field reconstruction process. The developed computational capability targets exterior radiation applications. Thus a new formulation for treating irregular frequencies in the IVBEM is developed and implemented in the field reconstruction process. The computation of the appropriate velocities for the piston-type sources is based on deriving transfer functions between the field points, where the acoustic pressure of the original field is prescribed, and the velocity assigned on each element of the generic source. A singular value decomposition solver is integrated with the IVBEM computations in order to evaluate the velocity boundary conditions from the transfer functions. Finally, an algorithm that identifies the optimum field points from a set of candidate points for prescribing the acoustic pressure of the original field is developed. [Research supported by a University Research Program award from Ford Motor Co.]

3:30

4pSA8. Wave-number solution for a string with distributed mass properties. Mauro Pierucci (Dept. of Aerosp. Eng., College of Eng., San Diego State Univ., San Diego, CA 92182)

A structure with spatially variable properties when excited at a single frequency will excite many wave numbers. In this paper the vibration of a string with a mass distribution $m(x)$ given by a Fourier series expansion is analyzed. The equation governing the wave-number solution $w(k)$ for the string displacement is obtained via Fourier transform. The final result is a finite-difference equation. The finite-difference equation relates all of the displacement wave numbers to each other. The solution to this equation is obtained by rewriting it as a system of simultaneous equations that are then solved by standard linear algebra techniques. Wave-number spectral solutions are obtained for different mass distribution. In one case the mass distribution is assumed to vary in a discontinuous manner between adjacent bays, in another case the mass is assumed to vary in a ''triangular'' manner. The wave-number solutions are presented in 3-D graphs so that the magnitude of the displacement can be seen as a function of the wave number and either the magnitude of the additional mass, the area over which the mass is distributed, or the length of the bay. In the limiting case as the mass distribution becomes very concentrated, the classical pass/stop band, familiar with periodic structures, is recovered.

3:45

4pSA9. Love wave surface acoustic wave sensor for ice detection on aircraft. Vasundara V. Varadan, Sunil Gangadharan, and Vijay K. Varadan (Ctr. for the Eng. of Electron. and Acoust. Mater., Penn State Univ., University Park, PA 16802, vvvesm@engr.psu.edu)

This paper presents the design, fabrication, experimental results, and theoretical validation of a Love wave surface acoustic wave sensor for detecting the phase change from liquid water to solid ice. The sensing of this phase transition is due to the shear horizontal nature of Love waves which couple to a solid (ice) but not to a liquid (water). An SiO_2 film of thickness 3.2 μm deposited on an ST cut quartz wafer via plasma-enhanced chemical-vapor deposition acts as the guiding layer for Love waves. Testing is carried out with the water or ice placed directly in the propagation path of Love waves. An oscillation frequency shift of 2 MHz is observed when water on the sensor is frozen and melted cyclically. The contribution to the frequency shift is explained in terms of the acousto-electric effect (high permittivity and conductivity of water relative to ice), mass loading, and elastic film formation (solid ice). An arrangement for wireless interrogation of the sensor is proposed which is particularly attractive for aircraft and rotorcraft applications obviating the need for complex wiring and local power sources.
t

4:00

4pSA10. Bifurcation and chaos in flexural vibration of a baffled plate in mean flow. Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202) and Jason Zhu (General Motors Truck Group, Milford, MI 48380)

This paper presents the results of an investigation of flexural vibration of a finite plate clamped to an infinite baffle in mean flow. The plate flexural displacements are solved by the Galerkin method. The critical mean flow speed is determined by a general stability theory and the Routh

algorithm. Results show that the instabilities of a baffled plate are mainly caused by an added stiffness due to acoustic radiation in mean flow, but controlled by structural nonlinearities. The added stiffness is shown to be negative and increase quadratically with the mean flow speed. Without the inclusion of structural nonlinearities, the plate has only one equilibrium, namely, its undeformed flat position. Under this condition, the amplitude of plate flexural vibration grows exponentially in time everywhere, known as absolute instability. With structural nonlinearities, the plate may possess multiple equilibria. When the mean flow speed exceeds the critical value, the plate becomes unstable around all equilibria, jumping from one equilibrium position to another. Since this jumping is random, the plate flexural vibration may seem chaotic. This chaotic behavior disappears when viscous damping is introduced. Accordingly, the plate settles down to a ''buckled-down'' position owing to the hydraulic fluid-loading effect. [Work supported by ONR.]

4:15

4pSA11. Transverse wave motion of piezoelectric (622) layer due to step potential. C. Anandam, R. Natrajan (Dept. of Mathematics, A. C. College of Technol., Anna Univ., Chennai-600 025, India), and H. S. Paul (Acoustical Foundation, Gandhi Nagar, Adyar, Chennai-600 020, India)

The propagation of transverse waves in a piezoelectric layer of class (622) is investigated, when excited by either (a) step electric potential, keeping zero stress or (b) step shear stress, keeping zero potential. The resulting electric potential or shear stress is presented graphically. For case (a), electric potential linearly increases as the thickness of the plate increases, whereas stress is maximum near nondimensional thickness $r=0.1$ and decreases gradually and becomes zero at the top surface. For case (b), electric potential becomes negative and linearly decreases up to nondimensional thickness, $r=0.87$ and then increases and becomes zero at the top surface with slight oscillatory nature as the thickness of the plate increases. It is also observed that mere application of time dependent electric potential in piezoelectric (622) class generates transverse waves, which is not possible in any other piezoelectric material.

THURSDAY AFTERNOON, 4 NOVEMBER 1999 DELAWARE A & B ROOM, 1:30 TO 4:30 P.M.

Session 4pSC

Speech Communication: Speech Perception: Audio and/or Visual Cues (Poster Session)

John W. Hawks, Chair
School of Speech Pathology and Audiology, Kent State University, Kent, Ohio 44242

Contributed Papers

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

4pSC1. Perceptual effects of place of articulation on voicing for audiovisually discrepant stimuli. Lawrence Brancazio, Joanne L. Miller, and Matthew A. Paré (Dept. of Psych., Northeastern Univ., 125 NI, 360 Huntington Ave., Boston, MA 02115, brancazio@neu.edu)

This research investigated effects of the visually specified place of articulation on perceived voicing. It is known that the /b/–/p/ boundary along a voice-onset-time (VOT) continuum falls at a shorter VOT than the /d/–/t/ boundary. Green and Kuhl [Percept. Psychophys. **45**, 34–42 (1989)] demonstrated that tokens from an auditory /ibi/–/ipi/ continuum dubbed with an /igi/ video and perceived as /idi/ and /iti/ due to the ''McGurk effect'' had a voicing boundary at a longer VOT than when presented only auditorily and perceived as /ibi/ and /ipi/. We extended this finding in two directions. First, using an auditory /bi/–/pi/ series with a video /ti/ for which the McGurk effect did not always occur, we compared visually influenced (/d,t/) and visually uninfluenced (/b,p/) responses for these audiovisually discrepant stimuli. The /d/–/t/ boundary was at a longer VOT than the /b/–/p/ boundary, affirming the boundary shift's perceptual, not stimulus-based, origin. Second, we tested the generalizability of Green and Kuhl's findings using auditory /di/–/ti/ tokens with /ti/ and /pi/ videos. Preliminary results suggest that boundaries for /b/–/p/ and /bd/–/pt/ percepts (with video /pi/) may occur at shorter VOTs than for /d/–/t/ percepts (with video /ti/). Thus, the visually induced voicing boundary shift apparently replicates across different stimulus configurations. [Work supported by NIH/NIDCD.]

4pSC2. Talker variability effects in auditory-visual speech perception. Arlene E. Carney, Bart R. Clement (Dept. of Commun. Disord., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, carne005@umn.edu), and Kathleen M. Cienkowski (Univ. of Connecticut, Storrs, CT 06268)

Talker variability effects have been demonstrated for many aspects of syllable, word, and sentence recognition for stimuli presented in the auditory-only modality. In the current investigation, talker variability effects were examined in multiple modalities—visual-only, auditory-only, and auditory-visual. Auditory-visual presentations were of two types: consonant [auditory and visual were of the same token—/visual bi and auditory bi/] or disparate [/visual gi and auditory bi or visual bi and auditory gi/]. The disparate condition has elicited ''the McGurk effect,'' in which listeners may report a fused response that is neither the auditory or the visual component of the stimulus. Eleven talkers (five male and six female) of varying cultural backgrounds were videotaped. Their productions of CV syllables were presented to adult listeners who reported their percepts. Individual talkers elicited different degrees of syllable fusion among listeners, despite their equivalent auditory intelligibility. In addition, individual listeners varied considerably in their overall perceptions, from complete fusion for all talkers to almost no fusion for any talker. Results suggest that listeners may employ different types of processing strategies for bimodal stimuli. [Research supported by NIDCD.]

4pSC3. Audibility and visual biasing in speech perception. Bart R. Clement and Arlene E. Carney (Dept. of Commun. Disord., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, clem0025@umn.edu)

Evidence of an integrative interaction between vision and audition comes from studies of audiovisual disparity, in which conflicting auditory and visual speech tokens may elicit a fused perceptual response that is different from that represented in either modality alone (i.e., the ''McGurk Effect''). Visual information in speech can bias and in some cases even dominate the auditory percept. Hearing-impaired listeners may be more susceptible to visual biasing in speech perception than normal-hearing listeners because of an increased reliance on visual information. It is argued here that the degree of audibility of the auditory component of the stimulus must be equated across the two groups before true differences in visual biasing can be established. In the current investigation, the effects of visual biasing on speech perception are examined as audibility is systematically degraded for 12 normal-hearing listeners and two male talkers. Results demonstrate that the degree of visual biasing is in fact related to degree of audibility. Further, individual normal-hearing listeners may differ distinctly with regard to how they integrate the two modes of information. Both of these factors must be considered when this phenomenon is studied in hearing-impaired listeners. [Research supported by NIDCD.]

4pSC4. Testing the relative salience of audio and visual cues for stop place of articulation. Stephen J. Winters (Dept. of Linguist., Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, swinters@ling.ohio-state.edu)

Recent OT analyses have appealed to putatively universal rankings of perceptual entities in order to motivate analyses of phonological phenomena. J. Jun [WCCFL **14**, 221–237 (1995)] bases a universal constraint ranking for susceptibility of stops to place assimilation on the inherent ''salience'' of place cues. E. Hume [WCCFL **17** (1998)] appeals to the ''perceptual vulnerability'' of labial stops in order to account for their cross-linguistic tendency to undergo metathesis. Both phonologists motivate their analyses with speculative and unquantified assumptions about place cue salience. This study used an objective definition of salience to empirically establish the relative strengths of audio and visual cues for stop place of articulation. Comparing results between audio-visual and audio-only groups showed that visual cues are strongest for labials but are also significantly higher for coronals than dorsals. Adding acoustic information between speech reception threshold and comfortable listening level conditions had weaker effects than adding visual information, but contributed most strongly to the intelligibility of dorsals. Overall results suggested that labial stops have the strongest cues for place regardless of modality, contrary to what Hume, Jun and other researchers would predict. [This material is based upon work supported by a National Science Foundation Graduate Fellowship.]

4pSC5. Auditory-visual integration in younger and older adults. Mitchell S. Sommers (Dept. of Psych., Washington Univ., Campus Box 1125, St. Louis, MO 63130) and Nancy Tye-Murray (Central Inst. for the Deaf, St. Louis, MO 63130)

This study examined age-related changes in the ability to integrate auditory and visual speech information. Visual enhancement, the advantage afforded by seeing as well as hearing a talker, compared with listening alone, was measured in younger (18–25) and older (over age 65) adults. In addition, dichotic identification of filtered speech (auditory–auditory speech integration) and binaural gap detection (auditory–auditory nonspeech integration) were tested to evaluate whether auditory-visual integration was related to a more global ability to integrate sensory signals, be they speech or nonspeech. Visual enhancement was significantly impaired in older, compared with younger adults. Importantly, visual-only (lipreading) performance of younger and older adults was nearly identical, suggesting that age-related declines in visual enhancement were not a result of reduced reception of visual information. Older adults also exhibited deficits in the ability to integrate two channels of auditory information and these deficits were similar for both speech (sentences) and nonspeech (gap detection) stimuli. Regression analyses indicated significant correlations between the three measures of sensory integration, visual enhancement, dichotic sentence identification, and binaural gap detection. Taken together, the findings suggest that age-related declines in visual enhancement are a result of a generalized decline in sensory integration abilities.

4pSC6. Geometrical displays of speech spectra as aids to lipreading. Steven L. Tait, Jr. and William J. Strong (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT 84602, strongw@acoust.byu.edu)

Many syllables are ambiguous to lip readers because of their similar appearance on the lips. An aid is necessary to make distinction of these syllables possible. In a previous study [E. J. Hunter and W. J. Strong, J. Acoust. Soc. Am. **102**, 3166–3167 (1997)], speech spectra were processed at 5-ms intervals and displayed as sequences of irregular decagons. Subjects were asked to discriminate between pairs of decagons representing ambiguous syllables. Subjects successfully discriminated most pairs, but were inconsistent in discrimination of voiced-/unvoiced-stop pairs. This study is concerned with display formats aimed at improving discrimination of ambiguous pairs to develop a more effective aid for lip readers. In particular, ''pie slice'' and ray formats are evaluated against polygons. Filled shapes are evaluated against open shapes. Five-band speech spectra are evaluated against ten-band spectra. Display color is evaluated against line thickness for representing overall sound level.

4pSC7. A real-time audio–video front-end for multimedia applications. Dmitry Zotkin, Ramani Duraiswami, Ismail Hariatoglu, Larry Davis (Inst. for Adv. Computer Studies, Univ. of Maryland, College Park, MD 20742, ramani@umiacs.umd.edu), and Takahiro Otsuka (ATR Media Integration & Commun. Res. Labs., Kyoto, Japan)

A real-time system combining auditory input received by a microphone array with video input is designed and implemented as a prototype smart video-conferencing system. The audio part of the system uses accurate algorithms for localization of the source in typical noisy office environments, and performs dereverberation and beamforming to enhance signal quality. The coordinates of the sound sources obtained are used to control the pan, tilt and zoom of a video camera. The video-processing part segments people in the image using an obtained background model, and can label and track them and get a close-up view of the head of the talker. Heuristic algorithms for intelligent zoom and tracking of people in video-conferencing scenarios are incorporated into the system. The current implementation is performed on a commercial off-the-shelf personal computer equipped with a data acquisition board and does not require expensive hardware. The speed, precision and robustness of the system are superior to existing ones. Demonstrations of the system in different scenarios will be presented, along with quantitative measures of its performance. The system is designed as a general purpose front-end to systems for speech and person recognition, and for newer human-computer interfaces. [Work supported in part by DARPA.]

4pSC8. Examining processor and stimulus list differences in word recognition by children using a cochlear implant. Stefan Frisch (Prog. in Linguist., Univ. of Michigan, Ann Arbor, MI 48109-1285), Ted A. Meyer, and David B. Pisoni (Indiana Univ. Med. Ctr.)

Boothroyd and Nittrouer [J. Acoust. Soc. Am. **84**, 101–114 (1988), J. Acoust. Soc. Am. **87**, 2705–2715 (1990)] and Rabinowitz *et al.* [J. Acoust. Soc. Am. **92**, 1869–1881 (1992)] have demonstrated a power-law relationship between the accuracy in phoneme recognition and the accuracy in word recognition by normal hearing adults, older adults, children, and adults using cochlear implants. The power-law relationship provides a simple mathematical model of context effects on speech perception. An analysis of spoken word recognition performance by children using cochlear implants reveals an analogous power-law relationship between phoneme and word recognition. Further, comparison between groups using two different speech processors (SPEAK and MPEAK) shows that, while overall performance is affected by the processor, the power-law relationship is unaffected. By contrast, comparison between different stimulus lists finds the power-law relationship is affected by the phonotactic structure of the 20 lexical items used and their familiarity to the listener. [Work supported by NIH, AAO-HNS.]

4pSC9. Vocal tract size and the intelligibility of competing voices. Peter F. Assmann (School of Human Development, Univ. of Texas at Dallas, Richardson, TX 75083)

When two people speak at the same time, it is easier to understand what either is saying if the voices differ in fundamental frequency ($F0$). Talkers differ in vocal tract size as well as $F0$, and their formant frequencies occupy different ranges, providing a further basis for voice segregation. To investigate the effects of vocal tract size, a set of declarative English sentences was produced by an adult male and processed using a speech vocoder. Sentences were presented in pairs, with the spectrum envelope shifted up or down by a fixed percentage in one of the sentences. A +20% shift (sufficient to shift the formants of an adult male into the female range) led to a 6% increase in word recognition accuracy. A −20% shift led to a 9% drop in accuracy, consistent with listeners' informal reports that the ''larger'' voice sounded muffled. Benefits of upwards shifts and adverse effects of downward shifts were restricted to the ''shifted'' member of the pair. Upward spectral shifts were accompanied by increased spectral tilt, while downward spectral shifts led to reduced spectral tilt. Hence the observed effects may be due to spectral masking, rather than sensitivity to vocal tract size per se.

4pSC10. The role of modulation spectrum amplitude and phase in consonant intelligibility. Steven J. Aiken, Donald G. Jamieson, Vijay Parsa, and Prudence Allen (Hearing Health Care Res. Unit, UWO, School of Commun. Sci. and Disord., Elborn College, London, ON N6G 1H1, Canada)

This paper considers an acoustic basis for speech intelligibility and evaluates various acoustically based speech intelligibility prediction algorithms. Earlier research indicates that speech intelligibility does not require preservation of spectral and temporal fine-structure, but is highly dependent on the preservation of the amplitude component of the modulation spectrum [R. Drullman, J. Acoust. Soc. Am. **97**, 585–592 (1995)]. This study assessed the importance of the phase component of the modulation spectrum using a 21-alternative forced-choice consonant perception test. Temporal and spectral fine-structure were removed by modulating a white noise carrier with 50 Hz low-pass filtered speech amplitude envelopes in 4, 8, or 24 discrete bands. Modulation spectrum phase was distorted by imposing a random delay in each discrete band. Behavioral results are discussed in light of intelligibility predictions generated by the articulation index [N. R. French and J. C. Steinberg, J. Acoust. Soc. Am. **19**, 90–119 (1947)] and the speech transmission index [H. J. M. Steeneken and T. Houtgast, J. Acoust. Soc. Am **67**, 318–326 (1980)].

4pSC11. Assessment of intelligibility of digitized speech by young adults under different listening conditions. Ramesh N. Bettagere (Dept. of Special Education and Commun. Sci. and Disord., Campbell Hall, No. 217, SLU879, Southeastern Louisiana Univ., Hammond, LA 70402)

The present study evaluated the intelligibility of tape-recorded speech sentences and digitized speech sentences digitized at low (4 kHz), moderate (8 kHz), and high (16 kHz) sampling rates by young adults under different listening conditions (quiet versus noise). Twenty-four young adults participated as subjects. The tape-recorded speech sentences and digitized speech sentences were presented to each subject in quiet and in the presence of background speech babble at a signal-to-noise ratio of 0 dB. The subjects were instructed to immediately write down each sentence soon after they heard it. A 4×2 analysis of variance with repeated measures was performed to assess the effects of mode of speech and listening conditions on total number of words correctly transcribed. Results showed that the main effects for mode of speech and listening condition were statistically significant. The interaction effects of mode of speech by listening condition were also statistically significant. Pairwise comparisons showed that significantly more words were transcribed in the quiet condition than in the corresponding noise condition for all the modes of speech except for the digitized speech at high sampling rate (16 kHz). The implications of the results are discussed.

4pSC12. Vowel intelligibility in clear and conversational speech. Sarah Hargus Ferguson and Diane Kewley-Port (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405)

Several studies have demonstrated that instructions to speak clearly yield significant improvements in speech intelligibility along with a wide variety of acoustic changes relative to conversational speech. The current study explored the relationship between acoustic properties of vowels and their identification in clear and conversational speech, as well as the effects of hearing loss on this relationship. The goals were, for both normal and impaired listeners: (1) to determine what acoustic factors underlie the intelligibility differences observed between speaking styles and (2) to explore the relative contribution of various acoustic cues to vowel identification, using stimuli that vary naturally in intelligibility. Monosyllabic words were excised from sentences spoken either clearly or conversationally and presented in a background of 12-talker babble for vowel identification by young normal-hearing and elderly hearing-impaired listeners. Vowel identification performance was correlated with the results of acoustic analyses to assess the relative importance of spectral, durational, and dynamic cues to vowel perception. Preliminary analyses suggest that spectral target and dynamic formant information are primary cues to vowel identity, while the importance of duration is significant but small. [Work supported by NIHDCD-02229 and the ASHA Foundation.]

4pSC13. Effects of talker differences in vowel production on vowel identifiability. Amy Neel (Dept. of Audiol. and Speech Sci., Purdue Univ., West Lafayette, IN 47907-1353, atneel@purdue.edu)

Normal talkers differ in speech intelligibility, and vowel production characteristics contribute to intelligibility. This study investigated the relation between talker's acoustic vowel characteristics and the identification of their vowels by normal-hearing listners. Static and dynamic characteristics were measured for ten American English vowels producted in /dVd/ context by two female and two male talkers. Listeners identified four types of resynthesized vowels for each talker, ranging from nearly natural tokens with dynamic formants and appropriate duration values to relatively impoverished tokens containing only static formants and no duration cues. Significant differences in identification scores among talkers were found. The talker with the best-identified vowels, a male, had the largest and most dispersed vowel space with little overlap of formant trajectories in the dynamic $F1\times F2$ space. The worst talker, also male, had a smaller vowel space with greater overlap of formant trajectories. For all four talkers, dynamic-formant vowels were better identified than static-formant vowels, and there were greater differences among talkers for dynamic-formant vowels than for static-formant vowels. The results confirm that formant movement contributes to vowel identification and that greater separation in the dynamic $F1 \times F2$ space contributes to better vowel identifiability.

4pSC14. Dialectal differences in diphthong perception. Matthew J. Makashay (Dept. of Linguist., Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, makashay@ling.ohio-state.edu)

This study attempts to determine if speech perception varies across dialects as production does. Synthetic stimuli based on one talker from Binghamton, NY (northern US) and one from Birmingham, AL (southern US) were presented to subjects from both regions. The stimuli were 18 vowel continua in CVC context. These 10-point continua had initial tokens containing nonhigh vowels, such as *hot* /hat/, with formant values and durations manipulated to result in final tokens containing diphthongs, such as *hot* /hɪt/. Other continua contained endpoints such as *sad* /sæd/ and *side* /saɪd/, or *bought* /bt/ and *bout*/baʊt/. The production of diphthongs differs between these dialects, varying from the Canadian raising of the North to the monophthongization of the South. The question to be resolved in this study is whether perception of diphthongs differs between the dialects. It was found in a direct boundary estimation task that southern subjects perceived southern tokens as diphthongs earlier in the continua than northern subjects did, while there was no significant difference between these groups for the northern diphthongs. [This material is based upon work supported under a National Science Foundation Graduate Fellowship.]

4pSC15. The perception of noise vocoded prevocalic stop bursts and formant transitions. Michael Kiefte (Univ. of Alberta, Edmonton, AB, Canada)

Theories of prevocalic stop consonant perception differ with respect to the amount of spectral resolution required by the model, e.g., gross spectral shape theories predict that human speech perception is more robust against decreased frequency selectivity where detailed cues such as vocalic formant transitions break down. Previous results in the perception of noise vocoded speech [M. Kiefte, Int. Conf. Phon. Sci. (1999)] showed that, although listeners' performance worsened significantly at bandwidths as low as 500 Hz, a spectral shape model using cepstral coefficients was better able to predict subjects' responses. However, a simulation which showed that release bursts require much less spectral resolution than vocalic formant transitions complicates the interpretation. Therefore, in this experiment, subjects were asked to identify the place of articulation for gated release bursts independently of vocalic portions for both unprocessed and noise channel vocoded speech. Preliminary results show that correct identification of stop place from only the release burst does not change significantly up to a channel bandwidth of 2000 Hz. Results suggest that responses from the previous experiment can be explained by the robustness of the burst in those stimuli. Gross versus detailed cue models are also evaluated with respect to responses to formant only stimuli. [Work supported by SSHRC.]

4pSC16. Is the perceptual magnet effect a matter of auditory sensitivity? Sreedivya Radhakrishnan, John W. Hawks, and Robert J. Otto (School of Speech Pathol. and Audiol., Kent State Univ., Kent, OH 44242)

Efforts to validate and extend Kuhl's [P. K. Kuhl, Percept. Psychophys. **50**, 93–107 (1991)] perceptual magnet effect for vowel prototypes in adult listeners has been somewhat elusive and particularly difficult to replicate in other than the /i/ vowel space. However, there is converging evidence that suggests the magnet effect, attributed to assimilation by a prototype, may more likely be due to differences in auditory sensitivity across the vowel space. Hawks' [J. W. Hawks, J. Acoust. Soc. Am. **95**, 1074–1084 (1994)] data on discrimination of formant changes for 17 synthetic vowels supported this notion, indicating that difference limens (DLs) based from an exemplar token for /i/ was among the largest (poorest) found, while tokens neighboring /i/ yielded the smallest DLs. If the perceptual magnet effect is a demonstration of differences in auditory sensitivity and Hawks' data adequately reflects auditory sensitivity in the vowel space, then a likely location to find a magnet effect is within the space for /u/, which yielded the poorest DLs in Hawks' study. The results of an experiment investigating this possibility through replication of Kuhl's original protocol in the vowel space for /u/ will be presented.

4pSC17. Cues to vocal-fold and vocal-tract lengths interact in talker-sex perception from vowel segments. Michael J. Owren (Dept. of Psych., 224 Uris Hall, Cornell Univ., Ithaca, NY 14853) and Jo-Anne Bachorowski (Vanderbilt Univ., Nashville, TN 37240)

Even brief sounds can be richly informative about talker characteristics. Earlier acoustic analysis confirmed predictions that the best statistical classification of vowels by talker sex occurred using acoustic correlates of dimorphism in vocal-fold and vocal-tract lengths [J.-A. Bachorowski and M. J. Owren, J. Acoust. Soc. Am. (in press)]. In follow-up perceptual testing, 24 participants in experiment 1 classified talker sex from 180 short segments that balanced fundamental frequency ($F0$) and estimated vocal-tract length (VTL) cues. Responses were 98% correct, with a mean latency of 478 ms. However, latencies were longer for male talkers that combined high $F0$ and short VTL values, as well as for female talkers with low $F0$ and long VTL. In experiment 2, 24 participants heard stimuli composed of small numbers of waveform cycles. Classification ''threshold'' was 1.8 cycles, with male talkers being more accurately classified than females. These results show that indexical cueing is an inherent component of even the shortest possible vowel segments, and that the acoustic features most closely related to sexual dimorphism interact in predictable fashion in influencing perceptual processing. $F0$ and VTL, the particular acoustic cues in question, are fundamentally related to features also known to influence speech processing.

4pSC18. The perceptual consequences of overlap in /s/ and /ʃ/ productions within a talker. Sheryl A. Clouse, Jessica L. Burnham, and Rochelle S. Newman (Dept. of Psych., Univ. of Iowa, E11 Seashore Hall, Iowa City, IA 52242)

A primary issue in speech perception is the apparent lack of invariance between the acoustic information in a signal and the listeners perception. Different intended phonemes may be produced with identical acoustic values. Previously, we examined fricative centroids and frication peaks for over 100 utterances beginning with /s/ and /ʃ/ from each of 20 different speakers, and found substantial overlap across talkers. In the present study, we examine the effect of this overlap on perception. Listeners in a phoneme identification task heard natural productions of /s/ and /ʃ/ syllables from speakers with either little or great overlap between categories. Although labeling accuracy was near ceiling, effects were found in listeners' reaction times. Listeners needed more time to interpret the speech of talkers who had substantial overlap in their fricative centroids, even when these talkers showed no overlap in their frication peaks. Listeners also had slower reaction times for talkers with overlap in their frication peaks, but not their centroids. This suggests that category overlap has measureable consequences to perception, and that peaks and centroids are sensitive to different aspects of the speech signal, both of which are perceptually impotant to listeners. [Work supported by NIDCD Grant R01-DC00219 to SUNY at Buffalo.]

4pSC19. An fMRI investigation of feature distinctions in consonant identification. Tobey L. Doeleman (Cornell Phonet. Lab., Cornell Univ., Ithaca, NY) and Joy Hirsch (MSKCC/Cornell Univ., College of Medicine, New York, NY)

Differences in neural activation associated with the voicing and place distinctions of phonemes were investigated using functional magnetic resonance imaging (fMRI). Two types of synthetic stimuli were presented auditorily in a forced-choice identification task: clear CV syllables and CV syllables whose initial consonants were ambiguous with regard to either place or voicing. The stimuli consisted of the consonants [p, b, t, d, k, g] followed by the vowels [a, i, u]. Subjects were asked to match each auditory stimulus with one of two visually presented choices representing either voicing-feature or place-feature alternatives. Results consistently showed more widespread activation for all subjects in Wernicke's area (Brodmann's area 22) associated with determining the voicing feature versus the place feature of a phoneme. This finding is consistent with previous behavioral results [T. L. Doeleman, J. Acoust. Soc. Am. **101**, 3111(A) (1997)] showing more voicing than place errors in consonant identification and suggests a difference in the relative difficulty of these featural decisions. The fMRI design also allowed for a comparison of activiation associated with identifying clear versus ambiguous phonemes. Results show a trend of greater activation, especially in the right hemisphere, associated with the ambiguous stimuli.

4pSC20. Contrast effects with sinewave analog anchors. Robert A. Fox and Sarah England (Dept. of Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

An area of interest for several decades has been the effect of the immediate phonetic context on vowel perception—especially in terms of the phonetic contrast effect. One of the questions regarding this easily replicated effect is whether it is a phonetic-level effect or an acoustic-level effect. The current experiment was designed to determine whether the same contrast effect on the identification of a /i/–/I/ synthetic vowel continuum would be obtained with synthetic vowel (SV) anchors (representing one of the two endpoints) and with sinewave analog (SA) vowels (substituting tone glides for vowel formants). The stimuli included a nine-point /i/–/I/ SV continuum (produced with the Klatt program) and a correlated SA continuum. There were three basic anchoring conditions: anchoring (1) with the SV endpoints, (2) with the SA endpoints (subjects identified these tokens as tones), and (3) with the SA endpoints after subjects had been trained to identify them as vowels. Results showed a significant anchor effect with SV anchors (as expected) and no anchor effects with the SA anchor prior to training. Subjects did show a significant anchoring effect with the SA anchors after training, but the effect was significantly smaller than with SV anchors.

4pSC21. The effects of high-intensity speech on consonant feature transmission in normal-hearing subjects. Benjamin W. Y. Hornsby, D. Wesley Grantham, Ralph N. Ohde, Daniel H. Ashmead (Dept. of Hearing and Speech Sci., Vanderbilt Univ., 1114 19th Ave. South, Nashville, TN 37212, ben.hornsby@vanderbilt.edu), and Timothy D. Trine (Starkey Labs., Inc., Eden Prairie, MN 55300)

The effect of high-speech presentation levels on consonant recognition and feature transmission was assessed in normal-hearing subjects. Consonant recognition of C/i/ nonsense syllables was measured at five overall speech levels ranging from 65 to 100 dB SPL. Audibility remained constant by mixing speech stimuli with a speech-shaped noise at a constant 0-dB signal-to-noise ratio. Consistent with the work of others, overall percent correct performance decreased as the presentation level of speech increased [Studebaker *et al.*, J. Acoust. Soc. Am. **105**, 2431–2444 (1999)]. Confusion matrices were analyzed in terms of relative percent information transmitted at each speech presentation level, as a function of feature. Six feature sets (voicing, place, nasality, duration, frication and sonorance) were analyzed. Results showed the feature duration (long consonant duration fricatives) to be most affected by increases in level while the voicing feature was relatively unaffected by increases in level. In addition, alveolar consonants were substantially affected by level while palatal consonants were not.

4pSC22. Detection of consonant voicing: A module for a hierarchical speech recognition system. Jeung-Yoon Choi (Res. Lab. of Electron., MIT, Cambridge, MA 02139)

This research describes a module for detecting consonant voicing in a hierarchical speech recognition system. In this system, acoustic cues are used to infer values of features that describe phonetic segments. A first step in the process is examining consonant production and conditions for phonation, to find acoustic properties that may be used to infer consonant voicing. These are examined in different environments to determine a set of reliable acoustic cues. These acoustic cues include fundamental frequency, difference in amplitudes of the first two harmonics, cutoff first formant frequency, and residual amplitude of the first harmonic, around consonant landmarks. Classification experiments are conducted on hand and automatic measurements of these acoustic cues for isolated and continuous speech utterances. Voicing decisions are obtained for each consonant landmark, and are compared with lexical and perceived voicing for the consonant. Performance is found to improve when measurements at the closure and release are combined. Training on isolated utterances gives classification results for continuous speech that is comparable to training on continuous speech. The results in this study suggest that acoustic cues selected by considering the representation and production of speech may provide reliable criteria for determining consonant voicing.

4pSC23. Learning to discriminate reduced-channel /bɑ/–/wɑ/ syllables in normal-hearing adults. Cammy L. Bahner, Thomas D. Carrell, and T. Newell Decker (Commun. Disord., Univ. of Nebraska, Lincoln, NE 68583-0738)

Acoustic information delivered to individuals with cochlear implants differs in many ways from the same information delivered via a normally functioning cochlea. One difference is that the number of frequencies that can be distinguished is sharply reduced in implanted listeners. This results in poor frequency selectivity where only sounds that fall into different frequency bands are distinguishable (all other things being equal). Shannon [Science **270**, 303–304 (1995)] used an electrical filtering and rectification system to simulate this aspect of cochlear implant function in normal-hearing listeners. In the present study software filtering was combined with envelope-shaped-noise (ESN) processing to accomplish the same end. The immediate goal was to examine the efficacy of a simple technique in training listeners to distinguish between reduced-channel versions of syllables. The stimuli tested were synthetically produced /bɑ/ and /wɑ/ syllables, identical in every aspect except for their onset duration. These syllables were filtered into eight frequency bands and ESN processed. Following this they were once-again filtered at the same frequencies to create reduced-channel versions of the original syllables. The resulting stimuli were presented to 16 normal-hearing adult subjects in behavioral and electrophysiological tasks. Results demonstrated that the training rapidly improved listeners' abilities to identify reduced-channel /bɑ/ and /wɑ/.

4pSC24. Effects of first formant onset properties on voicing judgments of prevocalic stops without $F1$ cutback. José R. Benkí (Prog. in Linguist., Univ. of Michigan, 1076 Frieze Bldg., Ann Arbor, MI 48109-1285, benki@umich.edu)

This study examines the effects of first formant ($F1$) onset properties on voicing judgments of prevocalic stop consonants in contexts that do not exhibit the $F1$ cutback covariation present in pretonic and utterance-initial voicing contrasts. Previous research [K. R. Kluender and A. J. Lotto, J. Acoust. Soc. Am. **95**, 1044–1052 (1994)] strongly suggests that $F1$ onset frequency effects on voicing judgements are due to auditory factors, and not to listener experience with natural covariance with $F1$ cutback and $F1$ onset frequency. The present study extends the previous findings to voicing contrasts in contexts that are not signaled by $F1$ cutback, such as before unstressed vowels in English. Categorization data were collected for a continuum between *rabid* (/b/) and *rapid* (/p/) in which both closure duration and $F1$ onset properties were manipulated using LPC resynthesis of natural speech. Preliminary results indicate that lower $F1$ onset frequencies of the unstressed vowel condition more voiced percepts of the preceding stop consonant, consistent with findings for utterance-initial stop consonants.

4pSC25. Effects of stimulus uncertainty, consonantal context and training on formant frequency discrimination. Diane Kewley-Port (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405, kewley@indiana.edu)

Words in natural speech vary in their predictability, e.g., stimulus uncertainty ranges from low to very high across sentences. The ability to discriminate formants is strongly affected by longer phonetic context [Kewley-Port and Zheng (in press)]. This study investigates the effects of stimulus uncertainty from minimal to high uncertainty and the phonetic contexts /V/ or /bVd/. The primary formants were from four female vowels /ɪ, ɛ, æ ə/. Using adaptive tracking, DLs for discriminating a small change in a formant was calculated in Delta Barks. In experiment 1, performance for five listeners, optimized by extensive training, began with minimal uncertainty, subsequently increasing uncertainty from 8- to 16- to 22 formants per block. Effects of higher uncertainty were less than expected, only decreasing performance by about 33%, although DLs for CVCs were 25% poorer than for isolated vowels. In experiment 2, performance in the 22-formant condition was tracked over 1 h for 37 listeners without formal laboratory training. DLs for untrained listeners were about 300% worse than for trained listeners, and comparable to untrained listeners in Kewley-Port and Zheng (in press). Results indicate that the effects of longer phonetic context degrade formant frequency discrimination more than higher stimulus uncertainty. [Work supported by NIHDCD-02229.]

4pSC26. Perception of stress and speaking style for selected elements of the SUSAS database. Robert S. Bolia and Raymond E. Slyh (Air Force Res. Lab., 2255 H St., Wright–Patterson AFB, OH 45433)

The SUSAS database [J. H. L. Hansen and S. E. Bou-Ghazale, EUROSPEECH **97**, 1743–1746 (1997)] is a collection of utterances recorded under conditions of simulated or actual stress, the purpose of which is to allow researchers to study the effects of stress and speaking style on the speech waveform. The aim of the present investigation was to assess the perceptual validity of the simulated portion of the database by determining the extent to which listeners classify its utterances according to their assigned labels. Seven listeners performed an eight-alternative, forced-choice, judging whether monosyllabic or disyllabic words spoken by talkers from three different accent classes (Boston, Generic Midwest, New York) were best classified as ''angry,'' ''clear,'' ''fast,'' ''loud,'' ''neutral,'' ''question,'' ''slow,'' or ''soft.'' Mean percentages of ''correct'' judgments were analyzed using a 3 (accent class)$\times$2 (number of syllables)$\times$8 (speaking style) repeated measures analysis of variance. Results indicated that, overall, listeners correctly classify the utterances only 58% of the time, and that percentage of correct classifications varies as a function of all three independent variables.

4pSC27. Prosodic disambiguation of syntactic ambiguity in discourse context. Shari R. Speer, Shari B. Sokol (Univ. of Kansas, 3031 Dole, Sp.-Lang.-Hear., Lawrence, KS 60045), Amy J. Schafer (UCLA, Los Angeles, CA), and Paul Warren (Wellington Univ. of New Zealand)

A cooperative boardgame task was used to examine how native speakers use prosodic structure to resolve syntactic ambiguity in discourse context. The game task required two speakers to use utterances from a predetermined set to negotiate the movement of gamepieces to goal locations. In one condition, the discourse contained two situations that had to be described using the same syntactically ambiguous word sequence. In the other condition, an identical syntactically ambiguous structure was used to describe only one situation. Sentences that could describe two situations had an ambiguous prepositional phrase attachment as in ''I want to move the square with the triangle,'' in which the move involved either a combined square-and-triangle piece or a triangle pushing a square to another position. Sentences describing only one situation involved only a cylinder pushing the square as in ''I want to move the square with the cylinder.'' Phonological analyses and phonetic analyses of duration and fundamental frequency were compared. Results indicate that speakers used prosodic phrasing to reflect situational syntactic ambiguity. Examples of relatively high and low variability in production will be discussed. [Work supported by NIH grant MH-51768, NZ/USA Cooperative Science Programme grant CSP95/01, Marsden Fund grant VUW604, and NIH research DC-00029.]

4pSC28. What is a neighbor in a neighborhood effect? Items that mismatch on the first phoneme still produce neighborhood effects. Rochelle S. Newman (Dept. of Psych., Univ. of Iowa, E11 Seashore Hall, Iowa City, IA 52242, rochelle-newman@uiowa.edu), James R. Sawusch, Paul A. Luce (State Univ. of New York at Buffalo, Buffalo, NY 14260), and Alicia Healy (Univ. of Iowa, Iowa City, IA 52242)

Previously, we presented results suggesting that neighborhood density could influence perception in phoneme identification tasks. In a typical task demonstrating this effect, subjects might hear two series, one ranging from beysh to peysh, while the other varied from beyth to peyth, where beyth is similar to more real words than peyth, and peysh is similar to more real words than beysh. Subjects were more likely to classify the ambiguous stimuli from each series as members of the category which makes it more wordlike. In the present series, similar series were created where the only neighbors for the endpoints were ones that differed in their initial consonant. We created two pairs of series: in the first, ʃoif–tʃoif and ʃof–tʃof, the neighbors which drove the bias were all ones which matched on the initial phoneme (e.g, choke). In the second series, zUʃ–zUtʃ and zEʃ–zEtʃ, the only neighbors for the items were ones that mismatched on the initial phoneme (e.g., push, mesh, fetch, etc.). Similar neighborhood effects were found for both series, suggesting that words in memory which do not match the beginning of a perceived item are still activated and can still influence perception.

4p THU. PM

4pSC29. Inhibition in phonological priming: Lexical or strategic effects? Lisa C. Shoaf and Mark A. Pitt (Dept. of Psych., The Ohio State Univ., 1885 Neil Ave., Columbus, OH 43210, contos.1@osu.edu)

Facilitatory (speeded) and inhibitory (slowed) response times are found in phonological priming experiments, in which the amount of word-initial phoneme overlap between a prime and target is varied (e.g., mark–must). While facilitation appears to be strategic, there is debate as to the nature of the inhibitory priming found when high-overlap prime-target pairs are used (e.g., musk–must). Some researchers propose that this inhibitory priming is due to lexical competition between simultaneously activated candidates; others suggest it is due to the use of a response strategy. Experiments in our laboratory attempted to resolve this debate. We tested for the presence of strategic effects by manipulating variables thought to influence strategy acquisition (i.e., isi, proportion of overlap trials), and then by examining participants' RTs to trials of varying overlap (e.g., prone–must, mark–must, muff–must, musk–must) collected over the course of the experiment. Results suggest that inhibition in the phonological priming paradigm is determined in large part by the development of a response strategy.

4pSC30. Phonemic effects in spoken-word recognition in Japanese. Anne Cutler (MPI for Psycholinguist., P.O. Box 310, 6500 AH Nijmegen, The Netherlands, anne.cutler@mpi.nl) and Takashi Otake (Dokkyo Univ., Soka, Saitama, 340 Japan)

Previous studies have shown that Japanese listeners are sensitive to the moraic structure of speech, and find it easier to manipulate or respond to morae than phonemes. We further examined moraic processing via two word reconstruction experiments, in which Japanese listeners heard three- or four-mora nonwords which could be changed into real words by substitution of a single mora. In experiment 1, listeners had to change the first mora of the nonword, in experiment 2 the final mora. We compared three types of substitution for CV morae: substitution preserving the C (e.g., *kimera* or *kamere* for, respectively, the first and last mora of the word *kamera*, which has three morae: ka-me-ra), substitution preserving V (*namera, kamena*), or substitution preserving neither (*nimera, kamene*). When C or V was preserved, responses were significantly faster and more accurate than when neither was preserved. In initial position, there was no difference between C- and V-preserving substitutions, but in final position, preservation of the C led to faster and more accurate responses than preservation of the V. These results confirm that spoken word recognition in Japanese is sensitive to vocabulary structure and similarity (inter alia at a submoraic level) between words.

4pSC31. Spoken word recognition and the influence of lexical competitors at the beginning of a word. Nadia Duenas and Michael Vitevitch (Speech Res. Lab., Dept. of Psych., Indiana Univ., Bloomington, IN 47405, mvitevit@indiana.edu)

Marslen-Wilson and Welsh (1978) suggest that the initial portion of a word activates multiple lexical candidates in memory, forming a ''cohort'' of competitors. Marslen-Wilson (1987) further states that the number of competitors in the cohort does not affect the speed and accuracy of spoken word recognition. Monosyllabic CVC words with the same number of lexical competitors (using a substitution-based metric to estimate competitor set size) but varying in the percentage of competitors sharing the same initial phoneme as the target word were presented in an auditory naming task and an auditory lexical decision task. Words with a smaller proportion of competitors sharing the same initial phoneme as the target word were responded to more quickly than words with a larger proportion of competitors sharing the same initial phoneme as the target word. These results suggest that the number of candidates activated in memory does affect spoken word recognition, and that the initial portion of a word is important in processing. The implications of these results for models of spoken word recognition are discussed. [Work supported by NIH-NIDCD Training Grant DC-00012.]

4pSC32. The basic units of rate normalization. Jessica L. Burnham and Rochelle S. Newman (Dept. of Psych., Univ. of Iowa, E11 SSH, Iowa City, IA 52242, rochelle-newman@uiowa.edu)

We investigated the units of rate normalization, and found that they are not based on phonemes, but on segments with obvious acoustic boundaries, emphasizing the role of basic auditory processing in speech recognition. Individuals vary their speaking rate, and listeners use the duration of adjacent segments to adjust for these changes [J. Miller and A. Liberman, Percept. Psychophys. **25**, 457–465 (1979)]. However, it has not been clear what these segments actually are. We examined whether two-phoneme sequences would produce two separate rate normalization effects (one for each phoneme), or only a single effect. We first used the series shkas–chkas, and examined effects of /k/ and /a/ duration on perception of the initial contrast. Altering the /k/ duration resulted in a rate normalization effect separate from that caused by varying the duration of the sequence as a whole. This suggests /k/ and /a/ are treated as separate units of speech. A /w/+vowel sequence showed a different result. For sequences such as ka, with obvious acoustic boundaries, each phoneme has a separate rate normalization effect. However, for sequences such as w, without such obvious cues, the sequence is treated as a single (larger) segment by the rate normalization process.

4pSC33. The effect of amplitude modulation on periodic- and aperiodic-formant-based sentences. Thomas D. Carrell, Camille C. Dunn, and Jennifer Gutzwiller (Commun. Disord., Univ. of Nebraska, Lincoln, NE 68583)

Amplitude modulation has been shown to increase the intelligibility of tone-analog sentences by 30%–60% [T. D. Carrell and J. Opie, Percept. Psychophys. **52**, 437–445 (1992)]. Tone-analog sentences are constructed from three or four time-varying sinusoidal waveforms that trace the formant frequencies of the first several formants of a natural utterance [Remez *et al.*, Science **212**, 947–950 (1981)]. Because these sentences have no fundamental frequency an important auditory grouping cue is missing. It has been argued that the observed intelligibility increments were based on the ability of amplitude modulation to supply auditory grouping cues similar to those missing from tone-analog sentences. This explanation was tested with whispered sentences and with reduced-channel envelope-shaped noise (ESN) sentences [C. L. Bahner and T. D. Carrell, 138th meeting Acoust. Soc. Am. (1999)]. In an experiment comparing the effect of amplitude modulation on tone-analog sentences, whispered sentences, and reduced-channel ESN sentences, it was found that the intelligibility of tone-analog sentences was improved whereas the intelligibility of the other two types of sentences was reduced. This pattern of results calls for a distinction between the effects of amplitude modulation on sentences constructed with periodic-based formants versus those constructed with aperiodic-based formants.

4pSC34. Interpreting Garner interference. Shawn Weil and Mark Pitt (Dept. of Psych., Ohio State Univ., 1827 Neil Ave., Columbus, OH 43210, weil.17@osu.edu)

The Garner (1974) speeded classification procedure has long been used in psychoacoustic, speech perception, and music perception research to assess the relative integrality or separability of stimulus dimensions. Interference between dimensions is indicated by a drop in performance between an orthogonal condition (two dimensions varied independently) and a control condition (one dimension held constant, the other varied). In a meta-analysis, the ubiquity of Garner interference was examined to understand better the meaning of interference and the theoretical claims that can be drawn from it. Within this context, a set of experiments revisited the source of interference found between talker and phonetic dimensions (Mullennix and Pisoni, 1990). Individual speaker variability (simulated by multiple single-speaker tokens) and word-initial consonant (/b/ vs /p/) were manipulated.

4pSC35. Laughter: Perceptual evaluations of laugh sounds with known acoustic variability. Jo-Anne Bachorowski (Dept. of Psych., Vanderbilt Univ., Nashville, TN 37240) and Michael Owren (Cornell Univ., Ithaca, NY 14853)

Acoustic analysis of laughter recorded under controlled social contexts revealed significant variability on a number of source- and filter-related acoustic measures. Here, the results of three experiments used to test listeners' ($n=84$) evaluations of this acoustic variability are described. The 80 laugh stimuli used in each experiment were selected on the basis of laugher sex, percentage of voicing, F0 contour, and overall laugh duration. Listeners evaluated each stimulus twice using one of three 4-point rating schemes (goodness for inclusion in a laugh track, sincerity, and the extent of positive emotional responding). The results indicate that listeners prefer laughs with voiced, song-like qualities, and suggest that particular patterns of acoustic features in this nonlinguistic human vocal signal are more likely to engender positive emotional responses in listeners. Further analyses will examine the links between listener evaluations, variation in laugh vowel sounds, and acoustic correlates of laugher body size. [Work supported by NSF.]

THURSDAY AFTERNOON, 4 NOVEMBER 1999 GARFIELD ROOM, 1:00 TO 2:45 P.M.

Session 4pSPa

Signal Processing in Acoustics: Signal Processing I: Multi-Channel Techniques

John Impagliazo, Chair

Naval Undersea Warfare Center, Code 8212, Building 679, Newport, Rhode Island 02841

Contributed Papers

1:00

4pSPa1. Exact solutions for the problem of source location from measured time differences of arrival. Ramani Duraiswami, Dmitry Zotkin, and Larry Davis (Inst. for Adv. Computer Studies, Univ. of Maryland, College Park, MD 20742, ramani@umiacs.umd.edu)

Source location using the difference in time of arrival of a signal (TDOA) at an array of receivers is used in many fields including speaker location. Using three TDOA values from four noncollinear receivers one can, in principle, solve for the unknown source coordinates in terms of the receiver locations. However, the equations are nonlinear, and in practice signals are contaminated by noise. Practical systems often use multiple receivers for accuracy and robustness, and improved S/N, and solutions must be obtained via minimization. A new family of exact solutions, for the case of four receivers located in a plane, is presented. These solutions can be evaluated using a small number of arithmetic operations. The performance of these solutions for several practical situations is examined via simulation, and possible geometries of receiver locations suggested. For the case where multiple receivers (>4) are used, a new formulation is presented, that incorporates the present solutions, imposes additional constraints on source location, and enables use of a constrained L_1 optimization procedure to achieve a robust estimate of the source location. The present estimator is compared with ones from the literature, and found to be robust and accurate, and more efficient. [Work partially supported by DARPA.]

1:15

4pSPa2. Blind estimation of the relative travel times and amplitudes of multipath using auto- and cross-correlation functions. John L. Spiesberger (Dept. of Earth and Environ. Sci., Univ. of Pennsylvania, 3440 Market St., Ste. 400, Philadelphia, PA 19104-3325, johnsr@sas.upenn.edu)

A location problem is considered where sound propagates along multipath which are impractical to model because the environment is poorly known. The acoustic bandwidth is assumed to be large enough so that the cross-correlation functions between pairs of receivers contain multiple peaks from multipath. The highest peak may not correspond to the difference in path lengths between the source and the receivers. Using similarities in the patterns of peaks in auto- and cross-correlation functions, an algorithm is developed to identify which cross-correlation peak corresponds to the difference in first arrivals, which can be used for locating the source if these arrivals are straight. The similarities are expressed with new ''correlation equations.'' The number of lag-type correlation equations is $O(\mathcal{R}^2N^2)$, where N is the typical number of multipath at each of $\mathcal{R}$ receivers. The correlation equations may be impractical to solve exactly. Accurate solutions are found in simulations for the numbers, relative travel times, and amplitudes of all the multipath with the aid of a new fourth-moment function which is a cross correlation of non-negative lags of an auto-correlation function with lags from a cross-correlation function. The technique relies on time series which are filtered to yield one dominant source.

1:30

4pSPa3. Race car trajectory determination by multi-source acoustic emission analysis. Matthew Barga, Yann G. Guezennec, and Giorgio Rizzoni (Dept. of Mech. Eng. and Ctr. for Automotive Res., The Ohio State Univ., 206 W. 18th Ave., Columbus, OH 43210-1107, guezennec.1@osu.edu)

This talk presents a signal processing methodology to extract racecar trajectory and engine speed information from a multiple microphone array placed at track side. Phase differences, as well as Doppler shifts of the engine acoustic emissions from an array of microphones, provide multiple sources from which a robust determination of the vehicle trajectory can be extracted. Furthermore, time-frequency analysis of the Doppler-corrected signals provide a rich source of information about the instantaneous engine speed. The combination of both methods provides valuable information about the vehicle and engine characteristics. The analysis method used in this work is applied to acoustic emission data recorded by a trackside array of four microphones. The data analyzed include synthetic validation and calibration data, as well as actual race data. Special attention was devoted to optimizing the microphone array configuration for maximum resolution given an array size. By carefully exploiting the inherent redundancies in the data, a robust and computationally efficient analysis method was successfully developed. The results demonstrate that valuable vehicle and engine parameters can be extracted from such an approach.

1:45–2:00 Break

2:00

4pSPa4. Normalizing inter-band scaling in minimum variance (MV) beamformer estimates of variable-bandwidth filtered wideband signals. Laura A. Drake, Aggelos Katsaggelos (Elec. & Computer Eng. Dept., Northwestern Univ., Evanston, IL 60208-3118), and Jun Zhang (Univ. of Wisconsin-Milwaukee, Milwaukee, WI 53201)

Narrow-band minimum variance (MV) beamforming can be used to estimate a wideband signal when the signal is first filtered into several narrow-band signals. This paper identifies and solves a new problem for narrow-band MV beamforming of wideband signals. ''Inter-band scaling''

can occur when narrow-band MV beamforming is applied to a wideband signal filtered by a variable-bandwidth filterbank. This situation can occur when the wideband signal to be estimated is speech. Then, it is often desireable to use a filterbank with higher-frequency resolution in the low, than in the high-frequency bands since speech has more information in low frequency than high. In this case, the power out of the MV beamformer in each band can depend not only on the actual signal power in that band, but also on the bandwidth. The inter-band scaling normalization method presented here is tested with both white Gaussian noise, and a segment of a speech signal. The tests show that the method is effective and does not distort the speech signal. Finally, this method should be extendable to other problems (such as other adaptive array processing methods) that require estimates of statistical measures of variable-bandwidth filtered signals.

2:15

4pSPa5. A spread spectrum technique for studying sound propagation in forested areas. Michelle E. Swearingen and David C. Swanson (Grad. Prog. in Acoust., P.O. Box 30, State College, PA 16804, swear@sabine.acs.psu.edu)

Spread spectrum methods are relatively new to acoustics, but have been shown to be quite useful, especially in determining the path taken by a particular sound pulse. Because of this property, spread spectrum signals should give a rough picture of the scattering off tree boles that occurs in a forest. The arrival times will determine the path taken, and the level changes will determine how much attenuation that path contributes. The experimental setup and preliminary data will be presented.

2:30

4pSPa6. Constrained maximum likelihood estimation of the principal curvatures on the object surface in an ultrasonic tactile sensing. Kenbu Teramoto (Dept. of Mech. Eng., SAGA Univ., SAGA, 8408502 Japan) and Noriko Mori (SAGA Univ., SAGA, 8408502 Japan)

The maximum likelihood estimation (MLE) provides a robust solution in matched-field imaging (MFI), which has evolved from various underwater acoustic applications successfully. In the identification of the principal curvatures during the medical or robotic applications of acoustic tactile sensing, matched field processing offers a reasonable detection of the reflected wavefront. Two major difficulties, however, are there in the principal curvature identification. The first reason is that the wavefront reflected by the elliptic paraboloidal or the hyperbolic paraboloidal surface cannot be described in the combination of the plane waves nor the spherical waves strictly. The second is that the principal curvature identification process becomes ill posed due to the nonlinear relationship between the principal curvatures and propagation time of flight. The MFI scenario, therefore, can solve the nonlinear optimization problem in order to identify the curvature. In this paper, the proposed identification algorithm seeks the unique KKT (Karush–Kuhn–Tucker) point in the augmented Lagrange function of the constrained likelihood function, which is defined over the observed signal field. Furthermore, several acoustical experiments show that the proposed tactile sensor can identify the principal curvatures of the following surface cases: (1) plane, (2) paraboloid, (3) elliptic paraboloid, and (4) hyperbolic paraboloid.

THURSDAY AFTERNOON, 4 NOVEMBER 1999 GARFIELD ROOM, 3:00 TO 4:15 P.M.

Session 4pSPb

Signal Processing in Acoustics: Signal Processing II: Applications in Acoustics

Charles F. Gaumond, Chair
Naval Research Laboratory, Code 7142, Acoustics Division, Washington, DC 20375-5320

Contributed Papers

3:00

4pSPb1. A minimum-variance frequency-domain algorithm for binaural hearing aid processing. Michael E. Lockwood, Douglas L. Jones, Mark E. Elledge, Robert C. Bilger, Marc Goueygou, Charissa R. Lansing, Chen Liu, William D. O'Brien, Jr., and Bruce C. Wheeler (Beckman Inst., Univ. of Illinois at Urbana–Champaign, 405 N. Mathews Ave., Urbana, IL 61801)

A new algorithm has been developed that allows optimal filtering methods to be applied individually to different narrow frequency bands. Using this technique it is possible to process binaural signals in a manner which dramatically reduces the amplitude of signals originating away from a desired receive direction. The algorithm was tested on artificially combined anechoic and reverberant signals with varying numbers of interfering sound sources. In addition to being computationally efficient, the algorithm was able to produce output which had a consistently positive intelligibility weighted SNR gain [Link and Buckley, J. Acoust. Soc. Am. **91**, 1662 (1992)], and in which the desired talker was noticeably easier to understand. Results of a paired-comparison listening test confirmed these results, and allowed an estimate to be made of the algorithm parameters which provided the best speech intelligibility for the output. These parameters included the window length, length of a filtered block, and amount of data which must be stored in memory. Collectively, the results show that this algorithm may be the first building block for a binaural hearing-aid processing algorithm which suppresses off-axis interference. [Research supported by the Beckman Institute.]

3:15

4pSPb2. The effect of reverberation on a new binaural noise cancellation algorithm for hearing aids. Marc Goueygou, Michael E. Lockwood, Mark E. Elledge, Robert C. Bilger, Douglas L. Jones, Charissa R. Lansing, Chen Liu, William D. O'Brien, Jr., and Bruce C. Wheeler (Beckman Inst., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801)

A new algorithm was developed to emulate the ''cocktail party effect'' in hearing aids [Liu *et al.*, ASA Symposium (1997)]. Speech signals are received binaurally; the interferers and the desired source are localized; and the signal from the strongest interferer is cancelled by frequency and time selective beamforming. Here we evaluate the algorithm's performance in real conditions including reverberation. Binaural recordings of speech uttered simultaneously by three different speakers around the receivers were made in three different rooms (an anechoic room, a mildly and a highly reverberant room). Noise cancellation was evaluated by the intelligibility weighted SNR (IWSNR) gain between processed and unprocessed signals [Link and Buckley, J. Acoust. Soc. Am. **91**]. Algorithm performance is severely degraded as the reverberation time increases: the IWSNR gain falls from 8–9 dB in the anechoic room to 1–2 dB in the highly reverberant room. Also, knowing of the exact location of the

sources does not add significant improvement. The results suggest that the room impulse response be deconvolved from the received signals to suppress the multiple reflections of sound. We suggest a new strategy combining noise cancellation with blind identification of the room impulse response. [Research supported by the Beckman Institute.]

3:30

4pSPb3. A real-time dual-microphone signal-processing system for hearing aids. Mark E. Elledge, Michael E. Lockwood, Robert C. Bilger, Marc Goueygou, Douglas L. Jones, Charissa R. Lansing, William D. O'Brien, Jr., and Bruce C. Wheeler (Beckman Inst., Univ. of Illinois at Urbana–Champaign, 405 N. Mathews Ave., Urbana, IL 61801)

A real-time dual-microphone-based signal-processing system has been developed that suppresses off-axis interference from multiple sound sources. The system employs optimal filtering methods to attenuate the amplitude of all signals other than that from a desired receive direction. The algorithm was implemented on a Texas Instruments TMS320C62 fixed-point digital signal-processor evaluation board installed in a PC. Input/output was accomplished via a 16-bit codec controlled by optimized assembly language routines. To simplify algorithm coding, the main algorithm was written in C and executes using approximately 55 processor time, allowing for future additions to enhance performance. It has a delay of 15 ms. The system can significantly improve the intelligibility of speech in noisy environments, as evidenced in off-line tests by a marked intelligibility weighted SNR gain [Link and Buckley, J. Acoust. Soc. Am. **91**, 1662 (1992)]. Initial real-time testing with actual binaural signals supports the conclusion that this performance gain will be maintained. The results show that this system may be used as a main building block for hearing aids to be used in noisy environments. [Research supported by the Beckman Institute.]

3:45

4pSPb4. Multimedia model for speech transmission through walls. Jiann-Ming Su and J. H. Ginsberg (The George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The paper will describe how an algorithm previously developed to study the distortion of a planar stress wave as it propagates at normal incidence through a multi-layered wall was adapted to study transmission of speech signals. The algorithm is a time domain numerical implementation of the method of characteristics [Ginsberg and Kim, J. Appl. Mech. **11**(2), S145–S151 (1992)]. The algorithm is used to produce a digitized waveform of the exiting signal corresponding to an incident signal consisting of a speech fragment. To assess the effects of distortion resulting from internal reflections, a speech pattern is digitized with the aid of standard multimedia tools and used as the input to the transmission algorithm. The transmitted signal is computed and converted to a convenient audio format for playback. To ensure that distortion in the output signal is truly the effect of the medium, the signal is processed again using a noise reduction routine developed by Boll [IEEE Trans. ASSP **ASSP-27**(2), 113–120 (1979)]. In a blind test, listeners are requested to identify, from both the raw and processed output, the original speech fragment. A demonstration will conclude the presentation.

4:00

4pSPb5. Further development of time delay spectrometry. Bradley Finch, Gareth Cook, and Anthony Zaknich (Ctr. for Intelligent Information Processing (CIIPS), Univ. of Western Australia, Nedlands 6009, Australia, finch@ee.uwa.edu.au)

The accuracy of conventional time delay spectrometry (TDS) measurements are limited by constraints on signal parameters. These constraints ultimately limit the resolution with which the frequency response can be measured. Poletti, Cook, and others have shown that more exact measurements, without these constraints, are possible. These theories require that the system response to the full complex chirp be known. This is usually done by exciting the system with two separate orthogonal sweep signals. In this paper, this theory is developed in a manner similar to that done by Vanderkooy for the conventional TDS measurement, to show that the more exact system response can be deduced from a single linear sweep. The new development is supported by experimental results comparing the measured system response of a series resonant circuit against the conventional TDS results originally reported by Vanderkooy. This method provides practitioners more convenience in making exact TDS measurements, and allows for further exploration of the application of this technique in time varying environments.

THURSDAY AFTERNOON, 4 NOVEMBER 1999 — FAIRFIELD ROOM, 1:30 TO 5:15 P.M.

Session 4pUW

Underwater Acoustics, Acoustical Oceanography and Animal Bioacoustics: The Effect of Man-Made Sound on Marine Mammals II

Peter L. Tyack, Chair

Department of Biology, Woods Hole Oceanographic Institution, 45 Water Street, Woods Hole, Massachusetts 02543-1049

Chair's Introduction—1:30

4p THU. PM

Invited Papers

1:35

4pUW1. Acoustic responses of Baleen whales to low-frequency, man-made sounds. Christopher W. Clark (Bioacoust. Res. Prog., Cornell Univ., Sapsucker Woods Rd., Ithaca, NY 14850), Peter L. Tyack (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), and William T. Ellison (Marine Acoustics, Inc., P.O. Box 340, Litchfield, CT 06759)

In the last 5 years, two projects were undertaken to evaluate impacts of man-made sounds on whales. Baleen whales were identified as ''at risk'' because of their use of low-frequency sound for communication, their endangered status, and studies showing responses to continuous noises at exposure levels >120 dB, *re*: 1 μPa. To evaluate the potential impact of operational ATOC (195-dB source intensity), humpback whales off Kauai were studied during the breeding season. To evaluate the potential impact of

U.S. Navy SURTASS LFA sonar, playback experiments were conducted on four species at exposures of 120–155 dB. LFA research was designed to obtain responses during feeding (blue and fin whales, southern California, September–October), migration (gray whales, central California, January) and breeding (humpbacks, Hawaii, March). For the ATOC source, humpbacks showed statistically significant but subtle responses over small time and spatial scales. There were no changes in singing, or larger-scale changes in distribution or relative numbers. For LFA experiments off southern California, whales did not change vocal rates or leave the testing area, and there were no immediately observable responses, even at exposure levels up to 150 dB. Tyack (this session) will discuss results from the gray and humpback whale LFA experiments.

2:05

4pUW2. Responses of Baleen whales to controlled exposures of low-frequency sounds from a naval sonar. Peter L. Tyack (Biol. Dept., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, ptyack@whoi.edu)

Playback experiments were conducted from 8–27 January 1998 using a moored SURTASS-LFA sound source to study behavioral responses of gray whales migrating off Pt. Buchon, California. Shore stations operated for over 150 h during 18 days, tracking about 1200 migrating whales. When the source was moored inshore near most migrating whales, whales avoided exposure to 42-s sound stimuli in the 160–330-Hz frequency band repeated every 6 min at received levels near 120 dB (playback source levels 170–185 dB *re*: 1 μPa at 1 m). Responses to the source broadcasting offshore at source levels of 185–200 dB were greatly reduced. Playbacks to humpback whales singing in Hawaiian waters occurred in February–March 1998. 17 playbacks and 5 control follows of singing humpbacks were conducted from a small vessel. Singers continued to sing throughout 7/17 playbacks, and stopped during the remaining 10 playbacks. During 4/10 playbacks when singers stopped, the singer stopped when it joined with another whale. The remaining 6 cessations of song were considered possible responses to playback. Most of these whales resumed normal behavior before the hour-long playback ceased. There is no trend for the vocal response to playback to scale with exposure level; both the ''vocal-response'' and ''no-vocal-response'' categories ranged in exposure from 120–150 dB; *re*: 1 μPa.

2:35

4pUW3. Marine mammal research program for the Pioneer Seamount ATOC experiment. Daniel Costa (Inst. of Marine Sci., Univ. of California, Santa Cruz, CA 95064) and John Calambokidus (Cascadia Research Collective, Olympia, WA)

Aerial surveys and satellite and archival tags were used to test the effect of the ATOC sound source on marine mammals around the Pioneer Seamount 85 km west of San Francisco, California. Control surveys were flown at least 48 h after the end of any previous transmission cycle and experimental surveys were flown after at least 24 h of sound transmissions. Sound transmissions consisted of 20-min periods of 195-dB, that is, 1-μP transmission repeated very 4 h. Most commonly sighted species by group were: (1) mystecetes: humpback whale (372 sightings), (2) large odontocetes: sperm whale (337 sightings), (3) small odontocetes: Pacific white-sided dolphin (306 sightings), and (4) pinnipeds: California sea lion (167 sightings). Although there were no significant differences in the number of sightings when the sound source was on or off, both humpback and sperm whales were generally seen farther from the sound source during experimental versus control surveys ($p<0.01$). The highest intensity of sound measured on tags carried by elephant seals during transmissions ranged from 118–137 dB for 60–90 Hz compared to ambient levels of 87–107 dB (60–90 Hz). On a gross level, animals did not alter return track or go to the surface, and often continued to dive closer to the sound source if on the descending segment of a dive evident. Some animals with the highest levels of exposure showing no effects while two animals showing minor changes in diving pattern had exposure levels greater than 132 dB. [Work supported by the Strategic Environmental Research and Development Program through RPA and the Office of Naval Research.]

3:05

4pUW4. The influence of seismic survey sounds on bowhead whale calling rates. Charles R. Greene, Jr. (Greeneridge Sci., Inc., 4512 Via Huerto, Santa Barbara, CA 93110), Naomi S. Altman (Cornell Univ., Ithaca, NY 14853), and W. John Richardson (LGL Ltd., Environ. Res. Assoc., King City, ON L7B 1A6, Canada)

During September in 1996–98, airgun arrays operated close to the coast of the Alaskan Beaufort Sea. It was of interest to know if the airgun pulse sounds affected migrating whales. Autonomous seafloor acoustic recorders were installed at various distances from shore and from the airguns. They recorded continuously for up to 22 days. Both the bowhead calls and the airgun pulses were clustered in time, requiring a greater quantity of data to isolate an influence than would be the case with nonclustered data. The consistent results from the three years were: (1) bowhead whales called frequently during their migration through the study area; (2) calling continued, although possibly at different rates, when the whales were exposed to airgun pulses; and (3) call detection rates at some locations differed significantly when airgun pulses were detectable versus not detectable. At least in 1996, patterns of call detection rates were consistent with aerial survey evidence that most bowheads were displaced from the area near the operating airguns. However, there was no significant tendency for call detection rates to change consistently at times when airgun operations started or stopped. [Work supported by BP Exploration (Alaska) and by Western Geophysical.]

3:35–3:50 Break

3:50

4pUW5. Modeling cetacean ear filters by means of evolutionary computation. Dorian S. Houser, David A. Helweg, Patrick W. B. Moore (SPAWARSYSCEN-San Diego, Code D351, 53560 Hull St., San Diego, CA 92152-5435), and Kumar Chellapilla (Univ. of California at San Diego, La Jolla, CA 92093-4007)

Modeling cetacean hearing is an important step in advancing models of echolocation and in predicting the potential impact of oceanic noise upon cetaceans. Modeling the response of cetacean ears to acoustic input can be a difficult task and the performance of such models is often suboptimal. Evolutionary programming was employed as a systematic method of optimizing the response of a bank of pseudo-Gaussian filter shapes to known audiometric functions in order to model the resonant systems of the basilar membrane. The response of the filter banks was tested against normalized audiograms of the bottlenose dolphin (*Tursiops truncatus*) and the predicted frequency dependent sensitivity of the Humpback whale (*Megaptera novaeangliae*). Filter banks were created that demonstrated comparable frequency responses to tonal stimuli across the (predicted) range of hearing in both species. The potential for incorporating these ear models into models of echolocation and their use in predicting the sensitivity of baleen whales to manmade noise will be discussed.

4:05

4pUW6. An inexpensive, portable, and rugged system for recording the low-frequency sounds of cetaceans. Catherine L. Berchok (Grad. Prog. in Acoust., The Penn State Univ., P.O. Box 30, State College, PA 16804, berchok@sabine.acs.psu.edu), Thomas B. Gabrielson, and David L. Bradley (The Penn State Univ., State College, PA 16804)

Many studies have been done on the low-frequency sounds of cetaceans. However, few of these studies have involved coincident photoidentification, biopsy, and behavioral observation. In order to make acoustical recordings in conjunction with the regular visual fieldwork done from inflatable boats on the baleen whales of the St. Lawrence, a recording system was designed to meet several requirements. It had to be very inexpensive; light enough to be carried onto the boat each day and deployed by hand; and rugged enough to survive being pounded and soaked when stored on the open boat. The resulting system consisted of two main parts: a recording system made up of DAT recorder, amplifiers, filters, batteries and voltage regulators; and a surface-motion isolation system made up of a spar buoy, omnidirectional hydrophone, cable and damping plate. A mixing circuit allowed for real-time detection of infrasonic signals through a pair of headphones. In addition to the description of the system and its calibration, samples of blue, fin, minke and humpback whale recordings will be presented, and a brief comparison of the system with and without the spar buoy will be shown. [Work supported by a graduate fellowship from NSF.]

4:20

4pUW7. Displacement of migrating bowhead whales by sounds from seismic surveys in shallow waters of the Beaufort Sea. W. John Richardson, Gary W. Miller (LGL Ltd., Environ. Res. Assoc., P.O. Box 280, King City, ON L7B 1A6, Canada), and Charles R. Greene, Jr. (Greeneridge Sci., Inc., Santa Barbara, CA 93110)

Seismic surveys for subsea oil deposits were conducted each summer, 1996–98, mainly in water <20 m deep. Airgun arrays were used, with 6–16 airguns and total volumes 560–1500 cu.in. Low-frequency sound pulses were created at intervals of 8–20 s. Effective source levels for horizontal propagation were lower than nominal source levels, but pulses were often detectable to 50+ km offshore. Westward autumn migration of bowhead whales near and offshore of the exploration area was monitored by aerial surveys flown daily, weather permitting, during the three seasons. Aerial survey data from days with and without airgun operations were compared. Most bowheads avoided the area within 20 km of the operating airguns; bowheads were common there on days without airgun operations. In 1998, numbers sighted 20–30 km away were also significantly reduced during airgun operations. Conversely, sighting rates just beyond the avoidance zone were higher on days with airgun operations. Broadband received levels of airgun pulses at 20 km were typically 120–130 dB *re*: 1 μPa (rms over pulse duration), lower than those previously demonstrated to cause avoidance by bowheads. Many migrating bowheads 20 to 50+ km offshore were exposed to weaker but presumably detectable pulses. [Work supported by Western Geophysical and BP Exploration (Alaska), Inc.]

4:35

4pUW8. Predicting the effects of sound exposure on fish, marine mammals, and human beings. Antoine David, Peter Ward, Tony Heathershaw (DERA, Southampton Oceanogr. Ctr., Southampton, UK), and Peter Varley (Element Ltd., Fareham, UK, elements@freenet.uk.com)

Fish and marine mammals are sensitive to sound and may be affected by sound energy from a wide range of sources. Current concern is focused on anthropogenic sound and in particular sound which emanates from active sonar devices. This may range from broadband short-duration events to narrow-band signals with long-time exposure. Frequencies may extend from a few hundred Hz to a few hundred kHz. In order to be able to predict the effect of sound on fish, marine mammals, and human beings, it is first of all necessary to predict the propagation of sound through the ocean environment and then to estimate the effect of a received sound pressure level on an environmental receptor. To achieve this latter objective, a generic threshold of hearing model has been developed which is suitable for use with sonars of arbitary frequency and duty cycle. The model, which is described in this paper, relates the effects of sound in both the frequency and time domains. This enables the onset of TTS, PTS, and an equivalent damage risk criteria to be predicted as a function of the distance between an environmental receptor and a sound source.

4:50–5:15

Panel Discussion

THURSDAY AFTERNOON, 4 NOVEMBER 1999 UNION B ROOM, 2:00 TO 3:30 P.M.

Meeting of Accredited Standards Committee (ASC) S1 on Acoustics

G. S. K. Wong, Chair S1
Institute for National Measurement Standards (INMS), National Research Council, Ottawa, Ontario K1A 0R6, Canada

P. D. Schomer, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics
U.S. CERL, P.O. Box 9005, Champaign, Illinois 61826-9005

T. J. Kuemmel, Vice Chair S1
Quest Electronics, 510 South Worthington, Oconomowoc, Wisconsin 53066

H. E. von Gierke, Vice Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics
1325 Meadow Lane, Yellow Springs, Ohio 45387

V. Nedzelnitsky, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
National Institute of Standards and Technology (NIST), Sound Building, Room A147, 100 Bureau Drive, Stop 8221, Gaithersburg, Maryland 20899-8221

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on their preparation of standards on methods of measurement and testing, and terminology, in physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Work in progress includes measurement of noise sources, noise dosimeters, integrating sound-level meters, and revision and extension of sound level meter specifications. Open discussion of Committee reports is encouraged.

Scope of S1: Standards, specifications, methods of measurement and test and terminology in the field of physical acoustics including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance and comfort.

THURSDAY AFTERNOON, 4 NOVEMBER 1999 UNION B ROOM, 3:45 TO 4:45 P.M.

Meeting of Accredited Standards Committee (ASC) S3 on Bioacoustics

R. F. Burkard, Chair S3
Hearing Research Laboratory, State University of New York at Buffalo, 215 Parker Hall, Buffalo, New York 14214

J. Franks, Vice Chair S3
Robert A. Taft Laboratories, 4676 Columbia Parkway, Mail Stop C27, Cincinnati, Ohio 45226

P. D. Schomer, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics
U.S. CERL, P.O. Box 9005, Champaign, Illinois 61826-9005

D. D. Reynolds, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC4, Human Exposure to Mechanical Vibration and Shock
3939 Briar Crest Court, Las Vegas, Nevada 89120

H. E. von Gierke, Vice Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics and ISO/TC 108/SC4, Human Exposure to Mechanical Vibration and Shock
1325 Meadow Lane, Yellow Springs, Ohio 45387

V. Nedzelnitsky, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
National Institute of Standards and Technology (NIST), Sound Building, Room A147, 100 Bureau Drive, Stop 8221, Gaithersburg, Maryland 20899-8221

Accredited Standards Committee S3 on Bioacoustics. The current status of standards under preparation will be discussed. In addition to those topics of interest, including hearing conservation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years. Open discussion of Committee reports is encouraged.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of mechanical shock and physiological acoustics, including aspects of general acoustics, shock, and vibration which pertain to biological safety, tolerance, and comfort.