Session 2aAA

Architectural Acoustics: Small Rooms—Big Challenges

Alexander U. Case, Chair
Fermata Audio and Acoustics, P.O. Box 1161, Portsmouth, New Hampshire 03802

Invited Papers

8:00

2aAA1. Maximum performance synergy: A new approach to recording studio control room design. Jeff D. Szymanski (Auralex Acoust., Inc., 8851 N. Hague Rd., Indianapolis, IN 46256, consulting@auralex.com)

Popular recording studio control room designs include LEDE(tm), RFZ(tm), and nonenvironment rooms. The common goal of all of these is to create an accurate acoustical environment that does not distort or otherwise color audio reproduction. Also common to these designs is the frequent need to have multiple ancillary recording rooms, often adjacent to the main control room, where group members perform. This approach, where group members are physically separated from one another, can lead to lack of ensemble in the finished recordings. New twists on old acoustical treatment techniques have been implemented at a studio in Nashville, Tennessee, which minimize the need for multiple ancillary recording rooms, thus creating an environment where talent, producer and recording professionals can all occupy the same space for maximum performance synergy. Semi-separated performance areas are designed around a central, critical listening area. The techniques and equipment required to achieve this separation are reviewed, as are advantages and disadvantages to this new control room design approach.

8:25

2aAA2. Control room design: An analytic perspective. Douglas Jones (Columbia College Chicago, 600 S. Michigan, Chicago, IL 60605)

The recording studio control room is a special case of small room. In this paper the subjective attributes of a control room and their objective counterparts will be explored, discussing the analytic tools that might be used to evaluate and predict the subjective performance. A number of areas where more research is needed will also be identified.

8:50

2aAA3. Modeling sound fields in small rooms. Sung Yoon and Rendell R. Torres (Prog. in Arch. Acoust., Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180-3590)

In computationally modeling and auralizing acoustics of small rooms, geometrical-acoustics methods do not apply at lower frequencies, and models must account for the wave-nature of sound fields, especially modal effects and scattering in the early impulse response. Common approaches include finite-difference methods, finite-element methods, boundary-element methods, edge-diffraction models, boss models, among others. This investigation describes different time- frequency-domain techniques for computing the sound field in small rooms and introduces new methods as well, including fast algorithms based on cellular automata for calculating scattering from edge discontinuities and other boundary conditions. The latter approach will be described in relation to the pioneering work on the lattice gas time domain (LGTD) approach by Sudo and Sparrow [AIAA J. **33** (1995)] and how it can be applied to modeling the typical timbral and temporal (reverberant) characteristics of small room acoustics. [Work supported by Rensselaer Polytechnic Institute.]

9:15

2aAA4. Controlling loudspeakers in rooms. Jan Abildgaard Pedersen and Poul Praestgaard (Acoust. Res., Bang & Olufsen, Denmark)

How to control the perceived timbre of a loudspeaker in different listening positions, using different loudspeaker positions in different listening rooms is a central problem, when designing and using loudspeakers is discussed. Loudspeaker directivity has been found to be one of the critical parameters in the solution to this problem. This paper presents three hypotheses for the optimal loudspeaker directivity and a novel realization of that directivity, i.e., the Acoustic Lense (ALT). The dominant problem at low frequencies has been found to relate to the fact, that the acoustic power output of loudspeakers is highly dependent on the loudspeaker position and on the acoustic properties of the listening room. This paper also presents a novel system for adapting a loudspeaker to its position and to the acoustic properties of the listening room, i.e., Adaptive Bass Control (ABC).

9:40

2aAA5. Variable acoustics in small rooms for music. Yasushi Shimizu (Adv. System Development Ctr., Yamaha, Japan, Hamamatsu 430-8650, Japan)

Small rooms for music are frequently discussed to consider variable acoustics with different program in practice room, rehearsal room as well as audio listening room. In practice room major criteria in acoustics is usually discussed how to determine reverberation with less than 1.0 s. RT for different use such as practice for different musical instrument. And on audio listening room, acoustical effect on sound reproduced by speakers is important especially for reproducibility of 3D audio representation and usually surface material behind stereophonic speakers gave significant effect on the accuracy of sound localization with the sound played by speakers. Rehearsal room acoustics is always discussed compared to the stage acoustics in primary concert hall and some electro-acoustic technique is discussed how to simulate stage acoustics. This paper presents examples of acoustical design for different practice rooms, rehearsal room and audio listening room. A different RT setting for each room with different program is discussed as well as different configuration of surface material especially in audio listening room. Finally rehearsal room with electro-acoustics system to reproduce stage acoustics of specific concert hall is discussed and measurement result is presented.

10:05–10:20 Break

10:20

2aAA6. Low frequency evaluation and treatment of small rooms. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774)

At low frequencies, the acoustical coupling of the listener and loudspeakers with reflections from the room's boundary surfaces and its modal pressure distribution can cause significant acoustical distortion. For rectangular room's, software programs exist to predict the magnitude of these effects. However, there is no substitute for experimental measurements. When evaluating small rooms, it is often desirable to isolate the modal effects from the speaker-boundary effects, so that appropriate corrective measures can be applied. A MLS measurement procedure will be presented. After the room dimensions and listener/loudspeaker positions are optimized, one can apply dedicated low frequency absorbers to further control low frequency problems. The characterization of low frequency absorbers will be described, using a 7-ton, 22 long impedance tube.

Contributed Papers

10:45

2aAA7. Improvement of low frequency sound variability in a small enclosure by use of a source with frequency-independent radiated power. Alexandra Loubeau and Jiri Tichy (Grad. Prog. in Acoust., The Pennsylvania State Univ., P.O. Box 30, State College, PA 16804, aloubeau@psu.edu)

The irregularity in the low frequency response encountered in small rooms adversely affects the transmission of sound from a source to a receiver. Variations in the frequency response, which are dependent on source position, receiver position, and room absorption, result in perceived inconsistencies in sound quality throughout the room. This computational study examines the effect of constant sound power radiation on the sound field variability with frequency in a small enclosure. Effects of the parameters listed above are investigated. Frequencies of interest are 20–80 Hz. Source positions include the room corner, typical subwoofer locations, and typical full-range loudspeaker locations. Grids of receiver positions are selected around the center of the room, forward of center, back of center, and to the side of center. Room absorption values explored correspond to reverberation times of 0.5 s, 1 s, and 2 s. The benefit of constant sound power radiation is shown using the metric of standard deviation. Depending on the location of the source and receiver, the standard deviation of the frequency response decreases by approximately 2–3 dB with constant sound power radiation. [Research supported by Penn State ARL.]

11:00

2aAA8. Real-time virtual room acoustic simulation. James P. Carneal, Jan Johnson, Troge Johnson, and Marty Johnson (Vib. and Acoust. Labs., Dept. of Mech. Eng., VPI&SU, Blacksburg, VA 24061, jcarneal@vt.edu)

A realistic virtual room acoustic simulation has been implemented on a PC-based computer in near real-time. Room acoustics are calculated by the image source method using realistic absorption coefficients for a variety of realistic surfaces and programmed in MATLAB. The resulting impulse response filters are then applied in near real-time using fast convolution DSP techniques using data being read from a CD-ROM. The system was implemented in a virtual acoustic room facility. Optimizations have been performed to retain the realistic virtual room effect while minimizing computations through limited psycho-acoustic testing. In general, realistic anechoic to reverberant virtual rooms have been re-created with six 8192 coefficient filters. To provide realistic simulations, special care must be taken to accurately reproduce the low frequency acoustics. Since the virtual room acoustic facility was not totally anechoic (as are most anechoic chambers), inverse filters were applied to compensate for over-amplified acoustics at frequencies below 350 Hz.

11:15

2aAA9. Acoustical evaluation of preschool classrooms. Wonyoung Yang and Murray Hodgson (School of Occupational & Environ. Hygiene, Univ. of British Columbia, 2206 East Mall, 3rd Fl., Vancouver, BC V6T 1Z3, Canada)

An investigation was made of the acoustical environments in the Berwick Preschool, Vancouver, in response to complaints by the teachers. Reverberation times (RT), background noise levels (BNL), and in-class sound levels (Leq) were measured for acoustical evaluation in the classrooms. With respect to the measured RT and BNL, none of the classrooms in the preschool were acceptable according to the criteria relevant to this study. A questionnaire was administered to the teachers to assess their subjective responses to the acoustical and nonacoustical environments of the classrooms. Teachers agreed that the nonacoustical environments in the classrooms were fair, but that the acoustical environments had problems. Eight different classroom configurations were simulated to improve the acoustical environments, using the CATT room acoustical simulation program. When the surface absorption was increased, both the RT and speech levels decreased. RASTI was dependent on the volumes of the classrooms when the background noise levels were high; however, it depended on the total absorption of the classrooms when the background noise levels were low. Ceiling heights are critical as well. It is recommended that decreasing the volume of the classrooms is effective. Sound absorptive materials should be added to the walls or ceiling.

11:30

2aAA10. Estimating reverberation time in classrooms by the power method. Richard D. Godfrey (Owens Corning Sci. & Technol., 2790 Columbus Rd., Granville, OH 43023)

The Sabin equation relates reverberation time to room volume and absorption. If an acoustic source of known power is placed in a room, the absorption is also related to the power per bandwidth, volume, and pressure. Combining the Sabin equation with the power equation and solving for reverberation time yields a relationship for reverberation time as a function of volume, pressure, and the power of the source. This approach has been reported to deviate from the decay method at low frequency. In a laboratory reverberation room, it yielded reasonable results in the frequency range of interest in classrooms. Reasonable results were also achieved in real classrooms in the 500- to 2000-Hz frequency range. Based on these findings, it may be possible to make acceptable reverberation time measurements in less time with less sophisticated instrumentation than the decay method.

11:45

2aAA11. A survey of the acoustical quality of seventeen libraries at Princeton University. Benjamin Markham (Acentech, Inc., 33 Moulton St., Cambridge, MA 02134, bmarkham@acentech.com)

The purpose of this study was to identify objective acoustic measures that correlate with the subjective responses of students and administrators to libraries at Princeton University. The motivation for this study was to determine what is necessary in order to provide a comfortable acoustic environment for users of a new science library to be built on campus. On 31 March 2003, Acentech, Incorporated evaluated 17 library spaces and interviewed a number of students and librarians at Princeton. Based on the results of the survey, the author proposes that a comfortable acoustic environment in a library is an environment that provides freedom from distraction; in other words, casual conversation and other noises in the library will not distract users reading or studying in the library. In order to provide such an environment, a library must have (1) appropriate levels of background sound, (2) a physical barrier between noise-producing and noise-sensitive sections, and (3) sufficient sound absorbing material in the space. Measured quantitative metrics support these conclusions.

TUESDAY MORNING, 11 NOVEMBER 2003 SABINE ROOM, 8:25 A.M. TO 12:00 NOON

Session 2aAO

Acoustical Oceanography and Underwater Acoustics: Aubrey L. Anderson Memorial Session on Acoustics of Gas-Bearing Sediments

Anthony P. Lyons, Cochair

Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804-0030

Michael D. Richardson, Cochair

Naval Research Laboratory, Code 7430, Stennis Space Center, Mississippi 39529

Chair's Introduction—8:25

Invited Papers

8:30

2aAO1. The nature and location of gassy sediment sections in the continental shelf and slope in the northwestern Gulf of Mexico. William Bryant (Dept. of Oceanogr., Texas A&M Univ., College Station, TX 77843-3146)

In the northwestern Gulf of Mexico continental shelf and upper slope gassy sediments are a pervasive phenomena and an important consideration relative to engineering and acoustic activities on the sea floor. An examination of seismic data from over a thousand M.M.S. geohazard reports and core logs of 1,670 foundation boreholes drilled to an average subbottom depth of 125 m on the continental shelf and upper slope in the northwestern Gulf, has revealed that gassy sediment sections are most abundant near the Mississippi River Delta, in the sediment fill of buried stream channels that were eroded during the early and late Wisconsinan, and in Miocene and Plio–Pleistocene depocenters on the continental shelf and upper slope. Out of the 1,670 bore holes examined 1,158 (68%) contained indications of gassy sediments most of which is of biogenic origin. Large patches of gassy sediments exist, some exceeding 10 km in size but most are less than 500 m. The examination of 500 piston cores, up to 40 meters in length, taken on the mid and lower continental slope areas were almost void of gassy sediments as the result of the halokeiesis of allocthonous salt.

8:50

2aAO2. A global survey of the distribution of free gas in marine sediments. Peter Fleischer (Naval Oceanogr. Office, Stennis Space Center, MS 39522-5001), Tim Orsi (Planning Systems, Inc., Slidell, LA 70458), and Michael Richardson (Naval Res. Lab., Stennis Space Center, MS 39529-5004)

Following the work of Aubrey Anderson in the Gulf of Mexico, we have attempted to quantify the global distribution of free gas in shallow marine sediments, and have identified and indexed over one hundred documented cases in the scientific and engineering literature. Our survey confirms previous assumptions, primarily that gas bubbles are ubiquitous in the organic-rich muds of coastal waters and shallow adjacent seas. Acoustic turbidity as recorded during seismo-acoustic surveys is the most frequently cited evidence used to infer the presence of seafloor gas. Biogenic methane predominates within these shallow subbottom deposits. The survey also reveals significant imbalances in the geographic distribution of studies, which might be addressed in the future by accessing proprietary data or local studies with limited distribution. Because of their global prevalence, growing interest in gassy marine sediments is understandable as their presence has profound scientific, engineering and environmental implications.

9:10

2aAO3. Marine gas hydrate: Fabric, quantification and free gas content. Examples from Hydrate Ridge–Cascadia margin.
Friedrich Abegg (GEOMAR Res. Ctr. for Marine Geosciences, Wischhofstr. 1-3, 24148 Kiel, Germany, fabegg@geomar.de), Johannes Freitag (Alfred Wegener Inst. for Polar and Marine Res., Columbusstr., 27568 Bremerhaven, Germany), and Gerhard Bohrmann (Univ. Bremen, Am Fallturm 1, 28334 Bremen, Germany)

A recent advance in the investigation of internal gas hydrate structure and in small scale quantification is based on a combination of a new coring device and computerized x-ray tomography (CT). This approach has been chosen because gas hydrate is only stable within a special (low) temperature and (high) pressure field. A MultiAutoclaveCorer, developed by the Technical University of Berlin, is in principle similar to a piston corer and has the size of a multiple corer. During the deployments on Hydrate Ridge, two pressure vessels filled with seafloor samples could be recovered. They were CT-investigated after the cruise in a medical clinic close to San Francisco. As a result, a 3-D-dataset of the cores is available which allows to quantify the components' gas hydrate, sediment and free gas and also shows the distribution and orientation of the gas bubbles. One of the pressure vessels showed a distinct gas hydrate horizon where the gas hydrate content reached close to 50 vol%. Within this horizon there was a free gas content of 2.4 vol%. The preferential bubble orientation, compared to free gas in soft marine sediments, is horizontal, not vertical, which is an indicator for the mechanism of gas hydrate formation.

9:30

2aAO4. Data-model comparisons for scattering by a bubbly sediment. Darrell R. Jackson and Dajun Tang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

As part of the Coastal Benthic Boundary Layer Program, seafloor backscattering measurements were made at 40 kHz in Eckernfoerde Bay, Germany, in areas of high concentration of gaseous methane. Although it had long been known that methane bubbles have a profound influence on acoustic scattering, this was the first time that contemporaneous measurements were made of acoustic scattering and bubble size distributions. The latter measurements were part of a collaboration between Aubrey Anderson's group and NRL-SSC and made detailed model-data comparisons possible. Acoustic interferometry was used to conclude that the dominant scattering layer was about 1 m below the sediment–water interface. A single scattering model accounted for the angular dependence and level of scattering, however, the model gave unphysical results at very small grazing angles. A multiple scattering model has been developed to remedy this defect. [Work sponsored by ONR.]

9:50

2aAO5. Ups and downs of a gas horizon under environmental control. Thomas F. Wever (Forschungsanstalt der Bundeswehr für Wasserschall und Geophysik, Klausdorfer Weg 2-24, 24148 Kiel, Germany, Wever@fwg-kiel.de)

Repeated measurements with a sub-bottom profiler along reference tracks in Eckernförde Bay (Western Baltic) led to the recognition that acoustic turbidity (the "gas horizon") within the muddy sediments changed its depth. Soon a dependence on the annual temperature cycle with a 3–4 month delay became obvious. In addition, short-term variations correlated with atmospheric pressure changes. For a further investigation, a tower equipped with an echosounder, pressure, and temperature probes at different sediment depths was deployed for four years in the center of the bay. Data were recorded every hour. The data show the migration of the annual temperature peak and low through the sediment and its long-term influence on acoustic turbidity depth. Short-term ("high-frequency") variations can be correlated with changes in total effective pressure. Repeated investigations of the total methane content in sediment cores during different times of the year and the analysis of the cores with the use of computer tomography under the guidance of Aubrey L. Anderson proved a constant total methane content. This led to the explanation that the environmental factors pressure and temperature cause variations of gas solubility that move up and down and thus control the existence or absence of free gas bubbles within the sediment.

10:10–10:30 Break

10:30

2aAO6. A biological source of bubbles in sandy marine sediments. D. V. Holliday, Charles F. GreenlawIII (BAE Systems, 4669 Murphy Canyon Rd., San Diego, CA 92123, van.holliday@baesystems.com), David Thistle (Florida State Univ., Tallahassee, FL 32306), and Jan E. B. Rines (Univ. of Rhode Island, Narragansett, RI 02882)

Gas in sediments, even in small quantities, will modify the propagation and scattering of sound. Shallow water littoral environments are often sufficiently well lit by sunlight to support healthy populations of benthic and epibenthic marine microalgae. Photosynthesis in marine algae produces oxygen. Oxygen saturation levels as high as 600% have been measured in the pore water of a sandy sediment at 1 mm depth, decreasing to 100% saturation at ca. 4.5 mm. Bioturbation and physical processes routinely mix materials at the surface of the seabed, including algae, to depths of at least a few cm. Mixing times and depths vary with the sediment

type and the species and abundance of organisms present, but time scales of minutes to hours are common. While light is rapidly attenuated with depth in sand, measurements show that it can penetrate to depths of a few mm. Physical and biological mechanisms are suggested which could produce gas bubbles in oxygen saturated pore water. Laboratory measurements of sound scattering from marine algae on a sand surface suggest a possible method for *in situ* bubble detection in shallow marine environments. [Work supported by ONR.]

10:50

2aAO7. Frequency-dependent acoustic properties of gassy marine sediments. Angus I. Best (Challenger Div. for Seafloor Processes, Southampton Oceanogr. Ctr., European Way, Southampton SO14 3ZH, UK, aib@soc.soton.ac.uk), Michael D. J. Tuffin, Justin K. Dix, and Jonathan M. Bull (School of Ocean and Earth Sci., Southampton Oceanogr. Ctr., Southampton SO14 3ZH, UK)

Acoustic velocity and attenuation were measured during two *in-situ* experiments in gassy intertidal muds in Southampton Water, United Kingdom. The horizontal transmission results gave frequency-independent velocity (1431 m/s) and attenuation (4 dB/m) over the frequency range 600 to 3000 Hz, representative of the soft (non-gassy) muds shallower than about 1 m. The results from a vertical transmission experiment straddling the top of the gassy zone (about 1 m depth) showed strong frequency-dependent velocity and attenuation over 600 to 3000 Hz. They showed velocity and attenuation maxima predicted by the Anderson and Hampton model, associated with gas bubble resonance. Moreover, attenuation maxima shifted in frequency with water depth over a tidal cycle that was monitored, suggesting variations in gas bubble size with hydrostatic pressure. X-ray CT images on a sealed core from the site revealed vertically-aligned, centimeter-scale, gas-filled cracks in the muddy sediments. Ultrasonic (300 to 700 kHz) velocities and attenuations were higher in the gassy zone than in the nongassy parts of the core. Overall, the results give a fascinating insight into the acoustical behavior of gassy sediments that could be used to extract sediment physical properties information from seabed acoustic reflection data. [Work supported by NERC].

11:10

2aAO8. Determining gas bubble morphology and size distribution in mud using CT imagery. Kevin B. Briggs and Allen H. Reed (Seafloor Sci. Branch, Naval Res. Lab., Stennis Space Center, MS 39529-5004)

Sampling for the measurement and description of methane gas bubbles *in situ* was accomplished using the method of Abegg and Anderson [Mar. Geol. **137**, 137–147 (1997)]. Sediment cores from East Bay, off the mouth of the Mississippi River, were collected by divers and placed into aluminum pressure transfer chambers while on the seafloor. With the cores at seafloor pressure within the chambers, they were transferred to an x-ray computed tomography (CT) scanner where high-resolution images were made of the sediment within the cores. Data, in the format of series of cross-sectional images of x-ray attenuation reconstructed in 3-D, were evaluated in terms of spatial distribution, sizes, and shapes of bubbles. CT imagery was obtained from a GE LS medical CT scanner at a local hospital and the Naval Research Laboratory's new HD-500 industrial CT scanner specifically designed for core sample imaging. The medical scanner provided images of stationary cores at 625-μm intervals with a rotating x-ray source and was able to resolve bubbles down to 625 μm in diameter. The industrial scanner provided images of rotating cores at 25-μm intervals with a stationary source and was able to resolve bubbles down to 10 μm in diameter.

Contributed Papers

11:30

2aAO9. Gas bubbles in marine mud—How small are they? Allen H. Reed and Kevin B. Briggs (Seafloor Sci. Branch, Naval Res. Lab., Stennis Space Center, MS 39529)

Free gas in marine mud poses a challenging problem in the realm of ocean acoustics as it readily attenuates (i.e., scatters or absorbs) energy, such that objects lying below the gassy sediment are acoustically masked. Gas-laden sediments were located in 10- to 120-m water depth adjacent to the South Pass of the Mississippi River in East Bay using a 12-kHz transducer and the Acoustic Sediment Classification System. Several cores were collected in this region for physical property measurements. Some of the cores were x-rayed on medical and industrial computed tomography (CT) scanners. Volumetric CT images were used to locate gas bubbles, determine shapes and sizes to within the limits of the CT resolution. Free gas in the East Bay sediments was relegated to worm tubes as well as isolated pockets as was the case in Eckernförde Bay sediments [Abegg and Anderson, Mar. Geol. **137**, 137–147 (1997)]. The primary significance of the present work is that gas bubbles have been determined to exist in the tens of μm size range, which is significantly smaller than the smallest bubbles that were previously resolved with medical CT ($\sim$440 μm) with NRL's HD-500 micro-CT System. [Work supported by ONR and NRL.]

11:45

2aAO10. Sensitivity of bubble resonance to environmental inputs using the Anderson model. Warren Wood and Michael Richardson (Naval Res. Lab., Code 7430, Stennis Space Center, MS 39529)

The Anderson model relates the expected sound speed and attenuation (both a function of bubble resonance) through bubbly sediment to physical properties of the gas (including shape and temperature) as well as the anelastic properties of the surrounding sediment. This mathematical formulation allows relatively straightforward assessment of the sensitivity of the sound speed and attenuation to each of the approximately 20 inputs to the model. Our sensitivity analysis was performed at several locations in model space that correspond to realizations of frequently encountered, realistic, bubbly sediments. We first determine a baseline multi-frequency acoustic response from a starting sediment and gas type, then compare the baseline response to responses from a perturbed model. Coupled parameters are perturbed simultaneously (by differing percentages) such that at no time are we simulating the response of an unrealistic sediment. Results suggest a generally greater sensitivity to gas and pore fluid properties than to sediment grain, or frame properties.

Session 2aBB

Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Ultrasound Contrast Agents

Constantin-C. Coussios, Cochair
Department of Aerospace and Mechanical Engineering, Boston University, Boston, Massachusetts 02215

Preston S. Wilson, Cochair
Mechanical Engineering Department, University of Texas at Austin, Austin, Texas 78712-0292

Invited Papers

8:30

2aBB1. Methods for blood flow measurements using ultrasound contrast agents. J. Brian Fowlkes (Dept. of Radiol., Univ. of Michigan, Kresge III R3315, Ann Arbor, MI 48109-0553)

Blood flow measurements using ultrasound contrast agents are being investigated for myocardial perfusion and more recently in other organ systems. The methods are based largely on the relative increase in echogenicity due to the concentration of bubbles present in the ultrasound beam. In the simplest form, regional differences in blood volume can be inferred but the possibility exists to extract perfusion from the transit of contrast agent through tissue. Perfusion measurements rely on determining the flux of blood through a tissue volume and as such require knowledge of the fractional blood volume (FBV), i.e., ml blood/g tissue and the rate of exchange, commonly measured as the mean transit time (MTT). This presentation will discuss methods of determining each of these values and their combination to estimate tissue perfusion. Underlying principles of indicator-dilution theory will be provided in the context of ultrasound contrast agents. Current methods for determining MTT will include imaging of the intravenous bolus, in-plane contrast disruption with interval and real-time contrast recovery imaging, and control of contrast agent flow using arterial disruption (contrast interruption). The advantages and limitations of the methods will be examined along with current applications. [Work supported in part by NIH.]

8:55

2aBB2. A review of theoretical models for microbubble contrast agents. Charles C. Church and Xinmai Yang (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, University, MS 38677)

Theoretical investigation of the acoustic responses of albumin-encapsulated microspheres began nearly fifteen years ago when Albunex, the first agent approved for clinical use in the U.S., was still in development. Since that time, the number of potential ultrasound contrast agents has grown considerably. These agents utilize a variety of both shell materials (e.g., proteins, synthetic polymers, surfactants, lipids) and encapsulated gases (e.g., air, sulfur hexafluoride, perfluorocarbons). The shell may be very thick or vanishingly thin. A thorough understanding of the interaction between ultrasound pulses and contrast microbubbles is essential for the successful clinical application of a particular agent. In this talk, models developed by de Jong *et al.*, Church, Ye, Allen *et al.*, Khismatullin *et al.*, and others will be reviewed. The basis of these theories is a free bubble model supplemented by the effect of the encapsulating shell. The differences among these models lie primarily in their treatment of the encapsulating layer and, to some extent, the surrounding medium. The nature of various contrast agents will be discussed, and appropriate models for each will be described. Comparisons among models will include predictions of clinically significant acoustic responses (resonance frequency, scattering strength, nonlinearity, etc.).

9:20

2aBB3. Acoustical and optical monitoring of contrast microbubbles. Jingfeng Guan, Wen-shiang Chen, and Thomas Matula (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

We have undertaken a series of studies to better understand the response of ultrasound contrast microbubbles to pulsed ultrasound. Toward this end we have used acoustical and optical scattering techniques. The acoustic interrogation of shell-disrupted microbubbles immediately following HIFU was used to monitor microbubble (or fragmented daughter) dissolution. Results were compared to calculations, assuming a simple Gaussian distribution of fragmented microbubbles and the dissolution characteristics of a mixed-gas system (such as a perfluorocarbon gas bubble in air-saturated water). Good results were obtained by considering a multi-modal bubble size distribution. That is, the HIFU pulse created a distribution of smaller bubbles. In order to follow the instantaneous motion of a single microbubble during pulsed ultrasound, light scattering was employed. For this experiment, a diagnostic ultrasound system was used to force the microbubble into oscillation. Light scattered off the microbubble was collected by a photodetector. A simple bubble dynamics model was used to fit the data. At low pressures, consecutive data segments were averaged to increase the signal/noise. At higher pressures, the bubble (size or shell) properties required systematic adjustment in order to obtain a good fit, suggesting that the bubble evolved with each pulse.

9:45

2aBB4. Ultrasound contrast agents and their use in monitoring therapy. Katherine Ferrara, Paul Dayton, Michaelann Shortencarrier, and Dustin Kruse (Dept. of Biomed. Eng., Univ. of California–Davis, Davis, CA 95616)

The shell of ultrasound contrast agents can be modified to include a molecular targeting ligand, and the properties of the agent with and without molecular targeting can be used to monitor changes produced by a therapy. We have investigated the use of ligands targeted to an integrin expressed in cancer, whose expression correlates with tumor grade. Acoustic studies illustrate a 3- to 20-fold increase in echo amplitude from integrin-expressing cells exposed to the targeted contrast agent, as compared to controls, and depending on cell type, stimulation, and targeting ligand. Changes in integrin expression with therapy may be important in future studies. We have also developed a system to quantify small changes in vascular parameters due to effects of new anti-angiogenic drugs using the intrinsic properties of contrast agents. Regions containing intravascular contrast agents are identified using a strategy that combines subharmonic and phase inversion imaging. As predicted by a Rayleigh–Plesset analysis, this strategy can successfully detect flow over a range of transmission frequencies from 4–6 MHz. We demonstrate that regions of viable tumor as small as 1 mm, as verified by histology, can be detected and show similar morphology to images acquired with computed tomography (CT).

10:10–10:30 Break

Contributed Papers

10:30

2aBB5. Effect of contrast agent on the incidence of ultrasound-induced lung hemorrhage in rats. William D. O'Brien, Jr. (Dept. of Elec. and Computer Eng., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801), Douglas G. Simpson (Univ. of Illinois, Champaign, IL 61820), Leon A. Frizzell (Univ. of Illinois, Urbana, IL 61801), and James F. Zachary (Univ. of Illinois, Urbana, IL 61802)

The objective is to test further the hypothesis that if inertial cavitation in the vasculature of the lung is the physical mechanism responsible for ultrasound-induced lung hemorrhage, then the addition of cavitation nuclei to the blood will enhance the occurrence of lung hemorrhage. A factorial design was used to study the effects of two types of injected agents (IA; 0.25 mL per rat of saline or Optison given intravenously) and two levels of pulsed ultrasound exposure (UE; *in situ* peak rarefactional pressures of 2.74 and 5.86 MPa) on the incidence and size of lung lesions. Ten 10-to-11-week-old Sprague–Dawley rats were exposed at each of the four combinations of IA and UE at 3.1 MHz for 10 s (1-kHz PRF, 1.2-microns PD). Rats administered contrast agent prior to exposure did not have an increase in lesion occurrence or size compared to rats that received saline with no contrast agent. These results provide further evidence that the mechanism of lung hemorrhage is not inertial cavitation. These findings are consistent with another group's results from another species (mouse) showing no increase in the area of lung hemorrhage using a different contrast agent (Albunex) when exposed to pulsed ultrasound. [Work supported by NIH Grant No. R01EB02641.]

10:45

2aBB6. Role of ultrasound contrast agent models in applications and design. John S. Allen, Shin Yoshizawa, Yukio Kaneko, and Yoichiro Matsumoto (Dept. of Mech. Eng., Univ. of Tokyo, Hongo 731, Bunkyo-ku, Tokyo 113, Japan, allenj@fel.t.u-tokyo.ac.jp)

The earliest ultrasound contrast agents were constructed with an air core surrounded by a protein shell. However, contrast agents are currently being produced and developed with a variety of internal gases and shell materials. The agent's shell has been constructed with lipid and polymer materials. These developments have been driven more by empirical experimental fittings rather than a rigorous study of material properties and a construction process that involves both theory and experiment. Furthermore, for diagnostic applications the optimization of the resonance behavior as a function of shell material and thickness is desired. Also the role of the shell in destruction remains an outstanding issue. Recently developed contrast agent models based on hyperelastic materials are further developed to investigate both more extensive strain energy functions and different types of damping from the shell materials. The role of viscous and viscoelastic damping is highlighted. From these radial equations, corresponding versions can be derived in the membrane limit. These models may be used with previously developed bubble dynamics codes, which have a comprehensive formulation for the internal phenomena, and for the analysis of recent experimental results involving therapeutic applications.

11:00

2aBB7. Acoustical correction to the bubble dynamics equation for elastic media. Stanislav Y. Emelianov (Dept. of Biomed. Eng., Univ. of Texas, Austin, TX 78712-1084), Mark F. Hamilton, Yurii A. Ilinskii, G. Douglas Meegan, and Evgenia A. Zabolotskaya (Univ. of Texas, Austin, TX 78713-8029)

Nonlinear oscillations of microbubbles are being investigated extensively for innovative medical and industrial applications. The Rayleigh–Plesset equation successfully describes the dynamics of gas bubbles in liquids and viscoelastic media. To model bubble dynamics in weakly compressible elastic media such as soft tissue, effects of shear stress should be taken into account. These effects were included recently by the authors in a model for bubble oscillations that differs from the Rayleigh–Plesset equation by a single term associated with nonlinear elasticity of the medium surrounding the bubble. Here, we augment this dynamical equation to account for acoustic radiation and thus describe nonlinear bubble oscillations in slightly compressible elastic media. The bubble is assumed to be spherical and the surrounding elastic medium to be isotropic. The analysis is performed in Eulerian coordinates, and attention is devoted to terms associated with radiation loss. Our analysis builds on results obtained by Prosperetti and Lezzi [J. Fluid Mech. **168**, 457 (1986)] for a bubble in liquid. The stress tensor for the elastic medium is expressed through invariants of the Green deformation tensor in spherical coordinates. Explicit expressions for components of the stress tensor are obtained using Mooney's potential function. [Work supported by IR&D at ARL:UT.]

11:15

2aBB8. Forced linear oscillations of microbubbles in blood capillaries. Elisabetta Sassaroli and Kullervo Hynynen (Dept. of Radiol., BWH, Harvard Med. School, Boston, MA)

A numerical investigation of the forced oscillations of ultrasound contrast agents in blood capillaries are investigated by means of a simplified model based on the assumption of small amplitude oscillations of the microbubbles acoustically driven. The natural frequencies of oscillations, the thermal and viscous damping coefficients, the amplitude resonances, as well as the average power absorbed by the system, bubble plus vessel, are computed for different kinds of contrast agent bubbles, containing air, octafluropropane, and perflurobutane. It is found that viscous damping is much larger than thermal damping and that the natural frequencies depend much more on the system parameters than the nature of the gas contained in the bubble. Our motivation for this study lies in the possibility of using bubbles as vectors for directed drug delivery and gene transfection into living cells.

11:30

2aBB9. An interfacial rheological model of ultrasound contrast agents: Theory and experiments. Kausik Sarkar, Dhiman Chatterjee (Mech. Eng., Univ. of Delaware, Newark, DE 19716, sarkar@me.udel.edu), William T. Shi, and Flemming Forsberg (Thomas Jefferson Univ., Philadelphia, PA 19107)

A new interface model with zero thickness and intrinsic surface rheological parameters has been developed for the encapsulation of a microbubble contrast agent. A modified Rayleigh–Plesset equation with these parameters (surface tension γ and surface dilatational viscosity κ^s) was obtained. The rationale for the zero-thickness interface lies in the anisotropy and the inhomogeneity in the structure of the encapsulation. The parameters of the model were determined using in vitro attenuation data for various contrast agents (Albunex, Optison, Sonazoid, and Quantison). The linearized dynamics was used to match data from a controlled set of experiments performed at smaller pressure amplitudes (<0.1 MPa). The model was investigated for its ability to predict the nonlinear response at higher amplitudes (>1.0 MPa) for two test cases: Optison (Mallinckrodt, St. Louis, MO) with $\gamma=0.9$ N/m, $\kappa^s=0.08$ msP and Sonazoid (Amersham Health, Oslo, Norway) with $\gamma=0.6$ N/m, $\kappa^s=0.01$ msP. In contrast to existing models, the new model captured the distinct characteristics of subharmonic emissions—initiation and rapid growth—for these agents. The present model seems to be a reliable descriptor of contrast agent behaviors, and therefore, may be used to characterize different agents and to design next generation agents by tuning the model parameters. [Work supported by DOD.]

11:45

2aBB10. Parameters affecting the detection of microbubbles in blood. Constantin-C. Coussios, Paolo Zanetti, and Ronald A. Roy (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

The ability to detect small gas bubbles in blood depends on the relative magnitude of the acoustic power backscattered from the microbubbles (''signal'') to the power backscattered from the red blood cells (''noise''). Erythrocytes are weak, Rayleigh scatterers, and therefore the backscattering coefficient (BSC) of blood increases as the fourth power of frequency throughout the diagnostic frequency range. Microbubbles, on the other hand, are either resonant or super-resonant in the range 10–30 MHz. Above resonance, their total scattering cross-section remains constant with increasing frequency and the directivity pattern of the scattered wave changes significantly. Therefore, increasing the detection frequency may lead to a reduction in signal-to-noise ratio. An active cavitation detector (ACD) was utilized to observe the gradual obscuring of a steel target in blood with increasing frequency, and to measure the BSC of suspensions of Optison® microspheres in blood, as a function of microsphere concentration, hematocrit and frequency in the range 10–30 MHz. The experimental results were compared with theoretical predictions of the BSC of Optison® and blood, in order to determine whether the presence of tightly packed red blood cells affects the acoustic response of the microbubbles. [Work supported by the ASA, the US Army, and the NSF.]

TUESDAY MORNING, 11 NOVEMBER 2003 PECOS ROOM, 8:00 A.M. TO 12:00 NOON

Session 2aEA

Engineering Acoustics: New Developments

Kim C. Benjamin, Chair
NAVSEA Division Newport, 1176 Howell Street, Newport, Rhode Island 02841

Contributed Papers

8:00

2aEA1. Broadband sonar considerations for small underwater vehicle applications. Kim C. Benjamin (NAVSEA Div. Newport, 1176 Howell St., Newport, RI 02841)

As underwater vehicles become more prevalent, so too does the design and integration of compact broadband sonar arrays. Rather than rely on conventional tonpilz technology, where the bandwidth is governed by the width of the transducer's mechanical resonance, designers must consider other transducer technologies that are better suited to small vehicle packaging constraints. These constraints include operational ruggedness, light weight, conformability, and low cost. This talk advocates the use of 1–3 piezocomposite and discusses the rationale behind such a selection. In going to a wideband material such as a 1–3 piezocomposite, with typical mechanical quality factors (Qm) around 2, the selection of where to place the resonance frequency differs from that of its tonpilz counterpart. For the tonpilz array, the sonar operational bandwidth is totally governed by the resonance response of the array element and is typically limited to approximately its 3-dB or half power points. Relaxor-based ferroelectric materials such as single crystal PMN-PT, which exhibit extremely large electromechanical coupling coefficients, cannot attain 3-dB bandwidths of a decade or more when configured in a tonpilz design. This presentation will discuss a 1–3 piezocomposite-based approach that places mechanical resonance near the upper band edge of an operational bandwidth of 1 decade (10–100 kHz).

8:15

2aEA2. The comb waveform as an efficient method for wideband measurements. Walter H. Boober, Gorham Lau, Kim C. Benjamin (NAVSEA Div. Newport, 1176 Howell St., Newport, RI 02841), and Kenneth M. Walsh (K&M Eng. Ltd., Middletown, RI 02843)

An efficient acoustic calibration technique based on a uniformly-weighted comb waveform is presented. The method takes advantage of the linear, time invariant nature of the measurement configuration and the comb's wide bandwidth to capture all spectral components of interest for a device under test in a single ping. Results comparing single ping comb measurements with conventionally obtained CW measurements are presented. The examples given illustrate the accuracy and utility of this technique for the calibration of broadband systems.

8:30

2aEA3. *In situ* calibration of sonar arrays. L. D. Luker and S. E. Forsythe (Naval Undersea Warfare Ctr. Div. Newport, USRD Code 216, Bldg. 1171B, Rm. 205, 1176 Howell St., Newport, RI 02841, lukerld@npt.nuwc.navy.mil)

The transmitting and receiving properties of the channels of sonar arrays can change with time resulting in a degradation of the array's performance. Fortunately, the degradation in performance can be minimized, perhaps even eliminated, if the changes in a channel's transmitting or receiving properties are compensated for in the array's beamformer elec-

tronics. However, this requires up-to-date knowledge of the acoustic performance of each of the array's channels. This paper describes a procedure for the *in situ* calibration of sonar arrays when the vessel they are installed on is in open water. It can be used to determine changes in the electroacoustic performance of the projecting and receiving channels of the array. The method used is based on a procedure for *in situ* comparison calibration of transducers [A. L. Van Buren, "Procedure for the *in situ* calibration of sonar transducers," J. Acoust. Soc. Am. **90**, 48–52 (1991)] that uses sound-propagation factors measured when the vessel is first deployed to account for the influence of the vessel's structure. Results are presented that show comparisons of the measured degradation of numerous channels in a planar array using an independent acoustic measurement and the *in situ* method. [Work supported by ONR.]

8:45

2aEA4. Shear mode properties of single crystal ferroelectrics. E. A. McLaughlin and H. C. Robinson (NUWC, Code 2132, 1176 Howell St., Newport, RI 02841, mclaughlinea@npt.nuwc.navy.mil)

Single crystal ferroelectrics or piezocrystals were recently introduced into the electroactive materials community. The 33-mode electromechanical coupling factor of piezocrystals is typically greater than 0.90, which is significantly larger than typical values for piezoelectric ceramics (0.62–0.74). For sonar projector applications this large k33 has been responsible for more than doubling the bandwidth of active sonar arrays over what is currently achievable with ceramics. Last year a crystal grower produced a cut of lead magnesium niobate-lead titanate (PMN-PT) single crystal with piezoelectric shear coefficient values of 7000 pm/V and shear coupling factors of 0.97. (For PZT5H, d15 is 730 pm/V.) This piezocrystal d15 coefficient implies significantly improved sensitivity and signal-to-noise ratio for accelerometers and hydrophones, while the high coupling promises bandwidth increases greater than those realized in 33-mode projectors using piezocrystals. This research studies the shear-mode behavior of PMN-PT piezocrystals for use in sensors and projectors. By measuring the response of the materials to high and low level electrical bias and excitation fields, frequency, and temperature, the materials' effective material properties as a function of these operational variables were determined. [Work sponsored by ONR and NUWC ILIR.]

9:00

2aEA5. Frequency shift of a rotating-imbalance vibration source caused by radiative damping in the surrounding medium. Stephen R. Novascone (Idaho Natl. Eng. Lab., 225 N. Fremont, Idaho Falls, ID 83415-2110), Michael J. Anderson (Univ. of Idaho, Moscow, ID 83844-0902), David M. Weinberg (Idaho Natl. Eng. Lab., Idaho Falls, ID 83415-2110), and Jack H. Cole (Univ. of Arkansas)

The motion of vibrating bodies in a surrounding fluid is often used to infer the transport properties of the fluid. A new sensor configuration is presented that consists of a rotating imbalance source radiating into an unbounded fluid medium. Under these circumstances, the reaction of the fluid medium onto the vibration source includes a steady state torque that opposes the applied torque required to sustain the rotating imbalance. This reaction torque causes a shift in frequency of the vibration source. The frequency shift is related to the density of the surrounding fluid medium and vibration source characteristics. A description of measurements taken with a rotating-imbalance source located in unbounded water and air is provided. The total mass, eccentricity, and length of the source were 4.1 kg, $3.28(10^{-4})$ kg m, and 0.432 m, respectively. Motive torque to drive the imbalance was provided by a permanent-magnet dc motor. For an applied dc voltage that caused the source to operate at a nominal frequency near 150 Hz, a frequency shift of approximately 11 Hz was observed when the source was moved from air to water. Experimentally measured frequency shifts compared favorably with predictions provided by a nonlinear steady state model of the source and surrounding medium.

9:15

2aEA6. Amplitude and frequency experimental field measurements of a rotating-imbalance seismic source associated with changes in lithology surrounding a borehole. Stephen R. Novascone (Idaho Natl. Eng. Lab., 2525 N. Fremont, Idaho Falls, ID 83415-2110), Michael J. Anderson (Univ. of Idaho, Moscow, ID 83844-0902), David M. Weinberg (Idaho Natl. Eng. Lab., Idaho Falls, ID 83415), and Jack H. Cole (Univ. of Arkansas)

Field measurements of the vibration amplitude of a rotating-imbalance seismic source in a liquid-filled borehole are described. The borehole was a cased oil well that had been characterized by gamma-ray cement bond and compensated neutron litho-density/gamma-ray logs. The well logs indicated an abrupt transition from shale to limestone at a depth of 2638 ft. The vibration amplitude and frequency of a rotating-imbalance seismic source was measured versus applied voltage as the source was raised from 2654 to 2618 ft through the shale–limestone transition. It was observed that the vibration amplitude changed by approximately 10% in magnitude and the frequency changed approximately 15% as the source passed the shale–limestone transition. The measurements were compared to predictions provided by a two-dimensional analytical model of a rotating-imbalance source located in a liquid-filled bore hole. It was observed that the sensitivity of the experimentally measured vibration amplitude of the seismic source to the properties of the surrounding geologic media was an order of magnitude greater than that predicted by the two-dimensional analytical model.

9:30

2aEA7. Development of a stress gradient enhanced piezoelectric actuator composite (GEPAC) with integrated ultrasonic NDE capablity. Antoine Bechet, Yves Berthelot, and Christopher Lynch (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

Stress gradient enhanced piezoelectric composites (GEPACs) are unimorph based curved actuators that can be primarily used as low-frequency actuators embedded within an aircraft skin. The actuator is made of a thin PZT ceramic bonded between two composite layers with different coefficients of thermal expansion. Actuators have been manufactured with segmented upper electrodes so that they can also be used at ultrasonic frequencies to monitor continuously the integrity of the actuator itself (cracks), or of the actuator/skin assembly (debonding). This study will focus on the detection of debonding at the interface between the piezoelectric layer and the outer layers. The input signal is a one cycle burst at 1 MHz. The time domain signals recorded experimentally are compared with predictions from a finite element model (ABAQUS explicit). It is shown that the presence of a debonding defect can convert a symmetric pulse into a primarily antisymmetric wave. A simple correlation technique is used to estimate the size of the debonding defect. The ultrasonic NDE operation superposed to the low frequency actuation is demonstrated. [Work supported by the AFOSR.]

9:45

2aEA8. Acoustics of automotive catalytic converter assemblies. Nolan S. Dickey, Ahmet Selamet (Dept. of Mech. Eng., Ohio State Univ., 930 Kinnear Rd., Columbus, OH 43212-1443, dickey.21@osu.edu), Steve J. Parks, Kevin V. Tallio, Keith D. Miazgowicz, and Paul M. Radavich (Ford Motor Co., Dearborn, MI 48121)

In an automotive exhaust system, the purpose of the catalytic converter is to reduce pollutant emissions. However, catalytic converters also affect the engine and exhaust system breathing characteristics; they increase backpressure, affect exhaust system acoustic characteristics, and contribute to exhaust manifold tuning. Thus, radiated sound models should include catalytic converters since they can affect both the source characteristics and the exhaust system acoustic behavior. A typical catalytic converter assembly employs a ceramic substrate to carry the catalytically active noble metals. The substrate has numerous parallel tubes and is mounted in a housing with swelling mat or wire mesh around its periphery. Seals at the ends of the substrate can be used to help force flow

through the substrate and/or protect the mat material. Typically, catalytic converter studies only consider sound propagation in the small capillary tubes of the substrate. Investigations of the acoustic characteristics of entire catalytic converter assemblies (housing, substrate, seals, and mat) do not appear to be available. This work experimentally investigates the acoustic behavior of catalytic converter assemblies and the contributions of the separate components to sound attenuation. Experimental findings are interpreted with respect to available techniques for modeling sound propagation in ceramic substrates.

10:00–10:15 Break

10:15

2aEA9. Whistle suppression at a duct–sidebranch interface by boundary layer perturbation. Andrew Madden, Ahmet Selamet (Ctr. for Automotive Res., The Ohio State Univ., 930 Kinnear Rd., Columbus, OH 43212, selamet.1@osu.edu), and Kevin V. Tallio (Ford Motor Co., Dearborn, MI 48124)

Whistles are generated when shear layer instabilities in a main duct couple with acoustic resonances in a sidebranch. Such flow-acoustic coupling phenomenon is suppressed by perturbing, thus thickening the boundary layer near the interface of the adjoining ducts in an experimental setup, consisting of a closed sidebranch connected to a main duct. The boundary layer is perturbed by inserting pins that protrude into the main duct immediately upstream of the sidebranch duct. The pin height, diameter, and spacing are varied. The sound pressure is measured at the inlet of the main duct using a sound level meter, followed by a spectral analysis to determine the whistle frequencies. Pressure is also measured at the end of the sidebranch by a piezoelectric transducer to quantify the resonant amplitudes. The large-magnitude whistles produced by flow-acoustic coupling are suppressed in nearly all cases by the use of pins which provide, at times, more than 10 dB reduction at peak amplitudes. Furthermore, the introduction of pins into the main flow does not increase the background noise in the duct.

10:30

2aEA10. Special short wave finite elements for flow acoustics. Pablo Gamallo and Jeremy Astley (ISVR, Univ. of Southampton, Highfield, Southampton SO17 1BJ, UK, rja@isvr.soton.ac.uk)

Short wave problems of practical interest in acoustics include the modeling of sound attenuation in the ducted regions of turbofan aircraft engines where the convective and diffractive effects of the mean flow are significant. Also, in this case the acoustic wavelength is generally much smaller than the length scale of the scattering geometry and smaller than the characteristic length scale for variations in the mean flow. The Partition of Unity Method (PUM) [J. M. Melenk and I. Babuška, Comput. Methods Appl. Mech. Eng. **139**, 289–314 (1996)] is a proved efficient numerical method for solving short wave problems in the absence of flow (Helmholtz equation) [Lagrouche *et al.*, Int. J. Numer. Methods Eng. **54**, 1501–1533 (2002)]. The PUM approach is based on the use of a discrete set plane of wave as a local basis for the spatial discretization. When flow is present the wave number of each locally defined plane wave becomes dependent upon the magnitude and direction of the mean flow. The implementation of such scheme for the harmonic acoustic propagation within an irrotational mean flow is proposed in this work. One- and two-dimensional model problems are presented to demonstrate the effectiveness of the approach. [Work supported by EPSRC.]

10:45

2aEA11. Noise control for a ChamberCore cylindrical structure using long T-shaped acoustic resonators. Deyu Li and Jeffrey S. Vipperman (Dept. of Mech. Eng., Univ. of Pittsburgh, 531 Benedum Hall, Pittsburgh, PA 15261, jsv@pitt.edu)

The Air Force Research Laboratory, Space Vehicles Directorate has developed a new advanced composite launch vehicle fairing (referred to as "ChamberCore"). The ChamberCore is sandwich-type structure fabricated from multi-layered composite face sheets separated by channels that form passive acoustic chambers. These acoustic chambers have a potential to create an acoustic resonator network that can be used to attenuate noise inside the closed ChamberCore cylindrical structure. In this study, first, the feasibility of using cylindrical Helmholtz resonators to control noise in a mock-scale ChamberCore composite cylinder is investigated. The targeted frequencies for noise control are the first four acoustic cavity resonances of the ChamberCore cylinder. The optimal position of the Helmholtz resonators for controlling each targeted cavity mode is discussed, and the effects of resonator spacing on noise attenuation are also experimentally evaluated. Next, six long T-shaped acoustic resonators are designed and constructed within the acoustic chambers of the structure and investigated. Several tests are conducted to evaluate the noise control ability of the resonators in the ChamberCore cylinder. Reductions ranging from 3.2 to 6.0 dB were observed in the overall mean-square noise reduction spectrum at the targeted inner cavity resonance frequencies. [Work supported by AFRL/DV.]

11:00

2aEA12. Design and resonant frequency prediction for long T-shaped acoustic resonators. Deyu Li and Jeffrey S. Vipperman (Dept. of Mech. Eng., Univ. of Pittsburgh, 531 Benedum Hall, Pittsburgh, PA 15261, jsv@pitt.edu)

The use of acoustic resonators is an effective way to control cavity resonances in small enclosures. One popular device is the long, T-shaped acoustic resonator which consists of three branches. Two branches (referred to as "Branch-1" and "Branch-2") are co-axial and both have one open end and one closed end, and the third branch (referred to as "Branch-3") is perpendicular to the co-axis and has two open ends. In practical cavity noise control, the optimal position of Branch-3, i.e., the length of Branch-1 or Branch-2 is determined by the mode shape of the controlled cavity mode, and the length of Branch-3 is typically chosen to be as short as possible to minimize the occupied space of the resonator. If the cross-sectional areas are given, the only design parameter is the length of Branch-1 or Branch-2. In this study, three new models are developed to calculate the end corrections for the three branches. The novel theory is also used to design long T-shaped acoustic resonators for control. In addition to the fundamental acoustic resonator mode, higher frequency resonances of the resonator can be used to target a specific cavity mode for control. Several tests are conducted to experimentally evaluate and validate these new models.

11:15

2aEA13. Infrasonic windscreen. Qamar A. Shams, B. Scott Sealey (M.S. 236, NASA Langley Res. Ctr., Hampton, VA 23681), Allan J. Zuckerwar (NASA Langley Res. Ctr., Hampton, VA 23681), and Laura M. Bott (Univ. of Toledo, Toledo, OH 43606)

Infrasonic windscreens, designed for service at frequencies below 20 Hz, were fabricated from a variety of materials having a low acoustic impedance, and tested in a small wind tunnel against four specifications: (1) attenuation of wind-generated sound (at a free-stream wind speed of 9.4 m/s), (2) the transmission of low-frequency sound from a known source (Janis subwoofer), (3) spectrum of sound generated from trailing vortices, and (4) water absorption (to determine the suitability for all-weather service). Windscreen materials included three woods (pine, cedar, and balsa), polyurethane foam, and Space Shuttle tile material. The windscreen outside diameter ranged from 0.0254 to 0.1016 m (1 to 4 inches), and wall thickness from 0.003175 to 0.01905 m (1/8 to 3/4 inch). A windscreen made of polyurethane foam revealed a wind noise attenuation of 10–20 dB from 0.7–25 Hz, transmission coefficient near unity from 10–25 Hz, and a spectral peak of 23 Hz due to vortex-generated sound. Data will be presented for a variety of windscreens.

11:30

2aEA14. A rumbling index development for sound quality analysis of a vehicle. Sang-Kwon Lee and Byung-Soo Kim (Dept. of Mech. Eng., Inha Univ., 253 Yonghyun Dong, Inchon)

Rumbling sound is one of the most important interior sounds of a passenger car, according to the conventional research. The evaluation of sound quality has been focused on the reduction of the A-weighted sound pressure level. However, an A-weighted sound pressure level cannot give the whole story about the sound quality based on human sensibility. The rumble index is required for the evaluation of the sound quality of a vehicle. In order to make a rumble index, the artificial neural network has been used. The sound metrics for the interior sounds of vehicles are used for the input of the neural network. The subjective ratio for the interior sounds of vehicles is used for the target of artificial neural network to train the weights of the neuron. For the train of the artificial neural network, 150 signals for rumbling sound are synthesized. The interior sounds of 16 vehicles are tested by using the trained artificial neural network. The output of artificial neural network becomes the rumbling index and has 90% correlation with the subjective results for interior sounds of 16 vehicles. [Work supported by BK21.]

11:45

2aEA15. Inversion for acoustic impedance using artificial neural network algorithm. Gee-Pinn Too and Suzi Hwang (Dept. of System and Naval Mechatronic Eng., Natl. Cheng Kung Univ., Tainan, Taiwan, z8008070@email.ncku.edu.tw)

A new approach for measuring acoustic impedance is developed by using Artificial Neural Network (ANN) algorithm. Instead of using impedance tube, a rectangular room or a box is simulated with known boundary conditions at some boundaries and a unknown acoustic impedance at one side of the wall. The training data basis for the ANN algorithm is evaluated by Similar Source Method which was developed earlier by Too (1999) for the estimation of interior and exterior sound field. The training data basis is constructed by evaluating of acoustic pressure at a field point with various acoustic impedance conditions at one side of the wall. The simulation result indicates that the prediction of acoustic impedance is very accurate with error percentage under 1%. Also, one point field measurement in the present approach provides a straightforward and easy evaluation than that in the two points measurement of the impedance tube approach.

TUESDAY MORNING, 11 NOVEMBER 2003 — SAN ANTONIO ROOM, 8:30 TO 11:35 A.M.

Session 2aMU

Musical Acoustics and Physical Acoustics: Wind Instruments and Nonlinearity I

Peter L. Hoekje, Chair

Department of Physics and Astronomy, Baldwin-Wallace College, 275 Eastland Road, Berea, Ohio 44017

Chair's Introduction—8:30

Invited Papers

8:35

2aMU1. Linear and nonlinear behavior of human and artificial lip reeds. Murray Campbell and Orlando Richards (School of Phys., Univ. of Edinburgh, Edinburgh EH9 3JZ, UK)

In a musical instrument of the lip reed aerophone class, the flow of air from the player's lungs into the resonating air column is modulated by the periodic opening and closing of the pressure-controlled valve formed by the player's lips. The nature of the operation of this valve has been the subject of considerable study in recent years. Since the pressure–flow relationship is strongly nonlinear, the behavior of the coupled system of lips and air column can only be modeled using the methods of nonlinear dynamics. Extensive studies of artificial lip reeds, in which the lips are simulated by water-filled latex tubes, have shown them to be capable of reproducing musically important features of human playing, including the lipping of notes both below and above an acoustic resonance of the air column. Measurements of the linear response of artificial reeds have guided the development of more realistic models of the lip reed, while studies of both real and artificial lips using a high-speed digital camera have shed fresh light on the nature of the lip motion at the large amplitudes typical of loud playing. [Work supported by EPSRC.]

9:05

2aMU2. Propagation of nonlinear acoustic waves and shocks in variable cross-section resonators. Bart Lipkens (Mech. Eng. Dept., Western New England College, 1215 Wilbraham Rd., Springfield, MA 01119)

Relative peak acoustic pressures in wind instruments such as oboes, trumpets, and trombones can reach levels of 10%. At these elevated levels nonlinear acoustic effects have to be taken into account to accurately represent the acoustic wave propagation in these wind instruments. A short survey is presented of some of the past work in nonlinear resonant oscillations in tubes. Included in this discussion is the effect of shaping the resonator on nonlinear wave propagation. A one-dimensional model for the analysis of nonlinear acoustic waves in a variable cross-section resonator is introduced [Ilinski *et al.*, J. Acoust. Soc. Am. **104**, 2664–2674 (1998)]. The resonator is assumed to be of an arbitrary axisymmetric shape. The model equations are derived from the fundamental gas dynamic equations for an ideal gas. The effects of bulk viscous absorption and absorption in the viscous boundary layer are included. Nonlinear spectral equations are developed which are integrated numerically to obtain the solution. The model has been modified to include a radiation impedance and the excitation of a harmonic spectrum. The effect of resonator shape, radiation condition, and harmonic excitation on nonlinear wave propagation is discussed.

9:35

2aMU3. Localized and cumulative nonlinearity in wind instruments. Joel Gilbert (Laboratoire d'Acoustique de l'Universit du Maine—UMR CNRS 6613, Ave. Olivier Messiaen, 72085 Le Mans, Cedex 9, France)

Nonlinearities are very common in wind instruments. A crucial one localized at the input of the wind instrument is responsible for the sound production mechanism. As an illustration, some recent measurements done at clarinet mouthpieces will be shown. Some other localized nonlinear effects take place at the open tube ends. They imply extra losses whose amount depends on the internal geometry of the termination. They control the sound extinction phenomena. It will be shown how the playing range of a clarinet-like instrument is determined by these extra losses. Besides localized nonlinearity, cumulative nonlinearity effects are present as well. The cumulative nonlinear propagation phenomena along the tube of brass instruments can lead to shock waves obtained when the player is playing very loudly with a ''brassy sound.''

10:05

2aMU4. Interaction of pipe tone and edge tone in organ pipe oscillation. A. W. Nolle (Dept. of Phys., Univ. of Texas, Austin, TX 78712, nolle@mail.utexas.edu)

Inharmonic partials in an organ pipe tone can occur in clusters, at uniform frequency separation. The primary object of this discussion is to demonstrate that these clusters arise in a nonlinear process: Edge tone oscillation is modulated at the fundamental frequency of the pipe tone. The mechanism is demonstrated by a laboratory analog, in which periodic deflection of a jet producing an edge tone is forced by a transverse acoustic flow, and also by a numerical analog. Additional examples show that the simultaneous occurrence of a pipe tone and an unmodulated edge tone can occur at low amplitudes of oscillation, where the nonlinear interaction is unimportant.

10:35

2aMU5. Feathering collisions in beating reed simulation. Tamara Smyth (Ctr. for Computer Res. in Music and Acoust., Dept. of Music, Stanford Univ., Stanford, CA 94305-8180), Jonathan S. Abel (Univeral Audio, Inc., Santa Cruz, CA 95063-3818), and Julius O. SmithIII (Stanford Univ., Stanford, CA 94305-8180)

Pressure controlled valves are the primary sound production mechanisms for woodwind and brass musical instruments, as well as for many bioacoustic vocal systems such as the larynx and syrinx (the vocal organ in birds). During sound production, air flow sets a reed or membrane into motion creating a variable height in the valve channel and, potentially, periodically closing the channel completely. Depending on how this event is handled, an abrupt termination of air flow between open and closed states can cause undesirable discontinuities and inaccuracies in a discrete-time simulation—particularly at relatively low audio sampling rates. A solution was developed by re-examining the behavior of the differential equation governing volume flow through a pressure-controlled valve, paying particular attention to this rather troublesome transition. A closed-form solution for the time evolution of volume flow is given and used to derive an update for volume flow. The result is a smoother, more accurate, and nearly alias-free transition from open to closed. ''Feathered collisions'' of this nature can refine the sound quality produced by the numerical simulation of beating reeds, such as in clarinets, at typical audio sampling rates.

Contributed Papers

11:05

2aMU6. Error metrics for predicting discrimination of original and spectrally altered musical instrument sounds. James W. Beauchamp (School of Music and Dept. of Elec. & Computer Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801, jwbeauch@uiuc.edu) and Andrew Horner (Hong Kong Univ. of Sci. & Technol., Clear Water Bay, Kowloon, Hong Kong)

The correspondence of various error metrics to human discrimination data was investigated. Time-varying harmonic amplitude data were obtained from spectral analysis of eight musical instrument sounds (bassoon, clarinet, flute, horn, oboe, saxophone, trumpet, and violin). The data were altered using fixed random multipliers on the harmonic amplitudes, and the sounds were additively resynthesized with estimated average spectral errors ranging from 1% to 50%. Listeners attempted to discriminate the randomly altered sounds from reference sounds resynthesized from the original data. Then, various error metrics were used to calculate the spectral differences between the original and altered sounds, and the R^2 correspondence between the error metrics and the discrimination data was measured. A relative-amplitude spectral error metric gave the best correspondence to average subject discrimination data, capturing over 90% of the variation relative to a Fourth-order regression curve, although other formulas gave similar results. Error metrics which used a small number of representative analysis frames gave results which compared favorably to using all frames of the analysis.

11:20

2aMU7. The rams horn in western history. David Lubman (David Lubman & Assoc., 14301 Middletown Ln., Westminster, CA 92683)

The shofar or rams horn—one of the most ancient of surviving aerophones—may have originated with early Neolithic herders. The shofar is mentioned frequently and importantly in the Hebrew bible and in later biblical and post-biblical literature. Despite its long history, contemporary ritual uses, and profound symbolic significance to western religion, no documentation of shofar acoustical properties was found. Since ancient times, shepherds of many cultures have fashioned sound instruments from the horns of herd animals for practical and musical uses. Shepherd horns of other cultures exhibit an evolution of form and technology (e.g., the inclusion of finger holes). The shofar is unique in having retained its primitive form. It is suggested that after centuries of practical use, the shofar became emblematic of the shepherd culture. Ritual use then developed, which froze its form. A modern ritual rams horn played by an experienced blower was examined. This rather short horn was determined to have a source strength of 92 dB (A) at 1 m, a fundamental frequency near 420 Hz, and maximum power output between 1.2 and 1.8 kHz. Sample sounds and detection range estimates are provided.

Session 2aNS

Noise and Architectural Acoustics: Acoustical Design Issues for the Health Care Industry

Kerrie G. Standlee, Cochair
Daly-Standlee and Associates, 4900 Southwest Griffith Drive, Suite 216, Beaverton, Oregon 97005

Jack B. Evans, Cochair
Engineering, Vibration and Acoustics, JEAcoustics, 5806 Mesa Drive, Suite 380, Austin, Texas 78731-3742

Chair's Introduction—9:00

Invited Papers

9:05

2aNS1. Acoustical criteria for hospital patient rooms: Resolving competing requirements. Bennett M. Brooks (Brooks Acoust. Corp., 27 Hartford Turnpike, Vernon, CT 06066, brooks@brooks-acoustics.com)

The acoustical criteria for patient rooms in hospitals, nursing homes, and rehabilitation facilities may be based on several needs. One important requirement is that noise levels in the room be conducive to restful sleep. Also, caregivers must have easy auditory and visual access to the patients, and be able to hear vital sign monitor alarms. This often means that patient rooms are located near central nurse stations and that patient room doors are left open. Further, the recently published federal privacy standards developed by the U.S. Department of Health and Human Services (HSS) under the Health Insurance Portability and Accountability Act (HIPAA) require that ''appropriate physical safeguards'' be put in place to protect the confidentiality of patient health information. The simultaneous and competing requirements for speech privacy, caregiver access, and good sleeping conditions present a serious acoustical challenge to health care facility designers. Specific facility design issues and potential solution strategies are presented.

9:25

2aNS2. Noise levels, spectra, and operational function of an occupied newborn intensive care unit built to meet recommended permissible noise criteria. M. Kathleen Philbin (Children's Regional Hospital at Cooper Univ. Hospital, One Cooper Plaza, Dorrance 755, Camden, NJ 08103, philbika@umdnj.edu) and Jack B. Evans (JEAcoust., Inc., Austin, TX 78731)

A group of clinical experts developed recommended permissible noise criteria for newly constructed or renovated hospital nurseries [Philbin *et al.*, J. Perinatol. **19**, 559–563 (2000); R. White, *ibid.* **23**, S1–22 (2003)]. These criteria are based principally on research regarding wake-up thresholds for term newborns and speech interference levels for adults. These criteria are: The overall continuous A-weighted, slow response, sound level at any bed or patient care area shall not exceed: (1) an hourly Leq of 50 dB, (2) an hourly L10 of 55 dB, and (3) a 1-s Lmax of 70 dB. A new hospital building was designed to meet these criteria by using specific acoustical criteria for the structure and space arrangement [J. B. Evans and M. K. Philbin, J. Perinatol. **20**, S105–S112 (2000)]. Acoustical criteria for sound isolation, background NC, structural vibration, and reverberation will be presented along with space arrangements that ensure staff efficiency, clinical safety, and family privacy. Post-occupancy measurements of sound levels and spectra along with photographs of a nursery in operation will be presented to illustrate how an ICU can have a quiet, highly functioning intensive care environment while meeting the operational goals and acoustical criteria.

9:45

2aNS3. Floor vibration evaluations for medical facilities. Chad N. Himmel (JEAcoust., 5806 Mesa Dr., Ste. 380, Austin, TX 78731, himmel@jeacoustics.com)

The structural floor design for new medical facilities is often selected early in the design phase and in renovation projects, the floor structure already exists. Because the floor structure can often have an influence on the location of vibration sensitive medical equipment and facilities, it is becoming necessary to identify the best locations for equipment and facilities early in the design process. Even though specific criteria for vibration-sensitive uses and equipment may not always be available early in the design phase, it should be possible to determine compatible floor structures for planned vibration-sensitive uses by comparing conceptual layouts with generic floor vibration criteria. Relatively simple evaluations of planned uses and generic criteria, combined with on-site vibration and noise measurements early in design phase, can significantly reduce future design problems and expense. Concepts of evaluation procedures and analyses will be presented in this paper. Generic floor vibration criteria and appropriate parameters to control resonant floor vibration and noise will be discussed for typical medical facilities and medical research facilities. Physical, economic, and logistical limitations that affect implementation will be discussed through case studies.

10:05–10:20 Break

10:20

2aNS4. Structure-borne sound from magnetic resonance imaging systems. Eric E. Ungar and Jeffrey A. Zapfe (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138)

Magnetic resonance imaging (MRI) systems are known to produce a considerable amount of audible noise. The recent tendency to install such systems on above-grade floors has led to increasing concerns about structure-borne noise transmission from the MRI to adjacent occupied areas. This paper presents the results of a study in which structure-borne noise forces produced by two operational MRI systems were determined via measurement of the floor vibrations induced by the systems and of the impedance of their supporting floors. Forces with known spectra were applied to the floors of planned MRI suites in a hospital extension and the corresponding noise in adjacent areas was measured. Similarly, airborne noise was introduced in the planned suites and the related noise in adjacent areas was measured. The results then were scaled to correspond to the measured MRI forces and airborne noise. It was found that in areas below the planned MRI installations structure-borne noise would predominate, unless it is mitigated. Structure-borne noise isolation of MRI systems, whose environments must meet stringent vibration criteria, is discussed briefly.

10:40

2aNS5. The MRI: A noise source of concern in the health care industry. Kerrie G. Standlee and Joseph C. Begin (Daly-Standlee & Assoc., Inc., 4900 SW Griffith Dr., Ste. 216, Beaverton, OR 97005, jbegin@acoustechgroup.com)

Two recent trends in the development and use of magnetic resonance imaging (MRI) equipment have created challenges for acoustical engineers: (1) the trend toward more powerful MRI machines with greater magnetic field strengths, and (2) the tendency of health care facilities to locate these machines, which were previously located in basements or on grade, on upper floors adjacent to (and in some cases above) other critical use areas. For newer, 3-T MRI machines, sound levels well over 100 dBA in the examination room are common. Along with these trends, some equipment manufacturers are now providing design recommendations to address the issues of airborne and structure-borne noise within hospitals and clinics. In addition, MRI manufacturers sometimes have strict requirements for acceptable levels of building vibration from other sources, to prevent potential image quality problems. This paper discusses experience gained during the course of addressing MRI-generated noise on several projects. Data for airborne sound levels measured inside MRI rooms and adjacent rooms and vibration levels measured below MRI units will be presented.

11:00

2aNS6. MRI structure borne noise, in and out. K. Anthony Hoover (Cavanaugh Tocci Assoc., 327 F Boston Post Rd., Sudbury, MA 01776, thoover@cavtocci.com)

MRI equipment typically requires very low levels of ambient floor vibration. However, this same equipment often generates levels of vibration that are sufficient to cause complaints of excessive MRI noise in nonadjacent rooms. This paper will review two projects. The first investigates levels of floor vibrations caused by flyovers of medivac helicopters. The second investigates a structural floor slab designed to satisfy MRI ambient-vibration criteria, but, as predicted, allowed for excessive structure borne MRI noise. Difficulties with standard vibration isolation of MRI equipment will be discussed.

11:20

2aNS7. Environmental noise issues associated with medical facility HVAC equipment. Ted N. Carnes, Howard K. Pelton, and Daniel W. Saenz (Pelton Marsh Kinsella, 1420 W. Mockingbird Ln., Ste. 400, Dallas, TX 75247, PeltonHK@c-b.com)

Medical facilities comprise a variety of types of buildings from research facilities to hospitals to professional office buildings to name but three. Many new hospitals are being located in close proximity to residential housing. Furthermore, as hospitals expand the buildings are closer to residential communities. Thus, the community noise aspects of these facilities are of even more importance than ever. This paper will examine how community noise intrusions can occur without proper planning, and how this can be avoided by knowing the community noise criteria, working closely with the design team to assist them with the proper planning and noise control design. Case histories will be presented that will include a hospital expansion with large rooftop equipment, new cooling towers and how these can be modeled as part of the overall hospital campus model to determine the community noise impact. In addition, a new chiller plant expansion adjacent to the residential areas will be examined, and how this can be modeled to show the various aspects of the community noise impact.

Session 2aPA

Physical Acoustics: Thermoacoustics and Resonators

Gregory W. Swift, Chair
Condensed Matter and Thermal Physics Group, Los Alamos National Laboratory, Los Alamos, New Mexico 87545

Contributed Papers

9:00

2aPA1. Regenerator-based thermoacoustic refrigerator for ice cream storage applications. Matthew E. Poese, Robert W. M. Smith, and Steven L. Garrett (Penn State Appl. Res. Lab., P.O. Box 30, State College, PA 16804, poese@psu.edu)

A regenerator-based chiller has been built in the ''bellows bounce'' style [J. Acoust. Soc. Am. **112**, 15 (2002)] to replace the vapor compression system in an ice cream sales cabinet. It utilizes a 6-in.-diam metal bellows to form a compliant cavity that contains the dynamic pressure oscillation (>50 kPa). The stiffness of the gas trapped in the bellows is resonated against the mass of the bellows-cap and the mass of a moving-magnet linear motor which is capable of high (>85%) electro-acoustic efficiency. A second resonator, operated well below its natural frequency, uses the gas stiffness of a 1-l volume nested within the bellows and the inertia of an ordinary loudspeaker cone to create the pressure difference across the regenerator that drives gas flow that is in-phase with pressure. The mass of the cone can be adjusted to vary the multiplication factor that is typically 5%–10% greater than the dynamic pressure within the bellows. The loudspeaker cone suffers none of the hydrodynamic losses associated with an acoustic inertance and eliminates problems with dc gas flow in the energy feedback path. The cold heat exchanger forms one surface of the pressure vessel permitting direct contact with any thermal load. [Work supported by Ben and Jerry's Homemade.]

9:15

2aPA2. The performance of a high-frequency thermoacoustic-Stirling engine. Kevin J. Bastyr and Robert M. Keolian (Penn State Univ., P.O. Box 30, State College, PA 16804)

A thermoacoustic-Stirling engine that operates at 400 Hz with a working fluid of 1-MPa helium is constructed. For proper acoustic phasing in this engine's regenerator, an acoustic power feedback path exists in the form of an annulus surrounding the regenerator. This feedback path is obtained by suspending an insulated, stainless steel sleeve containing a wire mesh regenerator, which is flanked by two heat exchangers, a short distance from one end of the larger diameter resonator. The ambient heat exchanger is a shell and tube exchanger, while the hot heater consists of nichrome ribbon wound on an aluminum silicate frame. Gedeon streaming is prevented by a diaphragm covering the end of the stainless steel sleeve adjacent to the ambient heat exchanger. A variable acoustic load provides a convenient means of testing this engine at various hot heater temperatures, while operating at different acoustic pressure amplitudes effects the acoustic power generated by the engine. [Work supported by ONR.]

9:30

2aPA3. Resonant self-pumped thermoacoustic heat transfer. Greg Swift and Scott Backhaus (Condensed Matter and Thermal Phys. Group, Los Alamos Natl. Lab., Los Alamos, NM 87545, swift@lanl.gov)

As a thermoacoustic engine or refrigerator is scaled up to higher power, it generally needs heat exchangers of a larger cross-sectional area. Constrained to lie adjacent to a stack or regenerator, large heat exchangers must use intricate geometries to interweave the thermoacoustic working gas and an external fluid (e.g., ambient cooling water or hot combustion gas) to bring them into intimate thermal contact. Such heat exchangers have many thermoacoustic passages in parallel, often entailing many parts and many joints that must be leak tight. Resonant self-pumped thermoacoustic heat transfer is a new alternative in which the thermoacoustic-to-external heat-transfer surface area is in the form of one long tube (or a few) extending far from the associated stack or regenerator. With a resonant sound wave in the tube and with diode-like nonlinear flow elements near the wave's velocity maxima, a strong time-averaged circulation of the working gas carries the heat. This concept will be described and preliminary experimental results will be presented. [Work supported by DOE's Office of Science and by Praxair, Inc.]

9:45

2aPA4. Driver parameter characterization method for a thermoacoustic cooler. Insu Paek, Luc Mongeau, and James E. Braun (Ray W. Herrick Labs., Purdue Univ., West Lafayette, IN 47907)

The efficiency of transducers used in electromechanically driven standing wave thermoacoustic coolers is governed by six parameters. The maximum efficiency is obtained around the frequency at which the total electrical input reactance is zero. The parameters in a linear circuit representation of the dynamic behavior of the driver are often frequency and amplitude dependent. This nonlinear response is due to the large acoustic power delivered and the resonant nature of the acoustic load. A method was developed to determine quickly and accurately driver equivalent parameters from independent measurements of the input voltage and current, as well as the piston acceleration and pressure. A system identification technique was used. Transfer function measurements were performed for a few driving frequencies close to the optimal operating condition of the thermoacoustic cooler. The data were used to calculate driver parameters off-line. Parameter values obtained for low piston force and displacement amplitudes were found to be significantly different than their values for larger amplitudes. The method can be used to find the optimal driver parameters for target operating conditions of a particular thermoacoustic cooler.

10:00

2aPA5. Heat transfer of heat exchangers in a standing wave thermoacoustic cooler. Insu Paek, James E. Braun, and Luc Mongeau (Ray W. Herrick Labs., Purdue Univ., West Lafayette, IN 47907)

Standing wave thermoacoustic coolers require heat exchangers to move heat in and out of the working gas. Within most finned-tube type heat exchangers, heat is transferred from the oscillating gas to a liquid flowing through the tubes. Experiments were performed to evaluate the thermal performance of heat exchangers immersed in a standing wave thermoacoustic cooler. The gas-side convection heat transfer coefficient was calculated from the measured data, and compared to steady flow heat transfer model predictions. A simple relation between the Reynolds number in steady flow and the time dependent acoustic Reynolds number was obtained. The results suggest that substituting this relation into known steady flow correlations may yield accurate predictions of the heat transfer coefficient in oscillating flow.

10:15

2aPA6. Minor losses in high-amplitude acoustic resonators with varying changes in cross section. Andrew J. Doller and Anthony A. Atchley (Grad. Prog. in Acoust., Penn State Univ., University Park, PA 16802, adoller@psu.edu)

In a previous paper [J. Acoust. Soc. Am. **112** (2003)], measurements of time-averaged pressure at the closed end of a high-amplitude resonator having a non-uniform cross section were presented. The motivation for this continued work is to better understand minor losses in high-amplitude oscillatory flows through changes in resonator cross section. The resonators consist of two sections of straight brass pipe of different diameters and joined through either step-like or conical couplers. They are driven at the end of the large diameter pipe with a rigid piston. The end of the small tube is rigidly terminated. In the current paper, data obtained from an improved experimental apparatus will be presented. The improvements consist of better temperature control and the ability to measure the power delivered to the resonator by the piston. Subtraction of the linear acoustic boundary-layer losses from the measured input power provides an upper estimate on the power dissipated through other mechanisms. The dependence of the time-averaged pressure and power dissipation on drive amplitude, junction type, and junction location in the standing wave will be discussed. [Research supported by ONR.]

10:30

2aPA7. Numerical simulation of hydrodynamic and thermal effects around a 2-D stack plate in a thermoacoustic refrigerator. David Marx and Philippe Blanc-Benon (Ecole Centrale de Lyon, Ctr. Acoustique, LMFA UMR CNRS 5509, 69134 Ecully Cedex, France)

Two-dimensional Navier–Stokes equations for an unsteady and compressible flow are solved around a 2-D stack plate of a thermoacoustic refrigerator. The computational domain is a resonator slice including the resonator end but not the acoustic source. The standing wave is sustained in the domain using the characteristics method. The same methods were already used for the simulation of the coupling between stack plates and heat exchangers (AIAA Paper No. 2003-3150). The oscillating flow above the 2-D plate induces a periodic vortex creation at the ends of the plate. Numerical results will be compared to PIV measurements. The effect of the acoustic Mach number and geometrical parameters are studied. The temperature difference between the ends of the plate is also investigated and compared to linear theory. Some deviations can be attributed to the difference between the mean temperature gradient in the plate and the mean temperature gradient in the fluid. A streaming flow is also observed above the plate. [Work supported by DGA.]

10:45

2aPA8. Open cycle traveling wave thermoacoustics. Nathan T. Weiland and Ben T. Zinn (Schools of Mech. and Aerosp. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, gte852f@mail.gatech.edu)

The concepts and basic operation of open cycle traveling wave thermoacoustic devices are discussed for the case of a simple thermoacoustic engine, where a steady flow of hot gas replaces the hot heat exchanger as the means for supplying heat to the engine. It has been determined that a large mean temperature difference must exist where the mean flow first meets the regenerator for the engine to produce any acoustic power, due to the joining conditions between the ideally isothermal environment in the regenerator and the isentropic environment of the adjoining open duct that contains the mean flow. The physics behind this temperature difference is discussed, as well as its effects on the various energy fluxes in the thermoacoustic engine. In particular, it is shown that the acoustic power output and thermal efficiency of the engine can be maximized with respect to the magnitude of this temperature difference.

11:00

2aPA9. Investigation of harmonic generation in thermoacoustic engines. James E. Parker, Mark F. Hamilton, and Yurii A. Ilinskii (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

One factor limiting the efficiency of thermoacoustic engines and refrigerators is the generation of harmonics. Not only do thermoviscous losses increase in relation to the generation of higher frequencies, but the stack is not designed to optimize thermoacoustic processes at these frequencies. We describe here a semi-analytical investigation of nonlinear effects associated with harmonic generation. The approach is based on a regular perturbation expansion of the acoustical quantities, combined with the assumptions and methodology underlying the linear theory developed by Rott. At first order, Rott's equations for the pressure and particle velocity are obtained and solved numerically. The second-order system consists of Rott's equations for the second harmonic driven by a forcing function containing products of the first-order solutions. The goal is to solve the third-order system, given by Rott's equations for the fundamental driven by products of the first- and second-order solutions, to determine the nonlinear correction to the fundamental component. The investigation was not completed at the time of writing, and we therefore present here the intermediate results for second-harmonic generation. These results provide partial explanations of how harmonic generation influences thermoacoustics. [Work supported by ONR and ARL:UT IR&D.]

11:15

2aPA10. Multiple existence of steady state solutions of acoustic streaming in a long two-dimensional rectangular box. Takeru Yano (Div. of Mech. Sci., Grad. School of Eng., Hokkaido Univ., Sapporo 060-8628, Japan, yano@mech-me.eng.hokudai.ac.jp)

The large amplitude standing wave excited in a resonator induces acoustic streaming of Rayleigh type outside the acoustic boundary layer on the wall of the resonator. The streaming motion with large Reynolds number is examined numerically in relatively long two-dimensional rectangular boxes. The two-dimensional incompressible Navier–Stokes equations with no external force are used as the governing equations for the streaming velocity. The steady velocity component at the outer edge of the acoustic boundary layer, which induces Rayleigh type streaming, is employed as the boundary condition for the Navier–Stokes equations. By using a finite-difference method, the existence of multiple steady state solutions is demonstrated. Some of the numerically obtained flow patterns bear a resemblance to those obtained experimentally. It may be remarked that all the flow patterns induced by the fifth mode standing wave in a relatively long box can be regarded as combinations of those induced by the second mode standing wave in a short box.

11:30

2aPA11. Update on the simulation of streaming in tapered resonators. Brian C. Tuttle and Victor W. Sparrow (Grad. Prog. in Acoust., Penn State Univ., 217 Appl. Sci. Bldg., University Park, PA 16802)

In a previous report [J. Acoust. Soc. Am. **110**, 2652 (2001)] the authors described a plan to simulate nonlinear acoustic streaming by way of a MacCormack finite-difference scheme running on a cluster-type multiprocessor computer. The object was to investigate how the tapering of a resonator tube affects the acoustic streaming. The experiments of Olson and Swift predict that a temperature-dependent viscosity in conjunction with tube geometry can minimize Rayleigh streaming for a particular taper angle. Recent progress in the present research includes a careful perturbation analysis of the model equations to incorporate additional nonlinear terms that are important to simulate flow near a boundary. A modified computational grid necessary to accommodate the tapered domain as well as the refined boundary layer is discussed. The method of driving the tube and issues arising in the simulation starting conditions are also described. [Work supported in part by ONR.]

Session 2aPP

Psychological and Physiological Acoustics: A Potpourri for the Informative

W. Jay Dowling, Chair
Program in Cognitive Sciences, University of Texas–Dallas, Richardson, Texas 75083-0688

Chair's Introduction—9:00

Contributed Papers

9:05

2aPP1. Stimulus-driven and knowledge-driven processes in attention to warbles. W. Jay Dowling (Prog. in Cognit. Sci., Univ. of Texas–Dallas, Richardson, TX 75083-0688, jdowling@utdallas.edu) and Barbara Tillmann (Univ. Claude Bernard Lyon I, Lyon, France)

Listeners identified warbles differing in amplitude-modulation rate (3–10 Hz). And measured RT while listeners maintained above 90% correct responses. After a practice session listeners identified target warbles following stimulus-driven or knowledge-driven cues. The stimulus-driven cue was a 250-ms "beep" at the target pitch (valid) or another pitch (invalid); the knowledge-driven cue was a midrange "melody" pointing to the target pitch (always valid). A 500-ms target warble followed the cue after delays of 0–500 ms (250–750 ms SOA). The listener pressed a key to indicate "slow" or "fast." RTs were shortest at the briefest delay. In contrast to results from a memory task, RTs here were much shorter, and we found no evidence for IOR or attentional blink. Listeners began generating responses while the target was still sounding. Invalid "beeps" slowed responses at the briefest (but not the longer) delays; adding a valid "beep" to the valid "melody" did not speed responses.

9:20

2aPP2. Interpreting electrically evoked emissions using a finite-element model of the cochlea. Niranjan V. Deo (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48109, ndeo@umich.edu), Karl Grosh (Univ. of Michigan, Ann Arbor, MI 48109), and Anand Parthasarathi (Bose Corp., Framingham, MA)

Electrically evoked otoacoustic emissions (EEOAEs) are used to investigate *in vivo* cochlear electromechanical function. Electrical stimulation through bipolar electrodes placed very close to the basilar membrane (in the scala vestibuli and scala tympani) gives rise to a narrow frequency range of EEOAEs, limited to around 20 kHz when the electrodes are placed near the 18-kHz best frequency place. Model predictions using a three-dimensional inviscid fluid model in conjunction with a middle ear model [S. Puria and J. B. Allen, J. Acoust. Soc. Am. **104**, 3463–3481 (1998)] and a simple model for outer hair cell activity [S. Neely and D. Kim, J. Acoust. Soc. Am. **94**, 137–146 (1993)] are used to interpret the experimental results. To estimate effect of viscosity, model results are compared with those obtained for a viscous fluid. The models are solved using a 2.5-D finite-element formulation. Predictions show that the high frequency limit of the excitation is determined by the spatial extent of the current stimulus. The global peaks in the EEOAE spectra are interpreted as constructive interference between electrically evoked backward traveling waves and forward traveling waves reflected from the stapes. Steady state response predictions of the model are presented.

9:35

2aPP3. Global electro-structural-acoustic filtering in the cochlea. Sripriya Ramamoorthy and Karl Grosh (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48105)

The active filtering in the cochlea is now believed to be due to the piezo-motility of the Outer Hair Cell (OHC), and the modulation of stereocilia resistance. Physiologically, the OHC resting potential is known to be modulated by the K+ ions in the cochlear fluid entering or leaving the OHC via gated channels, in response to structural motion, which in turn couples the electrical potential in the cochlear fluids to OHCs piezomotility. However, the electrotonic space-constant of the ionic fluid in the cochlea is not negligible compared to the structural-acoustic wavelength for all frequencies, thus limiting the currently existing models that adopt local structural-electrical interaction. The electrical wave interacts globally with the structural-acoustic wave. This paper proposes a three-dimensional global electro-structural-acoustic linear finite element model to analyze the coupled behavior in the active cochlea. The OHC is modeled as a piezoelectric bar. A cable model is used for the electrical voltages, and a micromechanical model based on Dallos (2002) is proposed. Modeling is maintained as physiological as possible. The active response of the cochlea to acoustic input, Electrically Evoked Oto-Acoustic Emissions, efferent feedback, and the potential of IHC in response to acoustic or electrical input are studied.

9:50

2aPP4. A nonlinear cochlear model with the outer hair cell piezoelectric activity. Xiaoai Jiang and Karl Grosh (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward, Ann Arbor, MI 48109, jiangxa@umich.edu, grosh@umich.edu)

In this paper we present a simple cochlear model which captures the most important aspect of nonlinearity in the cochlea—the nonlinearity caused by the piezoelectric-like activity of outer hair cells and the variable conductance of the outer hair cell stereocilia. A one-dimensional long-wave model is built to simulate the dynamic response of the fluid-loaded basilar membrane. The basilar membrane is simulated as isolated linear oscillators along the cochlear length, and its motion is coupled with the fluid pressure and the nonlinear force produced by the outer hair cells. As the basilar membrane moves, the fluid shears stereocilia, and the resulting ion flow changes the transmembrane potential of the outer hair cells and subsequently their length, leading to further movement of the basilar membrane. The piezoelectric-like activity of the outer hair cell is simulated by a current source, and stereocilia motion is modeled as a varying conductance that changes as the basilar membrane moves. A solution in the time domain will be presented. [Work supported by NIH.]

10:05

2aPP5. Effect of loudness on the haptic force-feedback perception in virtual environments. M. Ercan Altinsoy, Jens Blauert (Inst. of Commun. Acoust., Ruhr Univ. Bochum, D-44780 Bochum, Germany), and Richard H. Y. So (Hong Kong Univ. of Sci. & Technol.)

Due to increasing usage of the haptic modality in virtual reality applications, the perceptual aspects of the interaction between haptic and auditory modalities is becoming more important. In our daily life, force-feedback which we perceive in our hands after beating a drum as well as the loudness of the drum sound give us the required information about how much force we have applied when playing a drum. In this study, psychophysical experiments were conducted to investigate the effect of loudness on the haptic force-feedback perception (strength) by playing a virtual drum. Subjects were presented (1) only haptic force-feedback information, (2) only auditory information, and (3) auditory and haptic in-

formation together. In each experiment, subjects were asked how much force they had applied when playing the virtual drum. The results show that the magnitude of strength increases with increasing loudness in spite of no change in force-feedback as generated by a virtual drum and applied to the subject's hand. These results contribute to our knowledge on how to effectively combine haptic and auditory information in virtual environments. [Work supported by IGSN, Ruhr University Bochum.]

10:20–10:35 Break

10:35

2aPP6. Effect of a non-target sound event on motion perception of a target sound event. Tsuruyo Nishida (Dept. of Elec. and Computer Eng., Gifu Natl. College of Technol., 2236-2 Kamimakuwa, Shinsei-cho, Motosu-gun, Gifu 501-0461, Japan) and Kazuhiko Kakehi (Grad. School of Information Sci./CIAIR, Nagoya Univ., Furo-cho, Chikusa, Nagoya 464-8601, Japan)

Most psychoacoustics studies on sound localization and motion perception have been conducted for only one sound event. In this report, one- or two-sound events were presented and the direction and motion perception were investigated using two-sound phenomena to produce motion perception; the first was apparent motion and the second was a synthesized sound image. There were no significant differences in the correct response percentage for both two-sound phenomena. The correct response percentage for the direction of motion was significantly worse for two-sound events than that for one-sound event, even if the two-sound events are separately perceived in terms of frequency, space domain, and temporal order. The analysis of wrong responses showed that the perception depends on IEOIs and the combination of target and non-target moving conditions (e.g., moving or still). Furthermore, it was clarified with signal detection theory that the motion perception of target was strongly affected by the non-target conditions. It means that the process of perceiving the direction and motion of the target might be different than that of perceiving the sound feature.

10:50

2aPP7. Review and comparison of methods using spectral characteristics for the purposes of CASA, Computational Auditory Scene Analysis. Jerry Gregoire (Montana State Univ., 622 Cobleigh Hall, Bozeman, MT 59717, jgregoire@ece.montana.edu)

The brain has a remarkable ability to separate signals from a mixture. Although much progress has been made, the goal of replicating this has eluded us. Bregman (90) suggests that the brain divides this analysis into low or primitive, and high or schema processes. Schema processes require memory and subsequent interpretation. Conversely, primitive processes operate only on the innate properties of the incoming mixture and the sources that created it. The major parameters are intra aural time differences, spectrum, onset and offset, phase, and amplitude of each source. In this paper I review and compare previously proposed methods to analyze the frequency spectrum. This includes estimation of the fundamental frequency and partials produced by each source in the mixture.

11:05

2aPP8. Exposure smart hearing protector for reducing noise-induced hearing loss. Ed Nykaza and Tom Frank (Dept. of Acoust., Penn State, University Park, PA 16802, etn106@psu.edu)

The Exposure Smart Hearing Protector (ESHP) is a new device that can be used for measuring noise exposure levels (NELs) and the prevention of noise induced hearing loss (NIHL). The ESHP consists of two microphones, located in a right and left earplug, that are connected to a dosimeter. In practice, the user wears the ESHP. When the noise level exceeds a safe dose a warning light comes on. The user then inserts the earplugs. If the earplugs are correctly inserted and the noise level in the user's earcanal is below a safe level the warning lights go off. As a result, the ESHP measures the user's total daily noise exposure (unprotected and protected). To increase the efficiency of using the ESHP for preventing NIHL, the user downloads the information stored in the ESHP via a scanner into user friendly-software. The software can be used not only to record a user's daily NELs, but more importantly to determine if the user needs intervention because the NELs exceed a safe level. The purpose of this poster session is to demonstrate the ESHP and software, and to report the results of a pilot study. [Work supported by NIOSH/CDC Grant No. U60/CCU 315855.]

11:20

2aPP9. Subjective assessment for the number of channel signals to realize sound field based on wavefield synthesis. Toshiyuki Kimura (Res. Fellow of the Japan Society for the Promotion of Sci./CIAIR, Nagoya Univ., Furo-cho, Chikusa-ku, Nagoya 464-8601, Japan), Kazuhiko Kakehi, Kazuya Takeda, and Fumitada Itakura (Nagoya Univ., Furo-cho, Chikusa-ku, Nagoya 464-8603, Japan)

It is very important to study the affect of the number of channel signals in the sound field reproduction technique based on wavefield synthesis. Though there are many studies of objective affect such as wavefront accuracy, there are not as many studies of the subjective affect such as sound field perception. Therefore, the subjective assessment was designed to evaluate the effect of the number of channel signals, which were synthesized by convolving a sound source to room transfer functions of free field, on the directional perception. In the low-reverberation room, a circle loudspeaker array with a radius of 2 m was set on the horizontal plane. The subjects evaluated the directional perception of the sound image reproduced at distances of 3 and 4 m by playing simultaneous channel signals from the loudspeakers set at intervals of 10, 15, 20, 30, and 45 degrees, respectively. As a result, it was subjectively confirmed that 24 channel signals were enough to reproduce the directional perception of 5 degrees accuracy if the control area was the circle of radius 2 m.

11:35

2aPP10. Coupled volume/double slope subjective listening test. Michael Ermann and Rebecca Stuecker (Virginia Tech Dept. of Architecture, 201 Cowgill Hall, Blacksburg, VA 24061)

Can experienced listeners of music discern a double-sloped decay from a Sabine decay? Do they prefer the double slope? Concert hall designers use coupled-volumes and their signature double-slope sound decay in an effort to reconcile the inversely related qualities of reverberance and clarity. A simulated space, based on an actual built coupled-volume hall, was conceived in the room acoustics software CATT-Acoustic. Variations in the aperture sizes that sonically expose the main hall to the coupled volume generated both classic Sabine decays and double-sloped decays. The impulse responses generated were convolved with the same anechoic musical recording, grouped in pairs, and played for an opportunity-sample of 21 volunteers from the Architectural Acoustics section of the 145th meeting of the Acoustical Society of America in Nashville. Participants listened to the 11 recorded pairs over headphones and were asked to determine (1) if the two recordings sounded different, (2) which recording was more likely to have a double slope or had a more dramatic double slope, and (3) which of the two recordings they prefer.

11:50

2aPP11. Probabilistic neurotransmitter release for dynamic synapse neural networks and its application on pure tone recognition. Alireza A. Dibazar, Hassan H. Namarvar, Sageev George, and Theodore W. Berger (BME Dept., Univ. of Southern California, OHE 500, Los Angeles, CA 90089-1451)

In this paper, we propose a probabilistic neurotransmitter release for dynamic synapses neural network (DSNNs). The capabilities of DSNNs have already been investigated in the processing of spatio-temporal patterns of action potentials [J.-S. Liaw and T. W. Berger (1996)]. The deterministic model of synapse is substituted by probabilistic Markov model. The action potentials generated by auditory system are the inputs of the model. The probability of neurotransmitter release is then estimated from

the model. In general, the aim of this study is to present a robust pure tone recognition system based on DSNN. To generate action potential from music tone we have employed pulse code modulation method. The action potentials are plugged into the DSNN for recognition purpose. Our simulation results showed that the DSNN based on probabilistic release has 4% better recognition performance with respect to the deterministic model [A. A. Dibazar *et al.*, SFN 2002]. [Work supported in part by DARPA-CVS, NASA, and ONR.]

TUESDAY MORNING, 11 NOVEMBER 2003 NUECES ROOM, 9:00 A.M. TO 12:00 NOON

Session 2aSA

Structural Acoustics and Vibration: Sound/Structure Interaction

Jeffrey S. Vipperman, Chair
Department of Mechanical Engineering, University of Pittsburgh, Pittsburgh, Pennsylvania 15261

Contributed Papers

9:00

2aSA1. Computation of scattering by high-aspect ratio spheroids using a new wide-angle on-surface radiation condition approach. David C. Calvo (Naval Res. Lab., Code 7145, 4555 Overlook Ave. SW, Washington, DC 20375)

In a previous study [Calvo *et al.*, IEEE Trans. Antennas Propag. (2003)], a wide-angle on-surface radiation condition (OSRC) was introduced to compute scattering by hard and soft elliptical cylinders. This OSRC was based on a rational approximation of the exact Dirichlet-to-Neumann map for a circular cylinder. As a result of the wide-angle accuracy, exceptional results were obtained for scattering by hard elliptical cylinders when a plane wave was incident end-on with the target. In this talk, the form of the rational approximation is economized and the OSRC is extended to the case of three-dimensional convex bodies (spheroids). A comparison of scattered fields using the OSRC and the BEM methods shows excellent agreement. Owing to the tridiagonal form of the differentiation matrices appearing in the rational approximation, computational costs scale only linearly with frequency making this technique ideal for mid-to-high frequency target scattering computations for orientations where diffraction effects may be important.

9:15

2aSA2. Acoustic diffraction by a disk in a viscous medium. Anthony M. J. Davis (Univ. of Alabama, Tuscaloosa, AL 35487) and Raymond J. Nagem (Boston Univ., Boston, MA 02115)

The diffraction of a time-harmonic acoustic plane wave by a circular disk in a viscous fluid medium is considered by using the linearized equations of viscous fluid flow and the no-slip condition on the rigid disk. Sets of dual integral equations for the fluid velocity and pressure are derived for an arbitrary disk radius and an arbitrary angle of incidence. The dual integral equations are solved by an analytic reduction to sets of linear algebraic equations. In the cases of normal or tangential incidence, numerical results are presented for the fluid velocity in the plane of the disk and the scattered acoustic disturbance in the far field.

9:30

2aSA3. A 3-D hp-adaptive finite-element code for modeling acoustic scattering from elastic structures. David S. Burnett and Mario Zampolli (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19138 La Spezia, Italy, burnett@saclantc.nato.int)

The Centre has developed a 3-D hp-adaptive finite-element structural acoustics code, FESTA, for modeling acoustic scattering from underwater elastic structures. The project is using a unique commercial software system for developing customized application codes. To develop a code, the researcher derives the Galerkin residual equations and then links a self-written code with a vendor-written application-independent library of routines, which creates an application code with a commercial-quality GUI. For FESTA the entire fluid-structure domain is treated as a single continuum, which is modeled using only one type of finite element: a fluid-solid element. The wave equations for both media are derived from the same underlying equations of continuum mechanics, and then combined into a single wave equation, from which the fluid–solid finite element is derived. The code employs 3-D continuum mechanics throughout the computational domain. Thin structural components, such as plates and shells, are modeled with 3-D physics rather than plate or shell theories. The code is therefore fully 3-D, i.e., in both physics and geometry. This paper will describe the theoretical development and present several scattering and propagation models analyzed by FESTA.

9:45

2aSA4. Computing fluid-coupled resonance frequencies, mode-shapes, and damping loss factors using the singular value decomposition. John B. Fahnline (Appl. Res. Lab, Penn State Univ., Appl. Sci. Bldg., University Park, PA 16801, jbf@wt.arl.psu.edu)

In many acoustic design problems, it would be useful to be able to compute fluid-coupled resonance frequencies, mode shapes, and their associated damping levels. Unfortunately, conventional eigenvalue solution procedures are either computationally-inefficient, unreliable, or have limited applicability. Sophisticated methods for identifying modal parameters using the singular value decomposition have recently emerged in the area of experimental modal analysis, where the available data typically consists of velocity to force transfer function data as a function of frequency for several drive point locations. Here, these techniques are shown to be even more effective for coupled finite element/boundary element solutions because full matrices of transfer function data can be computed as a function of frequency. This allows the modes to be completely separated from each other, such that the modal parameters can be identified using simple methods designed for single degree of freedom systems. Several benchmark example problems are solved numerically including a baffled circular plate, an unbaffled rectangular plate, and a spring-mounted piston coupled to fluid within a rigid-walled pipe.

10:00

2aSA5. System identification and energy-based active noise control performance. Benjamin M. Faber and Scott D. Sommerfeldt (Brigham Young Univ., N283 ESC, Provo, UT 84602, bmf3@email.byu.edu)

An active noise control system, which minimizes acoustic energy density, can often become unstable due to significant changes in the acoustical environment in which it operates. Typically, the control system of interest, which is based on a modified version of the filtered-x LMS algorithm, is operated with a fixed set of system identification filter coefficients. These coefficients are determined off-line, prior to execution of the main control

algorithm. The performance of the control system depends on the quality of the system model obtained through the system identification algorithm. Changes in the system transfer function and impulse response, due to various factors, including temperature, boundary conditions, and sensor/actuator location, have been investigated. Results will be presented, which give indications regarding what types of changes to the system have the greatest effect on the transfer function and impulse response, and how sensitive the control system is to those effects. The effects of changing sampling rates and system identification filter lengths on the quality of the system model have also been investigated and will be discussed.

10:15–10:30 Break

10:30

2aSA6. Acoustic control in a tractor cabin using two optimally designed Helmholtz resonators. Patricia L. Driesch (United Technologies Res. Ctr., E. Hartford, CT 06108) and Gary H. Koopmann (Penn State Univ., University Park, PA 16801)

A virtual design methodology is developed to minimize the noise in enclosures with optimally designed, passive, 20 acoustic absorbers (Helmholtz resonators). A series expansion of eigenfunctions is used to represent the acoustic=20 absorbers as external volume velocities, eliminating the need for a solution of large matrix eigenvalue problems. A determination of this type (efficient model/reevaluation approach) significantly increases the design possibilities when optimization techniques are implemented. As a full-scale demonstration, the acoustic response from 90–190 Hz of a tractor cabin was investigated. The lowest cabin mode proposes a significant challenge to a noise control engineer since its anti-node is located near the head of the operator and often generates unacceptable sound-pressure levels. Exploiting the low-frequency capability of Helmholtz resonators, lumped parameter models of these resonators were coupled to the enclosure via an experimentally determined acoustic model of the tractor cabin. The virtual design methodology uses gradient optimization techniques as a post-processor for the modeling and analysis of the unmodified acoustic interior to determine optimal resonator characteristics. Using two optimally designed Helmholtz resonators, potential energy was experimentally reduced by 3.4 and 10.3 dB at 117 and 167 Hz, respectively.

10:45

2aSA7. Micromachined variable impedance acoustic waveguides for fluid-structure waves. Robert D. White and Karl Grosh (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48109, grosh@umich.edu)

Micromachined fluid filled variable impedance waveguides intended to mimic the mechanics of the passive cochlea have been fabricated and experimentally examined. These devices consist of a fluid filled chamber of dimension 37 mm by 6.25 mm by 0.1 mm fabricated on a pyrex wafer. One side of this chamber is constrained by a 3-cm-long, exponentially tapered, 0.3-μm thick tensioned membrane made of low-pressure chemical vapor deposited (LPCVD) silicon nitride. The membranes vary in width from 170 μm to 1.9 mm, and are fabricated using deep reactive ion etching (DRIE). The devices are filled with silicone oil or de-ionized water. Experimental tests demonstrate acoustically excited traveling fluid-structure waves with wave numbers well below the free-plane wave wave number, and with phase accumulations as high as 25 rad (at 6.5 kHz) in the 3 cm length of the device. Comparison of experimental measurements with both numerical (finite-element) and asymptotic (Liouville–Green) models assist in understanding the results. A thin-layer finite-element approximation is developed to take advantage of the device aspect ratio and increase computational efficiency. Viscous fluid effects are included in the models to accurately capture damping phenomena. [Work supported by ONR and NSF.]

11:00

2aSA8. Acoustic signature of an AUV. Joe Cuschieri and Susan Frandsen (Ctr. for Acoust. and Vib., Dept. of Ocean Eng., Florida Atlantic Univ., 101 N. Beach Rd., FL 33004)

In this presentation further results from an experimental analysis of the acoustic signature of an Ocean Explorer class AUV are presented. The results to be presented include, data collected in an open water environment with the AUV operating at cruising speed, measurements performed in a reverberant test tank on an AUV model, and measurements in the reverberant test tank of an AUV under typical operating conditions. Different operating conditions and different mountings for the main propulsion and control module (podule) inside the AUV are considered. Inside the podule are the propulsion motor and the motors for the control surfaces with penetrations for the main propulsion shaft and the shafts for the control surfaces. Considered in the measurements is the influence of the propeller, the mounting of the podule, and the covering the podule with a compliant layer. It is shown that for the type of AUV considered here, the type of mounting of the podule is not very significant and that significant energy is transferred through the water trapped in between the podule and the AUV hull. Furthermore, the propeller has a significant influence on the acoustic signature.

11:15

2aSA9. Correlation of acoustic and vibration signatures of faulty bearings. Nagarjuna Jillella, Corrinne Darvennes, and Sally Pardue (Mech. Eng., Brown Hall, Tennessee Technolog. Univ., Cookeville, TN 38505)

The use of acoustic based measurements for machinery health monitoring of rolling element bearings is explored. Acoustic data was taken with several types of microphones placed very close to the bearings, to minimize the effect of background noise. Accelerometer measurements were also made on the system. A frequency spectrum analysis of acoustic and vibration data was done. The experimentation was performed on a Machine Diagnostics Trainer unit. The shaft is driven by a motor and supported by two bearings that can be easily replaced to incorporate known defects. Three bearings were used to make measurements: a good bearing, a bearing with an inner race defect, and a bearing with an outer race defect. Frequency spectrums and rms values will be presented showing the effects of the transducer type and location, and of the bearing defects.

11:30

2aSA10. Computational and experimental techniques for structural acoustics. Timothy F. Walsh (Dept. of Computational Solid Mech. and Structural Dynam., Sandia Natl. Labs., P.O. Box 5800, M.S. 0835, Albuquerque, NM 87185), Hartono Sumali, and Jeffrey L. Dohner (Sandia Natl. Labs., Albuquerque, NM 87185)

In this paper a discussion of various computational and experimental methods under development at Sandia for evaluating structural acoustic interactions will be presented. The driving applications range from the microscale to the macroscale. An overview will be given of Salinas, a massively parallel structural dynamics code that has been developed at Sandia, and its embedded structural acoustic capabilities. Salinas can perform coupled structural acoustic simulations of the interior and exterior of large-scale structures as well as Micro-Electro-Mechanical Systems (MEMS). An ongoing research effort in Salinas for including viscosity in the acoustic computations will be discussed. Viscosity is typically ignored in commercial acoustics codes. From the experimental side an acoustic wave tube will be described, which is to be used for validating structural acoustic computations and other acoustic experiments of interest. Also, acoustic experimental facilities for MEMS applications will be described. [Sandia is a multiprogram laboratory operated by Sandia Corporation, a Lockheed Martin Company for the United States Department of Energy's National Nuclear Security Administration under Contract No. DE-AC04-94AL85000.]

11:45

2aSA11. Computer methods in vibrational ecology problems. Samuil A. Rybak (N. N. Andreev Acoust. Inst., Moscow 117036, Russia), Sergey A. Makhortykh (Inst. of Mathematical Problems of Biol. RAS, Pushchino Moscow reg. 142290, Russia), Stanislav A. Kostarev (Lab. of Acoust. Vib. Tunnel Assoc., Moscow 107217, Russia), and Aleksandra R. Gatina (Moscow State Univ., Vorobievy Gory, Moscow, Russia)

In the paper formulations of direct vibrational ecology problems are described. Linearly spaced source of acoustic waves in the solid media is considered (e.g., tunnel in soil, simulated by the thin elastic shell). In this case the shell is exited by the inner force changing in time and spatial variables. The proposed computer prognosis method is realized in VibLab software of the Russian Tunneling Association for two versions of algorithms: in homogeneous and stratified two-dimensional media. Three general stratification types are considered: two waveguides (subsurface and waveguide with the axis on the depth H) and stratification with linearly increased wave speed. On the basis of these approximations the amendments, respectively, liquid soil case were calculated. A vibrational field in the ground is described by cylinder waves; modulation of this field type is assessed by a simple final elements scheme. The aspects of application of the proposed algorithms in inverse problems of media parameters determined are also discussed in the report. [Work supported by the Russian Foundation for Basic Research Grants No. 01-02-16127 and No. 02-02-17143.]

TUESDAY MORNING, 11 NOVEMBER 2003

RIO GRANDE EXHIBIT HALL, 9:00 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication: Speech Perception in Normal and Impaired Hearing (Poster Session)

Judith L. Lauter, Chair

Department of Human Services, Stephen F. Austin University, Nacogdoches, Texas 75962

Contributed Papers

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

2aSC1. "The perceptual bases of speaker identity" revisited. William D. Voiers (Dynastat, Inc., 2704 Rio Grande, Austin, TX 78705, bvoiers@aol.com)

A series of experiments begun 40 years ago [W. D. Voiers, J. Acoust. Soc. Am. **36**, 1065–1073 (1964)] was concerned with identifying the perceived voice traits (PVTs) on which human recognition of voices depends. It culminated with the development of a voice taxonomy based on 20 PVTs and a set of highly reliable rating scales for classifying voices with respect to those PVTs. The development of a perceptual voice taxonomy was motivated by the need for a practical method of evaluating speaker recognizability in voice communication systems. The Diagnostic Speaker Recognition Test (DSRT) evaluates the effects of systems on speaker recognizability as reflected in changes in the inter-listener reliability of voice ratings on the 20 PVTs. The DSRT thus provides a qualitative, as well as quantitative, evaluation of the effects of a system on speaker recognizability. A fringe benefit of this project is PVT rating data for a sample of 680 voices. [Work partially supported by USAFRL.]

2aSC2. Children's discrimination of vowel sequences. Jeffry A. Coady, Keith R. Kluender, and Julia Evans (Dept. of Psych., Univ. of Wisconsin–Madison, 1202 W. Johnson St., Madison, WI 53706)

Children's ability to discriminate sequences of steady-state vowels was investigated. Vowels (as in "beet," "bat," "bought," and "boot") were synthesized at durations of 40, 80, 160, 320, 640, and 1280 ms. Four different vowel sequences were created by concatenating different orders of vowels for each duration, separated by 10-ms intervening silence. Thus, sequences differed in vowel order and duration (rate). Sequences were 12 s in duration, with amplitude ramped linearly over the first and last 2 s. Sequence pairs included both same (identical sequences) and different trials (sequences with vowels in different orders). Sequences with vowel of equal duration were presented on individual trials. Children aged 7;0 to 10;6 listened to pairs of sequences (with 100 ms between sequences) and responded whether sequences sounded the same or different. Results indicate that children are best able to discriminate sequences of intermediate-duration vowels, typical of conversational speaking rate. Children were less accurate with both shorter and longer vowels. Results are discussed in terms of auditory processing (shortest vowels) and memory (longest vowels). [Research supported by NIDCD DC-05263, DC-04072, and DC-005650.]

2aSC3. A comparison of speech recognition training programs for cochlear implant users: A simulation study. Marie E. McCabe (Dept. of Psych., Dyer Hall, ML 0376, Univ. of Cincinnati, Cincinnati, OH 45229-0376, mccabeme@email.uc.edu) and C.-Y. Peter Chiu (Univ. of Cincinnati, Cincinnati, OH 45229-0376)

The present simulaton study compared two training programs with very different design features to explore how each might improve the ability of listeners with normal hearing to recognize speech generated by a cochlear implant simulator. The first program, which focused training on specific areas of difficulty for individual patients across multiple levels of linguistic content (e.g., vowels, consonants, words, and sentences), was modeled after a standard program prescribed by one of the US manufacturers of cochlear implants. The second program consisted of exposure to multiple sentences with feedback regardless of subjects' performance level, and had been used in previous studies from this laboratory. All speech materials were reduced spectrally to simulate an 8-channel CIS cochlear implant processor with a "6mm frequency upshift" [Fu and Shannon, J. Acoust. Soc. Am. **105**, 1889 (1999)]. Test sessions were administered to all subjects to assess recognition of sentences, consonants (/aCa/), and vowels (in /hVd/ and /bVt/ contexts) pre- and post-training. In a subset of subjects, a crossover design, in which subjects were trained first with one program and then with the other, was employed. Results will be discussed both in terms of theory and practice of therapeutic programs for cochlear implant users.

2aSC4. Auditory color constancy. Keith R. Kluender (Dept. of Psych., Univ. of Wisconsin, Madison, WI 53706) and Michael Kiefte (School of Human Commun. Disord., Dalhousie, Halifax, NS, Canada)

It is both true and efficient that sensorineural systems respond to change and little else. Perceptual systems do not record absolute level be it loudness, pitch, brightness, or color. This fact has been demonstrated in every sensory domain. For example, the visual system is remarkable at maintaining color constancy over widely varying illumination such as sunlight and varieties of artificial light (incandescent, fluorescent, etc.) for which spectra reflected from objects differ dramatically. Results will be reported for a series of experiments demonstrating how auditory systems similarly compensate for reliable characteristics of spectral shape in acoustic signals. Specifically, listeners' perception of vowel sounds, characterized by both local (e.g., formants) and broad (e.g., tilt) spectral composition, changes radically depending upon reliable spectral composition of precursor signals. These experiments have been conducted using a variety of precursor signals consisting of meaningful and time-reversed vocoded sentences, as well as novel nonspeech precursors consisting of multiple filter poles modulating sinusoidally across a source spectrum with specific local and broad spectral characteristics. Constancy across widely varying spectral compositions shares much in common with visual color constancy. However, auditory spectral constancy appears to be more effective than visual constancy in compensating for local spectral fluctuations. [Work supported by NIDCD DC-04072.]

2aSC5. Processing of speech and non-speech stimuli in children with specific language impairment. Madhavi L. Basu and Aimee M. Surprenant (Purdue Univ., West Lafayette, IN 47907)

Specific Language Impairment (SLI) is a developmental language disorder in which children demonstrate varying degrees of difficulties in acquiring a spoken language. One possible underlying cause is that children with SLI have deficits in processing sounds that are of short duration or when they are presented rapidly. Studies so far have compared their performance on speech and nonspeech sounds of unequal complexity. Hence, it is still unclear whether the deficit is specific to the perception of speech sounds or whether it more generally affects the auditory function. The current study aims to answer this question by comparing the performance of children with SLI on speech and nonspeech sounds synthesized from sine-wave stimuli. The children will be tested using the classic categorical perception paradigm that includes both the identification and discrimination of stimuli along a continuum. If there is a deficit in the performance on both speech and nonspeech tasks, it will show that these children have a deficit in processing complex sounds. Poor performance on only the speech sounds will indicate that the deficit is more related to language. The findings will offer insights into the exact nature of the speech perception deficits in children with SLI. [Work supported by ASHF.]

2aSC6. A novel probabilistic framework for event-based speech recognition. Amit Juneja and Carol Espy-Wilson (Dept. of Elec. and Computer Eng., Univ. of Maryland, AVW Bldg. ECE, College Park, MD 20742, juneja@glue.umd.edu)

One of the reasons for unsatisfactory performance of the state-of-the-art automatic speech recognition (ASR) systems is the inferior acoustic modeling of low-level acoustic-phonetic information in the speech signal. An acoustic-phonetic approach to ASR, on the other hand, explicitly targets linguistic information in the speech signal, but such a system for continuous speech recognition (CSR) is not known to exist. A probabilistic and statistical framework for CSR based on the idea of the representation of speech sounds by bundles of binary valued articulatory phonetic features is proposed. Multiple probabilistic sequences of linguistically motivated landmarks are obtained using binary classifiers of manner phonetic features—*syllabic*, *sonorant* and *continuant*—and the knowledge-based acoustic parameters (APs) that are acoustic correlates of those features. The landmarks are then used for the extraction of knowledge-based APs for source and place phonetic features and their binary classification. Probabilistic landmark sequences are constrained using manner class language models for isolated or connected word recognition. The proposed method could overcome the disadvantages encountered by the early acoustic-phonetic knowledge-based systems that led the ASR community to switch to systems highly dependent on statistical pattern analysis methods and probabilistic language or grammar models.

2aSC7. Effects of lag-time on dichotic word perception in children. Sheryl S. Shoemaker, Edward L. Goshorn, and Shannon Jette (Speech Dept., Louisiana Tech Univ., P.O. Box 3165 Tech Station, Ruston, LA 71272, egoshorn@ltparts.latech.edu)

Dichotic word perception is an essential component in audiological test batteries that are designed to identify auditory processing disorders (APD). Lag-time is the time difference between onsets of dichotic stimuli. The contribution of lag-time to perception of dichotic speech is not fully known and there is a lack of data available on children. Studies by Black (1955) and Berlin *et al.* (1973) found intermittent effects of lag-time on perception of dichotic speech in adults. This project examined the effects of 3 lag-times (0, 150, and 300 ms) on perception of NU-CHCIPS [Elliot and Katz (1980)] words presented dichotically at 50 dB HL through ER-3A insert phones. The sequence of lag-times and ear order was balanced. Subjects were six children age 7 to 10 years who had normal hearing and normal findings on standardized APD tests. Lag-times of 0, 150, and 300 ms yielded mean error rates of 1.3, 1.5, and 1.7 ms, respectively. Mean right ear and left ear error rates were 1.4 and 1.6, respectively. A two-way ANOVA showed no significant ($p>0.05$) effect for lag-time, ear preference, or their interaction. Implications for findings and development of APD treatment materials based on lag-time data will be presented.

2aSC8. Representational specificity of within-category phonetic variation in the mental lexicon. Min Ju and Paul A. Luce (Dept. of Psych., Univ. at Buffalo, Amherst, NY 14260, mju@buffalo.edu)

This study examines (1) whether within-category phonetic variation in voice onset time (VOT) is encoded in long-term memory and has consequences for subsequent word recognition and, if so, (2) whether such effects are greater in words with voiced counterparts (pat/bat) than those without (cow/*gow), given that VOT information is more critical for lexical discrimination in the former. Two long-term repetition priming experiments were conducted using words containing word-initial voiceless stops varying in VOT. Reaction times to a lexical decision were compared between the same and different VOT conditions in words with or without voiced counterparts. If veridical representations of each episode are preserved in memory, variation in VOT should have demonstrable effects on the magnitude of priming. However, if within-category variation is discarded and form-based representations are abstract, the variation in VOT should not mediate priming. The implications of these results for the specificity and abstractness of phonetic representations in long-term memory will be discussed.

2aSC9. Grammatical judgments and phonetic reality: A study of internal constraints in phonological variation. Richard File and Manuel Diaz-Campos (Indiana Univ., Ballantine Hall 844, 1020 E. Kirkwood Ave., Bloomington, IN 47405)

Previous investigations analyzing phonological variation in Spanish have pointed out that phonetic context is an important factor for predicting different variants of /s/. In many studies (Cedergren, 1973; Poplack, 1979, 1980, 1986; Ranson, 1993; Samper Padilla, 1990; Terrell, 1977, 1978,

2a TUE. AM

1979, 1986, etc.), it is claimed that /s/ aspiration is more likely to happen word-internally, preceding a consonant, while elision is more likely to happen word-finally preceding a pause. The methodology used in those studies relies on grammatical judgment for performing the analysis of phonetic context, which could present a perceptual bias that could misrepresent the reality of the realization of the segments. The present investigation examines the phonetic cues that might have an effect in the perception of syllable-final /s/ including the duration of the preceding vowel as well as the following context. Preliminary results show that when the preceding vowel had a shorter duration, retention was easily perceived, whereas when the preceding vowel was longer in duration, the perception of aspiration was favored. The results suggest that for a segment with the same degree of aspiration, deletion was perceived before a pause more often than before a consonant, especially when the preceding vowel was shorter in duration.

2aSC10. Do listeners perceive coarticulatory differences for normal talkers and neurologically impaired talkers? Kris Tjaden, Joan Sussman, and Ya-ju Yu (Dept. of Communicative Disord. & Sci., Univ. at Buffalo, 122 Cary Hall, 3435 Main St., Buffalo, NY 14214-3005, tjaden@acsu.buffalo.edu.)

Speech production studies suggest that the extent of coarticulation varies across talkers. The extent of coarticulation may even vary for repetitions of the same utterance produced by a single talker. Thus, an utterance or talker could be characterized by an average degree of coarticulation, a high degree of coarticulation, or a low degree of coarticulation. The perceptual consequences, if any, of this coarticulatory variability are not well understood, particularly for individuals with speech motor control disorders. The current study examined whether the listeners' speed and accuracy of vowel identification for naturally-produced CV sequences varied depending on the extent of anticipatory vowel coarticulation. Speakers and speech tokens included those characterized by an average degree of anticipatory coarticulation, a high degree of coarticulation, and a low degree of coarticulation. Healthy talkers, speakers with multiple sclerosis, and speakers with Parkinson's disease produced the stimuli. Consonants in CV syllables beginning with /s/, /k/, or /t/ followed by the vowels /i/ or /u/ were excised and presented to listeners for identification of the following vowel. While identification accuracy may be high across speakers and stimuli, the speed of response should vary with extent of coarticulation, if listeners are sensitive to coarticulatory cues in the acoustic speech stream.

2aSC11. Effects of frequency shifts and visual gender information on vowel category judgments. Catherine Glidden and Peter F. Assmann (School of Behavioral and Brain Sci., Univ. of Texas–Dallas, Box 830688, Richardson, TX 75083)

Visual morphing techniques were used together with a high-quality vocoder to study the audiovisual contribution of talker gender to the identification of frequency-shifted vowels. A nine-step continuum ranging from "bit" to "bet" was constructed from natural recorded syllables spoken by an adult female talker. Upward and downward frequency shifts in spectral envelope (scale factors of 0.85 and 1.0) were applied in combination with shifts in fundamental frequency, $F0$ (scale factors of 0.5 and 1.0). Downward frequency shifts generally resulted in malelike voices whereas upward shifts were perceived as femalelike. Two separate nine-step visual continua from "bit" to "bet" were also constructed, one from a male face and the other a female face, each producing the end-point words. Each step along the two visual continua was paired with the corresponding step on the acoustic continuum, creating natural audiovisual utterances. Category boundary shifts were found for both acoustic cues ($F0$ and formant frequency shifts) and visual cues (visual gender). The visual gender effect was larger when acoustic and visual information were matched appropriately. These results suggest that visual information provided by the speech signal plays an important supplemental role in talker normalization.

2aSC12. Part Ia: Spatial separation on McGurk effect applying three-dimensional sounds. Klaus A. J. Riederer (Lab. of Acoust. and Signal Processing, Helsinki Univ. of Technol., Otakaari 5 A, P.O. Box 3000, FIN-02015 TKK, Finland, klaus.riederer@hut.fi)

The dependence of *sound direction* on the *McGurk effect* [McGurk and McDonald, Nature (London) **264**, 746–748 (1976)] is less known. Jones and Munhall [Canadian Acoust. **25**, 13–19 (1997)] concluded with no spatial separation dependence, applying 30° horizontally spaced loudspeakers. Current *dual* study investigated the full 360° horizontal space applying *head-related transfer functions* (HRTFs) from a Cortex dummy head [Riederer, J. Audio Eng. Soc. (Abstracts) **46**, 1036 (1998), preprint 4846]. Dry acoustic /ipi/ and /iti/ recorded from a professional speaker were convolved with HRTFs, measured at azimuths $0°$, $\pm 40°$, $\pm 90°$, $\pm 130°$, and $180°$, headphones (Sennheiser HD580) equalized. DVcam-recorded visual /ipi/, /iti/ (and black screen) were randomly presented synchronously with the 3-D sounds using Presentation 0.20 [http://nbs.neuro-bs.com]. Totally 1024 incongruent audiovisual stimuli were perceived by eight 20–30-year-old normal hearing ($\leqslant 20$ dBHL) native subjects (2 female) as follows. Visual /ipi/ + auditory /iti/: /ipi/ 59.96%, /iti/ 15.63%, and /ipti/ 24.02%; visual /iti/ + auditory /ipi/: 66.02%, 22.07%, and 11.52%, respectfully. No significant dependence of spatial separation was found for the McGurk effect, except for *reaction times*. The obtained fusions were atypically weak, probably because visual /iti/ was less pronounced than visual /ipi/. [Work supported by Graduate School of Electronics, Telecommunication and Automation.]

2aSC13. Shifts in the perceived voicing boundary of bilingual listeners. Adrian Garcia-Sierra, Craig A Champlin (Univ. of Texas–Austin, 1 University Station A1100, Austin, TX 78712, gasa@austin.utexas.edu), and Maritza Rivera-Gaxiola (Univ. of Washington, Seattle, WA 98105)

Three groups of listeners, American-English monolinguals, Mexican-Spanish monolinguals, and Mexican-Spanish bilinguals, identified synthetic /ga/–/ka/ syllables varying in VOT. In order to induce a language set in the bilingual listeners, they were asked to produce syllables and phrases in either English or Spanish prior to the identification task. Bilinguals identification of the VOT stimuli varied depending on the language set condition, with the voicing boundary in each case shifting in the direction of values characteristic of monolingual listeners. This effect of language set replicates earlier findings [L. Elman, R. Diehl, and S. E. Buchwald, J. Acoust. Soc. Am. **62**, 971–974 (1977)]. Bilinguals velar consonant productions showed a corresponding shift in VOT values across the two language conditions. A Mismatch Negativity paradigm will be used in an attempt to record preattentive discrimination of the language-set-induced phonemic boundary shift in bilingual listeners. [Work supported by the Department of Communication Sciences and Disorders Univ. of Texas–Austin.]

2aSC14. Shifting perceptions of age in voice. Rahul Shrivastav, Harry Hollien, W. S. Brown, Jr., Howard B. Rothman (Inst. for Adv. Study of the Commun. Proc., Univ. of Florida, Gainesville, FL 32611), and James D. Harnsberger (Univ. of Florida, Gainesville, FL 32611)

A series of experiments have been carried out in order to identify the acoustical and perceptual correlates of the aging voice. The initial phase of the program was to identify those voice parameters which signal a person's age; the second phase was to systematically shift these parameters in

order to determine if a parallel change in perceived age would occur. This second study focused on temporal characteristics related to voice. In this instance, standard speech samples for 16 males aged 70–90 years were contrasted with those of 14 males aged 20–33 years. The features studied included the following: (1) sentence duration, (2) word duration, (3) diphthong duration, (4) consonant-vowel ratios, (5) number of pauses and (6) pause duration. Significant differences were found for all relationships. Subsequently, a preliminary study was carried out where the voices were synthesized and the temporal parameters for the two groups shifted toward each other. The preliminary data suggest that such modifications lead especially to the idea that the voices of older individuals actually were those of younger men.

2aSC15. Structural equation modeling for estimating the identification accuracy and detection time latency of English monosyllabic words. Sumiko Takayanagi, Lynne E. Bernstein, and Edward T. Auer, Jr. (Dept. of Commun. Neurosci., House Ear Inst., Los Angeles, CA 90057, stakayanagi@mailhouse.hei.org)

Structural equation modeling (SEM) was used to examine the statistical structure among sets of experiential (word age of acquisition and subjective familiarity) and lexical similarity (lexical equivalence class size and neighborhood density) variables for word identification and reaction time latency tasks. Stimuli were 240 vocoded monosyllabic English words with reduced intelligibility and altered similarity relationships. Participants detected a target word following a prime and on every trial reported the prime. The identification accuracy was estimated by words and phonemes correct, and detection latency was estimated by trimmed and harmonic mean RTs. A parsimonious SEM was chosen in terms of the chi-square and model fit indices that determine whether the models adequately described the particular associations of variables/interfactor relationships. The variable/factor error variances were constrained to be uncorrelated with each other in order to evaluate effects independently. A bootstrapping technique indicated that the regression weights of the top-down and bottom-up factors were small, but they were significant in the model. The variance accounted for (VAF) by the model was 7.1% for identification accuracy, and 5.2% for RT latency. The model also indicated that RT latency was highly influenced by prime identification accuracy (15% VAF). [Work supported by NIH/NIDCD00695.]

2aSC16. The effects of auditory and visual vowel training on speech reading performance. Carolyn Richie and Diane Kewley-Port (Dept. of Speech and Hearing Sci., Indiana Univ., IN)

Speech reading, the use of visual cues to understand speech, may provide a substantial benefit for normal-hearing listeners in noisy environments and for hearing-impaired listeners in everyday communication. However, there exists great individual variability in speech reading ability, and studies have shown that only a modest improvement in speech reading ability is achieved with training. The purpose of this investigation was to determine the effects of a novel approach to speech reading training on word and sentence identification tasks. In contrast to previous research, which involved training on consonant recognition, this study focused on vowels. Two groups of normal-hearing adults participated in auditory–visual (AV) conditions with added background noise. The first group of listeners received training on the recognition of 14 English vowels in isolated words, while the second group of listeners received no training. All listeners performed speech reading pre- and post-tests, on words and sentences. Results are discussed in terms of differences between groups, dependent upon whether training was administered, and a comparison is made between this and other speech reading training methods. Finally, the potential benefit of this vowel-based speech reading training method for the rehabilitation for hearing-impaired listeners is discussed. [Work supported by NIHDCD-02229.]

2aSC17. Two-C but not two-V: Segment similarity in learning an artificial lexicon. Sarah C. Creel, Richard N. Aslin, and Michael K. Tanenhaus (Dept. of Brain and Cognit. Sci., Univ. of Rochester, Rochester, NY 14627, screel@bcs.rochester.edu)

The role of segment similarity (C1_C2_or _V1_V2) in a word learning task was assessed using an artificial lexicon in a referential context. Learning consisted of 480 trials in which S's heard one of 40 CVCV nonsense strings, accompanied by an unfamiliar picture. In testing, participants heard the direction "Click on the [nonsense word]," and chose one of four pictures that matched the test item. On some trials, target lexical items (pibo) appeared with foils that contained matched consonants (pabu) or matched vowels (diko). There were higher rates of confusion errors to the matched-consonant items than to non-matched items, but no significant elevation in errors to matched-vowel items. A second experiment examined the role of differences in informativeness between C's and V's by inverting the numbers of C and V types (first experiment: 10 C, 5 V; second experiment: 5 C, 10 V). This made the consonants less predictive of word identity (more words contained the same consonants), and made the vowels more predictive of word identity. The matched-consonant effect remained undiminished while no corresponding matched-vowel effect emerged, ruling out a segment-informativeness explanation. Other accounts based on the syllable positions and confusability patterns of consonants are being explored.

2aSC18. An investigation of perceptual tolerance limits of stop constriction regions along the vocal tract. Kang Li and Brad Story (Dept. of Speech and Hearing Sci., Univ. of Arizona, P.O. Box 210071, 1131 E. 2nd St., Tucson, AZ 85721-0071, lik@u.arizona.edu)

In this study we explore the perceptual categories of constriction regions along the vocal tract for voiced stop consonants /b/, /d/, and /g/. The three consonants were imbedded in consonant-vowel (CV) syllables. Using an area function model, a series of stop-vowel syllables that differed only in the place of constriction at even spaces was constructed. A second series was generated by eliminating the third formant of each syllable in the first series. It was predicted that the continuum of place of constriction would result in the discontinuity of acoustic representation and the categorical perception. The predicted perception of each synthesized syllable was based on their spectrographic shapes. The results showed that the perception of the constriction continuum was categorical in both series. The boundaries of actually perceived regions agree reasonably well with the predicted ones. It turned out that the perception of the midway stimuli between two categories was ambiguous instead of being perceived as a member of one category or the other. The Distinctive Region Model based on the sensitivity function is discussed to explore a possible explanation for the ambiguity.

2aSC19. Otoacoustic emissions measured in children diagnosed with attention-deficit/hyperactivity disorder. Dennis McFadden, J. Gregory Westhafer, Edward G. Pasanen, David M. Tucker, and Caryn L. Carlson (Dept. of Psych., Univ. of Texas, Austin, TX 78712-0187, mcfadden@psy.utexas.edu)

Attention-deficit/hyperactivity disorder (ADHD) is generally acknowledged to be more prevalent in males than in females. Further, some precursors to ADHD appear early in life. Together these facts suggest that ADHD may be influenced by androgenic mechanisms operating early in development. This reasoning raises the question of whether the otoacoustic emissions (OAEs) of children with ADHD are masculinized. Click-evoked OAEs were measured for one click level in 8 boys and 3 girls diagnosed as ADHD/Combined, in 11 males and 5 females diagnosed as ADHD/Inattentive (IA), and in 17 male and 18 female controls. The ages of these samples ranged between 7 and 15. As in adults, the CEOAEs of the control males were weaker than those of the control females. Further, the CEOAEs of the ADHD/IA males were weaker than in the control males (a hypermasculinization) and the CEOAEs of the IA females were weaker than in the female controls (a masculinization). The CEOAEs of the Combined groups were slightly stronger (feminized) than those of the control males and females. One interpretation is that the IA subgroup of ADHD boys and girls (but not the Combined subgroup) was exposed to higher-than-normal levels of androgens sometime early in development. [Work supported by NIDCD.]

2aSC20. The time course of indexical specificity effects in the perception of spoken words. Conor T. McLennan and Paul A. Luce (Dept. of Psych. and Ctr. for Cognit. Sci., Univ. at Buffalo, 245 Park Hall, Buffalo, NY 14260, mclennan@buffalo.edu)

This research investigates the time-course of indexical specificity effects in spoken word recognition by examining the circumstances under which the variability in the speaking rate affects the participant's perception of spoken words. Previous research has demonstrated that variability has both representational and processing consequences. The current research examines one of the conditions expected to influence the extent to which indexical variability plays a role in spoken word recognition, namely the time-course of processing. Based on our past work, it was hypothesized that indexical specificity effects associated with the speaking rate would only affect later stages of processing in spoken word recognition. The results confirm this hypothesis: Specificity effects are only in evidence when processing is relatively slow. [Research supported (in part) by Research Grant No. R01 DC 0265801 from the National Institute on Deafness and Other Communication Disorders, National Institutes of Health.]

2aSC21. AXS and SOM: A new statistical approach for treating within-subject, time-varying, multivariate data collected using the AXS Test Battery. Judith L. Lauter and Chris Ninness (Dept. of Human Services, Stephen F. Austin State Univ., Box 13019 SFA Station, Nacogdoches, TX 75962, jlauter@sfasu.edu)

The Auditory Cross-Section (AXS) Test Battery [J. L. Lauter, Behav. Res. Methods Instrum. Comput. **32**, 180–190 (2000)], described in presentations to ASA in 2002 and 2003, is designed to document dynamic relations linking the cortex, brainstem, and body periphery (whether physics, physiology, or behavior) on an individually-specific basis. Data collections using the battery typically employ a within-subject, time-varying, multivariate design, yet conventional group statistics do not provide satisfactory means of treating such data. We have recently developed an approach based on Kohonens (2001) Self-Organizing Maps (SOM) algorithm, which categorizes time-varying profiles across variables, either within- or between-subjects. The treatment entails three steps: (1) z-score transformation of all raw data; (2) employing the SOM to sort the time-varying profiles into groups; and (3) deriving an estimate of the bounds for the Bayes error rate. Our three-step procedure will be briefly described and illustrated with data from a recent study combining otoacoustic emissions, auditory brainstem responses, and cortical qEEG.

2aSC22. Some effects of intonation contour on sentence intelligibility. James M. Hillenbrand (Speech Pathol. and Audiol., Western Michigan Univ., Kalamazoo, MI 49008)

This experiment was designed to measure the effects of pitch movement on sentence intelligibility. A source-filter synthesizer was used to generate three synthetic versions of 60 sentences drawn from the TIMIT multi-talker speech database: (1) an original pitch (OP) condition in which the fundamental frequency ($F0$) contour matched that of the original utterance, (2) a monotone pitch (MP) condition in which $F0$ was held constant at the median value measured from the original utterance, and (3) an inverted pitch (IP) condition in which the $F0$ contour was reflected around the median $F0$ value (i.e., pitch rises were changed to pitch drops, and vice versa). Results from 30 listeners showed a small but statistically reliable drop in intelligibility from the OP condition to either the MP or IP condition, with no difference between the MP and IP conditions. A second group of 22 listeners was tested on the same task, but with overall sentence intelligibility reduced by running all signals through a 2-kHz low-pass filter. As with the unfiltered signals, intelligibility was reduced for the MP and IP conditions relative to OP; however, the decrements in intelligibility were somewhat larger for the filtered signals, and inverting pitch caused a larger intelligibility decrement than flattening pitch.

2aSC23. Perceptual rate normalization in naturally produced bilabial stops. Kyoko Nagao and Kenneth de Jong (Dept. of Linguist., Indiana Univ.–Bloomington, 406 Memorial Hall, Bloomington, IN 47405, knagao@indiana.edu)

The perception of voicing categories is affected by the speaking rate, so that listeners' category boundaries on a VOT continuum shift to a lower value when the syllable duration decreases (Miller and Volaitis, 1989; Volaitis and Miller, 1992). Previous rate normalization effects have been found using computer-generated stimuli. This study examines the effect of speech rate on voicing categorization in naturally produced speech. Four native speakers of American English repeated syllables (/bi/ and /pi/) at increasing rates in time with a metronome. Three-syllable stimuli were spliced from the repetitive speech. These stimuli contained natural decreases in VOT with faster speech rates. Besides, this rate effect on VOT was larger for /p/ than /b/, so that VOT values for /b/ and /p/ overlapped at the fastest rates. Eighteen native listeners of American English were presented with 168 stimuli and asked to identify the consonant. Perceptual category boundaries occur at VOT values 15 ms shorter than the values reported for synthesized stimuli. This difference may be due to the extraordinarily wide range of VOT values in previous studies. The values found in the current study closely match the actual division point for /b/ and /p/. The underlying mechanism of perceptual normalization will be discussed.

2aSC24. The Ganong paradigm: Converging evidence supporting initial phoneme weighting. Erik C. Tracy and Mark A. Pitt (Dept. of Psych., The Ohio State Univ., 1885 Neil Ave. Mall, Columbus, OH 43210)

In the present experiment we investigate whether the initial phoneme is given more weight in word recognition [W. D. Marslen-Wilson and A. Welsh, Cognit. Psych. **10**, 29–63 (1978)] or if all phonemes in a word are weighted equally [C. M. Connine, D. G. Blasko, and D. Titone, J. Mem. Lang. **32**, 193–210 (1993)]. Using the Ganong paradigm [W. F. Ganong, JEP:HPP. **6**, 110–125 (1980)], participants were instructed to categorize a final ambiguous fricative in the target items, which included both words and pseudowords. Pseudowords were created by changing either the initial or a medial phoneme within the words. For example, the word diminish was altered to create the pseudowords timinish and dimimish. In addition, initial and medial phonemes were altered by either one or three distinctive features. The differences in the labeling of the final ambiguous fricative in the target items led to the conclusion that the initial phoneme is weighted more heavily. [Work supported by NIDCD.]

Session 2aSP

Signal Processing in Acoustics and Noise: Signal Processing for Aircraft Noise

Joe W. Posey, Chair
NASA Langley Research Center, Mail Stop 461, Hampton, Virginia 23681

Chair's Introduction—8:00

Invited Papers

8:05

2aSP1. Beamforming for aircraft noise measurements. Robert P. Dougherty (OptiNav, Inc., 10914 NE 18 St., Bellevue, WA 98004, rpd@optinav.com)

Phased array beamforming for aircraft noise source location has a long history, including early work on jet noise, wind tunnel measurements, and flyover testing. In the last 10 years, advancements in sparse 2-D and 3-D arrays, wind tunnel test techniques, and computer power have made phased array measurements almost common. Large aerospace companies and national research institutes have an advantage in access to major facilities and hundreds of measurement microphones, but universities and even consulting companies can perform tests with electret microphones and PC data acquisition systems. The type of testing remains a blend of science and art. A complex noise source is approximated by a mathematical model, and the microphones are deployed to evaluate the parameters of the model. For example, the simplest, but often the best, approach is to assume a distribution of mutually incoherent monopoles. This leads to an imaging process analogous to photography. Other models include coherent distributions of multipoles or duct modes. It is sometimes important to simulate the results that would have been obtained from single microphone measurements of part of the airplane in an ideal environment, had such measurements been feasible.

8:50

2aSP2. The rotating rake fan mode measurement system. Daniel Sutliff (NASA Glenn Res. Ctr., MS 54-3, 21000 Brookpark Rd., Cleveland, OH 44212)

An experimental measurement system was developed and implemented by the NASA Glenn Research Center in the 1990s. The system is a continuously rotating radial rake immersed into the duct. This rotating rake provides a complete map of the acoustic duct modes present in a ducted fan and has been used on a variety of test articles: from a low-speed, concept test rig to a full-scale production turbofan engine. The rotating rake has been critical in developing and evaluating a number of noise reduction concepts as well as providing experimental databases for verification of several aero-acoustic codes. This paper will describe the physical theory (Sofrin) and the analytical techniques (Moore) upon which the rotating rake is based will be described. Data processing and analysis as well as implementation issues will be discussed. Several Rotating Rake systems have been custom built for 3 facilities. In order of complexity of the turbo machinery test article, these are (1) the Advanced Noise Control Fan, (2) various 22-inch fan rigs in the NASA Glenn 9×15 wind tunnel, and (3) a full scale turbofan, the Honeywell TFE-731-60. Descriptions and measurement achievements of these systems will be provided (Heidelberg, Sutliff).

9:20

2aSP3. Novel error sensing microphone arrays for active control of turbofan rotor/stator tones. Bruce E. Walker, Alan S. Hersh (Hersh Walker Acoust., 780 Lakefield Rd., Unit G, Westlake Village, CA 91361), Edward J. Rice (E. J. Rice Consulting, Westlake, OH), and Daniel L. Sutliff (NASA Glenn Res. Ctr., Cleveland, OH)

Active control of turbofan rotor/stator interaction tones is complicated by the simultaneous presence of multiple duct propagation modes. In-duct error sensing microphone arrays that can adequately resolve these modes typically require duct lengths that are incompatible with modern compact engine design. Two alternative approaches have been investigated. For inlet noise, an external linear array of microphones was positioned in the near/far radiation field transition region and weighted to provide error signals resolved either by duct mode or by radiation angle. For the exhaust, radially spaced microphones have been placed on duct bifurcation panels to provide supplemental radial-mode resolution. The concepts were tested in combination with an adaptive segmented liner in a static duct and as part of an active stator-vane system in the ANCF research facility at NASA/Glenn Research Center. [Work sponsored by NASA/Langley Research Center.]

9:50

2aSP4. Methodologies for duct liner impedance eduction. Tony L. Parrott, Michael G. Jones (Structural Acoust. Branch, MS 463, 2 N. Dryden St., NASA Langley Res. Ctr., Hampton, VA 23681-2199, t.l.parrott@larc.nasa.gov), Willie R. Watson (NASA Langley Res. Ctr., Hampton, VA 23681-2199), and Charles D. Smith (Lockheed Martin Eng. and Sci. Co., NASA Langely Res. Ctr., Hampton, VA 23681-2199)

Methodologies and techniques are reviewed that are currently employed at Langley Research Center to educe (from primitive measurements) the impedance of acoustically absorbing liner structures. These structures are of interest for suppressing noise emission from aircraft engine nacelles. The accuracy and precision of the primitive measurements and their impact on the educed impedance of liners when exposed to high speed grazing flows is of special interest for aircraft engine nacelle applications. The test setups range from the classical standing wave tube for which the primitive measurement is a complex transfer function between two judiciously chosen locations, to an elaborate grazing flow duct arrangement (the Langley Grazing Incidence Tube) for which the primitive measurements (acoustic pressure and phase) are compromised by increased flow noise contamination due to high speed grazing flows up to a Mach number of 0.5. Results of different techniques/methodologies are compared on the basis of how the primitive measurements are processed and mapped into impedance spectra for different test setups. The results are compared on the basis of bias and precision errors that are specific to the impedance eduction methodology employed.

10:20–10:35 Break

10:35

2aSP5. *In situ* evaluation of aircraft interior noise reduction technologies. Jacob Klos and Daniel L. Palumbo (Structural Acoust. Branch, M.S. 463, NASA Langley Res. Ctr., Hampton, VA 23681, j.klos@larc.nasa.gov)

In order to quantify the performance of interior noise treatments under flight conditions, it is desirable to evaluate the noise reduction due to treatment of a limited portion of an aircraft fuselage. However, radiation from the untreated areas of the fuselage can corrupt an intensity measurement in front of the treated area. In the past, this problem of corrupting noise has been solved by acoustically isolating the treated area from the rest of the fuselage. In this presentation, a method to evaluate the performance of an acoustic treatment applied to an aircraft fuselage *in situ* using correlation analysis is documented. The insertion loss of the acoustic treatments is estimated from the ratio of the intensity, correlated to reference transducers, measured with and without the treatment applied. The formulation is presented for both single and multiple reference transducers. Several experimental studies and numerical simulations have been conducted, and the results are documented. Through these case studies, it is demonstrated that this method can be used to evaluate the insertion loss of fuselage treatments without having to acoustically isolate the treated area.

10:55

2aSP6. Signal processing for aircraft flyover noise synthesis and propagation. Stephen A. Rizzi, Brenda M. Sullivan (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA 23681, Stephen.A.Rizzi@nasa.gov), and Bryan A. Cook (AuSIM, Inc., Mountain View, CA 94043)

Subjective assessments of low noise aircraft flight operations require time histories of acoustic pressure at listener positions. Synthesized sound has an advantage over recordings by allowing the examination of proposed aircraft, flight procedures, and other conditions or configurations for which recordings are unavailable. A two-stage process for synthesizing flyover noise at listener positions on the ground is presented. The first stage entails synthesizing time histories at the flying source. Rizzi and Sullivan [J. Acoust. Soc. Am. **113**, 2245 (2003)] developed an approach for synthesizing sound from broadband sources (e.g., jet noise) based upon predicted 1/3-octave band source spectra. Further developments in source synthesis signal processing are presented here, including the introduction of temporal fluctuations and the synthesis of tone-dominated sources (e.g., fan noise). The second stage entails propagation of the synthesized sound from the flying source to the listener. Its signal processing aspects are also presented and include a dynamic absolute delay of sound and location (producing accurate Doppler shift and emission position), spreading loss, atmospheric attenuation, and binaural filtering. Whereas the first stage prediction analysis and synthesis are computed *a priori*, the second stage is performed in real-time, allowing creation of an immersive test environment.

Contributed Papers

11:15

2aSP7. Near-field acoustical holography in enclosed spaces. Courtney Burroughs (Appl. Res. Lab., Penn State Univ., State College, PA 16804)

In enclosed spaces, the locations of surfaces from which most of the noise radiation occurs can be difficult to determine in the presence of reverberation and contributions from other sources. By using microphones configured in an array that confirms to the surface of interest, it should be possible to map the acoustic field inside the enclosed space produced by the surface under the array using nearfield acoustical holography processing on the unsteady pressures measured by the microphones in the conformal array. Such an acoustic mapping could therefore be used to isolate contributions to the interior noise make by different surfaces. Because enclosed spaces rarely have simple geometries, it is often necessary to measure the propagation function (i.e., the Green's function) for the enclosed space. To explore the sensitivity of mappings to measurement and physical parameters, a mathematical model of the acoustic field inside a capped cylindrical shell was developed. Measurements at locations on a conformal array are simulated, maps of the interior acoustic fields developed by applying numerical nearfield acoustical holography algorithms on the simulated measurements and resulting acoustic mappings compared to acoustic fields predicted directly from known surface vibration distributions.

11:30

2aSP8. Supersonic naval missile sounds over San Nicolas Island. Charles R. Greene, Jr., Robert G. Norman (Greeneridge Sci., Inc., 4512 Via Huerto, Santa Barbara, CA 93110), Meike Holst (LGL Ltd. Environ. Res. Assoc., King City, ON L7B 1A6, Canada), and Charles I. Malme (Eng. & Sci. Services, Hingham, MA 02043)

Vandals and other missiles are launched occasionally from San Nicolas Island, CA, during Naval exercises and tests. Pinnipeds on the island beaches are exposed to the flight sounds, some of which are sonic booms from directly overhead. Environmental concerns led the Navy to support acoustic studies of the missile sounds at the beaches. The results show flat-weighted sound pressures from Vandals as high as 150 dB *re*: 20 μPa(peak) [140 dB *re*: 20 μPa(rms)] at a near-vertical distance of 400 m. Other flat-weighted pressures from Vandals were as low as 107 dB *re*: 20 μPa(peak) [95 dB *re*: 20 μPa(rms)] at a beach 3.9 km horizontally behind the launcher. Pulse durations and sound exposure levels were also measured. One-third octave band sound exposure levels were measured. All parameters (except one-third octave band levels) were also measured with *A* weighting. Other missiles measured include Tomahawk cruise missiles, Rolling Airframe Missile, Advanced Gun System, Terrier, and the Supersonic Sea-Skimming Target. [Work supported by U.S. Navy.]

11:45

2aSP9. Multiresolution analysis of noise generated by aircrafts during landing/take-off operations. Luigi Maxmilian Caligiuri and Adolfo Sabato (Univ. of Calabria, via P. Bucci, 87036 Rende (CS), Italy)

The evaluation of noise impact generated by aircrafts is usually realized by means of indicators such as Lden and/or by indexes obtained opportunely combining the values of single event levels relative to the events occurred within the measurement time interval. In any case is fundamental, to ensure the validity of result, to properly find and characterize the noise events produced by aircrafts landing/take-off operations, in particular distinguish them from that produced by other sources. Since the traditional analysis techniques are generally inadequate to describe these nonstationary signals, its necessary to employ time-frequency or time-scale based techniques. In this paper it will be shown the application of multiresolution time-scale and time-frequency analysis techniques, based on Wavelet (WT) and smoothed Wigner Ville transforms (SWVT), to noise signals related to aircrafts landing/take-off operations, also comparing the result with those obtained using a Windowed Fast Fourier Transform (WFFT). The effects of changing the parameters related to the smoothing WVT windows and to the WT base functions atoms will be investigated, showing that an adequate choose of such parameters will allow to us to properly analyze and characterize these noise events in order to quantify their impact.

TUESDAY AFTERNOON, 11 NOVEMBER 2003 BRAZOS AUDITORIUM, 1:00 TO 5:00 P.M.

Session 2pAA

Architectural Acoustics and Musical Acoustics: Electroacoustic Enhancement System in Rooms for Music

Yasushi Shimizu, Chair
200-41 Timitsuka, Hamamatsu 432-8002, Japan

Chair's Introduction—1:00

Invited Papers

1:10

2pAA1. Electronic architecture: An acoustician's approach. J. Christopher Jaffe (Jaffe Holden Acoust., 114A Washington St., Norwalk, CT)

Electronic architecture is a process for modifying the sound fields of an auditorium, concert hall, or theatre to recreate the sound fields of outstanding examples of individual performance spaces. This paper discuss how acousticians develop proper acoustic environments for specific program use in physical acoustic halls and how these design goals are incorporated into electronically enhanced facilities.

1:35

2pAA2. Control of early and late energy in rooms with the variable room acoustics system. Mark Poletti (Industrial Res. Ltd., P.O. Box 31-310, Lower Hutt, New Zealand, m.poletti@irl.cri.nz) and Steven Ellison (Level Control Systems, Sierra Madre, CA 91024)

Many electroacoustic systems seek to offer control of subjective sound quality using either regeneration of sound or by minimizing feedback to produce in-line sound control. In-line systems provide control of early energy from the stage but do not respond globally to all sound sources. Regenerative systems offer global enhancement of late energy but are less able to influence early energy. The Variable Room Acoustics System is a hybrid approach which contains a regenerative system for reverberation enhancement and an in-line system for early energy control. Both systems are optimized to provide maximum power gain. The principles of the system are outlined and results from recent installations are presented.

2:00

2pAA3. Active field control (AFC) -electro-acoustic enhancement system using acoustical feedback control. Hideo Miyazaki, Takayuki Watanabe, Shinji Kishinaga, and Fukushi Kawakami (Adv. System Development Ctr., Yamaha Corp., 10-1 Nakazawa-cho Hamamatsu, Japan)

AFC is an electro-acoustic enhancement system using FIR filters to optimize auditory impressions, such as liveness, loudness, and spaciousness. This system has been under development at Yamaha Corporation for more than 15 years and has been installed in approximately 50 venues in Japan to date. AFC utilizes feedback control techniques for recreation of reverberation from the physical reverberation of the room. In order to prevent coloration problems caused by a closed loop condition, two types of time-varying control techniques are implemented in the AFC system to ensure smooth loop gain and a sufficient margin in frequency characteristics to prevent instability. Those are: (a) EMR (electric microphone rotator) -smoothing frequency responses between microphones and speakers by changing the combinations of inputs and outputs periodically; (b) fluctuating-FIR -smoothing frequency responses of FIR filters and preventing coloration problems caused by fixed FIR filters, by moving each FIR tap periodically on time axis with a different phase and time period. In this paper, these techniques are summarized. A block diagram of AFC using new equipment named AFC1, which has been developed at Yamaha Corporation and released recently in the US, is also presented.

2:25

2pAA4. Various applications of Active Field Control (AFC). Takayuki Watanabe, Hideo Miyazaki, Shinji Kishinaga, and Fukushi Kawakami (Adv. System Development Ctr., Yamaha Corp., 10-1 Nakazawa-cho, Hamamatsu, Japan, watanabe@yarl.yamaha.co.jp)

AFC is an electro-acoustic enhancement system, which has been under development at Yamaha Corporation. In this paper, several types of various AFC applications are discussed, while referring to representative projects for each application in Japan. (1) Realization of acoustics in a huge hall to classical music program, e.g., Tokyo International Forum. This venue is a multipurpose hall with approximately 5000 seats. AFC achieves loudness and reverberance equivalent to those of a hall with 2500 seats or fewer. (2) Optimization of acoustics for a variety of programs, e.g., Arkas Sasebo. AFC is used to create the optimum acoustics for each program, such as reverberance for classical concerts, acoustical support for opera singers, uniformity throughout the hall from the stage to under-balcony area, etc. (3) Control of room shape acoustical effect, e.g., Osaka Central Public Hall: In this renovation project, preservation of historically important architecture in the original form is required. AFC is installed to vary only the acoustical environment without architectural changes. (4) Assistance with crowd enthusiasm for sports entertainment, e.g., Tokyo Metropolitan Gymnasium. In this venue, which is designed as a very absorptive space for speech intelligibility, AFC is installed to enhance the atmosphere of live sports entertainment.

2:50

2pAA5. Active acoustics for music rehearsal rooms. Ronald R. Freiheit (555 Park Dr., Owatonna, MN 55060, ron.freiheit@wengercorp.com)

The use of virtual acoustics has the ability to provide a new level of rehearsal experience for the musician. By integrating the signal processing of an active acoustic system (with time variant-gain before feedback) into a relatively small rehearsal room, musicians can now benefit from the experience of rehearsing in multiple acoustic environments including those of the actual performance venue in which they will perform. To effectively communicate the various acoustics environments, the musicians must be immersed in the sound field of the active acoustics without being able to discern source locations of the speakers. The system must also be cable of supporting the dynamic range of the musicians without presenting artifacts of its own such as system noise or audible distortion. An installation of such a system will be provided as a case study describing the challenges that were overcome for a successful implementation including areas such as adequate sound isolation, background noise levels and system security. The paper will also briefly discuss programming methodologies for the system. Anecdotal responses from musicians who have used the active acoustic rehearsal room and some unexpected issues will also be covered.

3:15–3:30 Break

3:30

2pAA6. Acoustical enhancement systems: Design criteria and evaluation of room acoustical parameters based on *in situ* measurements. Bjorn van Munster and Wim Prinssen (Systems for Improved Acoust. Performance, Runmolen 3, 5404 KP Uden, The Netherlands, b.v.munster@pbri.nl)

Acoustic enhancement systems have evolved significantly during the years. Where the early systems only aimed to increase the reverberation time in a hall, nowadays the increase of the reverberation time is only one of the features of such a system. Contrary and additionally to passive acoustics, an acoustic enhancement system enables a designer or acoustical consultant to change the acoustical characteristics of a hall in a more flexible way. Due to the sophisticated convolution processes and layout of such a system besides the reverberation time also, e.g., speech intelligibility and spaciousness can be improved or special effects can be added to shows. In this paper the applications of an enhancement system in general will be outlined in more detail. Furthermore, design criteria will be given which can be formulated for the installation of such a system. These criteria can be used to evaluate proposed designs, but also to estimate the required provisions to be included in the planning of a system installation. Besides, the paper describes the results of *in situ* measurements of one such system (SIAP) whereby the increase of the reverberation time is evaluated with respect to certain important room acoustical parameters, i.e., reverberant level, lateral efficiency and clarity.

3:55

2pAA7. An electronic enhancement system for Silva Concert Hall. Timothy E. Gulsrud (Kirkegaard Assoc., 954 Pearl St., Boulder, CO 80302, tgulsrud@kirkegaard.com), Scott D. Pfeiffer, and Frans H. H. Swarte (Kirkegaard Assoc., Chicago, IL 60607)

An electronic enhancement system and new orchestra shell have been designed for and installed in Silva Concert Hall, a 2500-seat multipurpose hall at the Hult Center for the Performing Arts in Eugene, Oregon. The enhancement system provided by Acoustic Control Systems B.V. features early reflection and reverberation modules in the house and a foldback system for performers on stage. The design considerations, interaction with the new shell, tuning, and in-hall measurements of the system are discussed, including reactions by musicians, administrators, and critical listeners.

Contributed Paper

4:20

2pAA8. Experimental evaluation of the omnidirectional behavior of platonic polyhedron loudspeakers. Sarah Rollins, Timothy Leishman, and Gordon Dix (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT 84602)

Many architectural acoustics measurements require the use of an omnidirectional source. For several years, the source predominantly used for such applications has been the dodecahedron loudspeaker with small in-phase drivers mounted in each face. While other platonic polyhedron loudspeakers (PPLs) have not been used as frequently, they also produce nearly omnidirectional fields over limited bandwidths. Above cutoff frequencies specific to their geometries, all PPLs depart from ideal omnidirectional behavior, with varying degrees of directivity. While these cutoff frequencies are typically higher for higher-order polyhedra, they commonly fall within the bandwidths of standard measurements. The five types of PPLs have been constructed and measured to gain greater insight into their omnidirectional behaviors. Their frequency responses were taken at 2664 points over a sphere (5-deg polar and azimuthal angle increments) in an anechoic chamber. The measurements were then processed to produce directivity balloon plots. However, to better compare directivities and find the source consistently producing the most omnidirectional field over a useful bandwidth, a frequency-dependent standard deviation formula was implemented. Average values of the standard deviation parameter produce figures of merit that further characterize omnidirectionality. [Research supported by funding from the NSF REU program.]

4:35–5:00

Panel Discussion

TUESDAY AFTERNOON, 11 NOVEMBER 2003 PECOS ROOM, 1:00 TO 5:00 P.M.

Session 2pAO

Acoustical Oceanography: Geoacoustic Inversion

Gopu R. Potty, Chair

Department of Ocean Engineering, University of Rhode Island, Narragansett, Rhode Island 02881

Contributed Papers

1:00

2pAO1. Geoacoustic inversion of broadband data in the Florida Straits. N. Ross Chapman and Yongmin Jiang (School of Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3055, Victoria, BC V8W 3P6, Canada)

Acoustic propagation experiments have been carried out in the Florida Straits with a multi-frequency broadband source that transmitted m-sequence pulses over a range of 10 km to a sparse-filled vertical line array. This paper presents results of matched field inversions of the acoustic field data at low frequencies to estimate geoacoustic model parameters for the experimental site. Two approaches were taken for the inversions. The first was a conventional matched field inversion using multi-frequency data centered at 200 and at 400 Hz from the vertical array. The second approach was designed to model the low frequency waveform at a single hydrophone. For the very long range experimental geometry, the waveform was modeled in terms of modes. Each inversion was cast as an optimization problem using the adaptive simplex simulated annealing algorithm. The inversions provide a comparison between approaches that take advantage of the spatial coherence in one case, and the time coherence in the received signal in the other case. Both inversions give similar results for the parameters of a simple geoacoustic bottom model, and the sensitivities and relative uncertainties of the model parameters are consistent for the two approaches. Notably, the inversions are sensitive to compressional wave attenuation.

1:15

2pAO2. Attenuation inversions in the East China Sea. Gopu Potty and James Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882)

Data from the ASIAEX East China Sea (ECS) experiment is used to estimate sediment attenuation coefficients as a function of frequency and depth. The ECS experiment offers a number of independent measures of sediment parameters using seismic and chirp surveys, gravity and piston cores and historic data. Modal amplitude ratios for modes 1 to 3 are used to obtain the modal attenuation coefficient and attenuation profiles. These inversions use broad-band data from Wide Band Sources (WBS) in the frequency range 20–100 Hz. The modes are detected, identified and their amplitudes measured using a time–frequency wavelet analysis. A joint inversion for modal attenuation coefficients, water depth, source and receiver depths, range and Empirical Orthogonal Functions in the water column is performed. The results will be compared with core data and frequency and depth dependence will be examined. [Work supported by ONR.]

1:30

2pAO3. Geoacoustic inversion results of low- to high-frequency source tow data from the ASIAEX East China experiment. Chen-Fen Huang and William S. Hodgkiss (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0238)

During the 2001 ASIAEX East China Sea experiment, source tow data was collected by a 16-element, 75-m aperture, autonomously recording vertical line array in approximately 105-m-deep water. Transmissions from two similar 6-km-long tracks were transmitted. In the first, CW tonals at 95, 195, 295, 395, 805, 850, and 905 Hz were transmitted from a J-15 transducer at a nominal depth of 46 m. In the second, CW tonals at 1.6, 2.4, 3.5, and 4.4 kHz were transmitted from an ITC-2015 transducer at a nominal depth of 49 m. An environmental model in the study area initially was derived by matched-field geoacoustic inversion using the low-frequency track. In this study, the high-frequency data from the second track are included in the parameter assessment. A multiple-stage geoacoustic inversion scheme is applied. Inversion results for seafloor geoacoustic parameters from these transmissions will be presented. [Work supported by ONR.]

1:45

2pAO4. Acoustic inversions from an explosive source of opportunity in the ASIAEX SCS experiment. Ying-Tsong Lin (Natl. Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 106, Taiwan), James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Nicholas P. Chotiros (Office of Naval Res., Arlington, VA 22217), Altan Turgut (Naval Res. Lab., Washington, DC), Ching-Sang Chiu (Naval Postgrad. School, Monterey, CA), and Steven G. Schock (Boca Raton, FL)

An acoustic inversion for bottom geoacoustic properties from an unexpected explosion in the South China Sea (SCS) is presented. Horizontal beamforming is performed to find the source azimuthal direction and the water wave part of the modal dispersion curve is used to find the source range. The localization result shows that the detonation occurred in the direction of east–north east and about 40 km distant from the receiver. To perform the bottom inverse, we use a broadband linear inverse technique employing the modal group velocity, which is inferred from the modal arrival time and source range. In the end, a four-layered bottom model is created, which is consistent with both our explosion data and independent source tow data. The resolution and variance of the estimate are also presented. In conclusion, this bottom model should be widely applicable for modeling broadband sound propagation in the ASIAEX SCS experiment site.

2:00

2pAO5. Integration versus optimization in self-noise geoacoustic inversion. David J. Battle, Peter Gerstoft, William A. Kuperman, William S. Hodgkiss (Marine Physical Lab., Univ. of California–San Diego, La Jolla, CA 92093-0238, davidb@mpl.ucsd.edu), and Martin Siderius (SAIC, La Jolla, CA 92037)

Self-noise geoacoustic inversion involves the estimation of bottom parameters such as sound speeds and densities by analyzing towed-array signals whose origin is the tow platform itself. As well as feeding into more detailed assessments of seabed geology, these parameters enable performance predictions for sonar systems operating in shallow water. In this presentation, the use of the Gibbs sampler to obtain complete probability distributions of seabed parameters is discussed. This contrasts with conventional maximum likelihood inversion, in which only the best fitting model is identified. Advantages of viewing parameter estimation problems from such a probabilistic perspective are discussed in relation to the MAPEX 2000 self-noise data, which have previously been subjected to maximum-likelihood inversion via genetic algorithm optimization.

2:15

2pAO6. Sub-bottom profiling by inverting ambient noise. Chris H. Harrison (SACLANT Undersea Res. Ctr., Viale San Bartolomeo 400, 19138 La Spezia, Italy, harrison@saclantc.nato.int)

In this paper a technique based on spectral factorization for restoring the phase of incoherent bottom sediment reflection coefficient measurements is presented, so that by Fourier transformation one can then obtain the minimum phase impulse response at each grazing angle. The method is developed and discussed in the context of another recently established technique for extracting the seabed's plane wave reflection coefficient from ambient noise data measured on a moored or drifting vertical array (VLA). Limitations of the phase restoration method are discussed, and using modeled data, comparisons are made between the "true" impulse response derived from the known complex reflection coefficient and the result of applying spectral factorization to the absolute value of the reflection coefficient. For instance, in both cases one can see clear, matching arrivals from each layer boundary at angles above critical. Finally the method is demonstrated on experimental reflection loss inferred from ambient noise measurements at three moored VLA sites and one VLA drift track in the Mediterranean Sea. Convincing angular variation of the impulse response is shown for the moored sites. Sub-bottom profiles (impulse response vs position) are shown for the drift track demonstrating that one can survey with just a single directional receiver.

2:30

2pAO7. Numerical modeling of air-to-sea transmission of light aircraft noise. Eric M. Giddens and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., UCSD, 8820 Shellback Way, La Jolla, CA 92093-0238, egiddens@ucsd.edu)

Recent experiments at SIO have shown that the acoustic signature of a light aircraft can be detected by sensors in the water column as well as buried in the underlying sediment and a method for extracting the sound speed and attenuation from this Doppler shifted signal has been proposed. To test the accuracy of this geoacoustic inversion technique, a numerical model of the air-water-sediment acoustic propagation, including the effects of a high-speed airborne source, has been developed based on the spectral method. Simulated acoustic data have been generated representing an aircraft flying over a microphone in the atmosphere, a vertical line array in the ocean, and a hydrophone buried 1-m deep in the sediment. The results of the geoacoustic inversion for sound speed and attenuation are compared to the known input parameter values of the model, giving a sense of the relative errors that may be expected when applying the technique to experimental data. [Work supported by ONR.]

2:45

2pAO8. The effects of ignored seabed variability in geo-acoustic inversion. Anna-Liesa S. Lapinski and David M. F. Chapman (Defence R&D Canada Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada)

In recent years, acoustic inversion techniques have been developed to predict ocean environmental properties, source position, etc. However, when applying such techniques to measured acoustic data, assumptions are invariably made with respect to the parametrization of the ocean environment due to unavailable information or to simplify the problem. In this work, the influences of range-dependent variability of the seafloor in the results of inversion are explored. That is, a typical inversion algorithm is applied while making the assumption that all parameters of the ocean-bottom (e.g., compressional speed, water depth, density, attenuation) are constant with range when in fact one parameter varies realistically with range. The inversion algorithm is applied to many unique realizations of a synthetic ocean-bottom; however, for each realization, the nonfluctuating parameters remain constant and the statistics regarding the fluctuating parameter such as the mean parameter value and the standard deviation of the fluctuations about the mean remain the same. It is shown that ignoring the variation in even a single seabed parameter leads to significant and correlated uncertainty in all inverted parameter values.

3:00–3:15 Break

3:15

2pAO9. Are buried river channels sources of geoclutter on the New Jersey Continental Margin? John C. Osler (DRDC Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, john.osler@drdc-rddc.gc.ca)

Geological features on a continental shelf may be responsible for anomalous acoustic scatter that are identified as (false) targets, or GeoClutter, on active sonar systems. Features on the New Jersey Continental Margin include a drainage system that formed when sea-level was much lower, ran across the shelf, and incised channels approximately 10 meters deep into the surrounding seabed. These channels have since been filled with sediments that are not apparent on bathymetric maps. The potential for these channels to create GeoClutter depends in part on the contrast in geoacoustic properties between the sediments filling the channels and the adjacent flanks. To study this matter, an experiment was conducted to measure the reflection loss from 1 to 10 kHz of channel fill and flank sediments in an area where GeoClutter has been observed and where there is supporting geophysical data. The measurements were made using the WARBLE technique [C. W. Holland and J. C. Osler, J. Acoust. Soc. Am. **107**, 1263–1279 (2000)], adapted for use in rapid environmental assessment using modified sonobuoys. Results from the experiment will be presented and the role of buried channels acting as sources of GeoClutter on the New Jersey Continental Margin will be discussed.

3:30

2pAO10. Propagation and attenuation of sound in a canonical model of shallow water with a thermocline and with a Biot sediment in the bottom. Allan D. Pierce, William C. Carey (Boston Univ., Boston, MA 02215, adp@bu.edu), and James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Objective is benchmark solution, analytic in internal detail, for testing geoacoustic inversion methods. The water column has two isovelocity layers separated by a narrow thermocline. The bottom is a porous (Biot model) sediment, with an elastic matrix coexisting with a fluid filling the pores; coefficients are frequency independent, slowly varying with depth, and analogous to those in works by Stoll, Badiey, and Chotiros. Source and receiver are in the middle region of the water column. Constant frequency problem formulation has field expressed as a Fourier integral over horizontal coordinates, with a z-dependent kernel having discontinuous slope at source depth. Equations for the bottom region yield ratio of derivative to kernel amplitude at bottom interface, this being a complex function of wave number and frequency. Singularities in kernel at discrete complex values of k correspond to the natural modes, imaginary parts of k are attenuation constants. Simplifying approximations, especially those of perturbation theory, are guided by order of magnitude estimates of dimensionless groups, with the suppositions that the orders of magnitude of propagation frequency, water depth, and propagation distance are 100 Hz, 100 m, and from 2 to 100 km, and that Biot model parameter magnitudes are comparable to known measured values.

3:45

2pAO11. Investgation of sediment properties with a rotated coordinates inversion technique. Tracianne Neilsen, Marcia Isakson, and Andrew Worley (Appl. Res. Labs., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78758)

Several models have been developed that describe the interaction of high-frequency sound with ocean sediments. This work presents how estimates of the seabed properties can be obtained with a simulated annealing inversion algorithm that minimizes the difference between measured and modeled reflection loss within the statistical nature of the data and the statistical nature of the parameters. Because there is often a correlation between how the various seabed properties influence the reflection loss of acoustic waves, the efficiency and accuracy of the inversion can be improved by using a rotated set of coordinates to navigate the search space in the inversion. The rotated coordinates are obtained by performing an eigenvalue decomposition of the covariance matrix of the gradients of the cost function. The resulting eigenvectors are the rotated coordinates, show the couplings between the seabed parameters, and are used to vary the parameters in the inversion. The corresponding eigenvalues indicate the relative sensitivities of the cost function to changes in the parameters and provide insights into how the inversion problem can be effectively decoupled [T. B. Neilsen, "An iterative implementation of rotated coordinates for inverse problems," J. Acoust. Soc. Am. **113**, 2574–2586 (2003)]. [Work supported by ONR.]

4:00

2pAO12. A new inversion method (SUB-RIGS) based on range-sensitive frequency-subspacing and reduced iterated grid searches. A. Tolstoy (ATolstoy Sci., 8610 Battailles Court, Annandale, VA 22003)

Propagation for a range-independent, multi-layered test case (Na or Workshop '97) indicates that the highly variable bottom sound-speed profile can be well represented by only a few, e.g., three, linear layers (one layer at the highest frequency, three at the lowest). Thus, at the low frequencies of interest here (less than 500 Hz) transmission loss as a function of range can usually be well predicted using only a few simple sediment layers. This suggested a new inversion method based on range-sensitive frequency-subspacing plus reduced iterated grid spacing (SUB-RIGS). The method is now fully developed and uses the RAMGEO PE propagation model. This paper will discuss the method, pitfalls of the method, and newly successful inversion results for selected test cases (TC1 and CAL from the recent Stennis workshop of May 2001).

4:15

2pAO13. Measurement uncertainty of seabed reflectivity. Charles Holland (Appl. Res. Lab., The Penn State Univ., State College, PA 16804)

The seabed reflection coefficient is a fundamental property of the ocean waveguide. Measurements of the frequency and angular dependence of the reflection coefficient can provide information about the geoacoustic properties of the seabed or can be used as an input to propagation models. The uncertainty of the measurements must be known in order to determine the prediction uncertainties for the acoustic field and/or the geoacoustic properties. An analysis indicates that reflection measurements [Holland and Osler, J. Acoust. Soc. Am. **107**, 1263–1279 (2000)] have a standard deviation from 0.5–1 dB at full angular resolution depending on frequency and experiment geometry. The dominant contribution to the error is source amplitude variability and a new processing approach was developed that reduces the error for frequencies above a few hundred Hz. A further reduction in the uncertainty can be obtained by averaging in angle, for example, a 1 angle averaging leads to a standard deviation of 0.1–0.5 dB. Errors in the angle estimate are a few tenths of a degree from 0–34° grazing angle: the crucial angular range for predicting long-range propagation or for geoacoustic property inversion. [Research sponsored by ONR.]

4:30

2pAO14. Optimally resolving Lambertian surface orientation. Ioannis Bertsatos and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

Sonar images of remote surfaces are typically corrupted by signal-dependent noise known as speckle. Relative motion between source, surface, and receiver causes the received field to fluctuate over time with circular complex Gaussian random (CCGR) statistics. In many cases of practical importance, Lambert's law is appropriate to model radiant intensity from the surface. In a previous paper, maximum likelihood estimators (MLE) for Lambertian surface orientation have been derived based on CCGR measurements [N. C. Makris, SACLANT Conference Proceedings Series CP-45, 1997, pp. 339–346]. A Lambertian surface needs to be observed from more than one illumination direction for its orientation to be properly constrained. It is found, however, that MLE performance varies significantly with illumination direction due to the inherently nonlinear nature of this problem. It is shown that a large number of samples is often required to optimally resolve surface orientation using the optimality criteria of the MLE derived in Naftali and Makris [J. Acoust. Soc. Am. **110**, 1917–1930 (2001)].

4:45

2pAO15. Geocorrection and filtering of 3D bottom images from multi-beam sonar records. Jerzy Demkowicz, Krzysztof Bikonis, Andrzej Stepnowski, and Marek Moszynski (Gdansk Univ. of Technol., Narutowicza 11/12, 80-952 Gdansk, Poland)

For the last decade multibeam sonars have been increasingly used for mapping and visualization of the seafloor to provide the "physical bases" for environmental studies. Increasing amount of digital (raster) echo records of high resolution from a multibeam sonar have enhanced the potential of computer modeling of the marine environment to improve our understanding of the bottom processes. However, the 3D bottom images as the result of merging different sonar transects do not comply exact geographical positions and should be corrected. Additionally, the raw sonar records are subject to systematic errors, random noise and outliers. In this paper, Kalman filtering techinque to generating optimal estimates of bottom surface from a noisy raw sonar records is proposed. The experiment on the surface indicates that after applying the Kalman filtering the outliers of raw records can be efficiently removed. Moreover, the two-step Kalman filtering method enables 3D seabed visualization in real time. The paper proposes the geographical corrections applied to the merged mutibeam sonar transects records. The 3D bottom relief before, and after the filtering method are presented.

TUESDAY AFTERNOON, 11 NOVEMBER 2003 TRINITY A ROOM, 1:00 TO 3:30 P.M.

Session 2pBB

Biomedical Ultrasound/Bioresponse to Vibration: HIFU and Scattering

Ibrahim M. Hallaj, Chair

Wolf, Greenfield and Sacks, PC, Federal Reserve Plaza, 600 Atlantic Avenue, Boston, Massachusetts 02210

Contributed Papers

1:00

2pBB1. Synchronization of HIFU therapy system with an arbitrary ultrasound imager. Neil Owen, Michael Bailey, James Hossack, and Lawrence Crum (Ctr. for Industrial and Medical Ultrasound, 1013 NE 40th St., Seattle, WA 98105)

Synchronization for image guided therapy using high intensity focused ultrasound (HIFU) and imaging ultrasound is achieved with a new technique that uses the focused transducer as a receiver that can detect the acoustic pulses created by the imaging probe. Without synchronization, interference from the high intensity source occludes the imager's display unpredictably, degrading the quality of the system. An imaging probe (Sonosite 180) is registered with a HIFU transducer ($d=33$ mm, roc $=55$ mm, $f=3.5$ MHz) such that the scan line bisects the single element focus. When acoustically coupled through a scattering medium, imaging pulses are passively detected with the HIFU transducer and electronically conditioned into a TTL level trigger. A LabVIEW program uses the trigger to create a pulse width modulated signal that controls the timing of HIFU excitation during treatment. Detection takes less than 1% of the time between displayed images when the imager is running at 20 frames per second. HIFU excitations are programmed to occur such that the single element focus is free of interference when viewed with the imager during treatment. With no electrical connections for this new, simple technique, an arbitrary imager can be selected for synchronized image guided therapy. [Work supported by NSBRI.]

1:15

2pBB2. Rapid continuous-wave pressure field calculations for spherically focused radiators. Robert McGough (Dept. of Elec. and Computer Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48824, mcgough@egr.msu.edu)

A new accelerated expression for the continuous-wave pressure field generated by a spherically focused radiator is obtained when the impulse response formulation is transformed and optimized for numerical evaluations. The resulting integral expression converges much more quickly than the impulse response approach, resulting in far fewer function evaluations for the same numerical error. The optimized integral expression is between two and seven times as fast as the impulse response approach, where the increase in speed depends on the peak value of the specified error. In addition, this new result completely eliminates the cone-shaped regions required for impulse response calculations, so the resulting computer code for the accelerated expression is less complicated than the corresponding code for the impulse response. Results also show that the new expression eliminates the numerical artifact that is encountered near the boundary between regions defined for impulse response calculations. All of these features are useful in thermal therapy computer simulations that employ spherically focused transducer geometries.

1:30

2pBB3. Design and evaluation of a 63 element 1.75-dimensional ultrasound phased array for treating benign prostatic hyperplasia. Khaldon Y. Saleh and Nadine B. Smith (Dept. of Bioengineering, 205 Hallowell Bldg., The Penn State Univ., University Park, PA 16802)

Focused ultrasound surgery (FUS) is a clinical method for treating benign prostatic hyperplasia (BPH) in which tissue is noninvasively necrosed by elevating the temperature at the focal point above 60 °C using short sonications. With 1.75-dimensional (1.75-D) arrays, the power and phase to the individual elements can be controlled electronically for focusing and steering. This research describes the design, construction and evaluation of a 1.75-D ultrasound phased array to be used in the treatment of benign prostatic hyperplasia. The array was designed with a steering angle of ± 13.5 deg in the transverse direction, and can move the focus in three parallel planes in the longitudinal direction with a relatively large focus size. A piezoelectric ceramic (PZT-8) was used as the material of the transducer and two matching layers were built for maximum acoustic power transmission to tissue. To verify the capability of the transducer for focusing and steering, exposimetry was performed and the results correlated well with the calculated fields. *In vivo* experiments were performed to verify the capability of the transducer to ablate tissue using short sonications. [Work supported by the Whitaker Foundation and the Department of Defense Congressionally Directed Medical Prostate Cancer Research Program.]

1:45

2pBB4. Optimized hyperthermia treatment of prostate cancer using a novel intravavitary ultrasound array. Osama M. Al-Bataineh, Nadine B. Smith (Dept. of Bioengineering, The Penn State Univ., University Park, PA 16802), Robert M. Keolian, Victor W. Sparrow (The Penn State Univ., University Park, PA 16802), and Lewis E. Harpster (Penn State Milton S. Hershey Medical Ctr., Hershey, PA 17033)

Localized uniformly distributed ultrasound-induced hyperthermia is a useful adjuvant to radiotherapy in the treatment of prostate cancer. A two-dimensional, 20$\times$4 element, transrectal phased-array probe was designed to deliver a uniform and controllable amount of heat directly to the prostate without damaging the rectal wall or surrounding tissue. A three-dimensional prostate model was created using anatomical markers from the Visible Human Project to optimize the array. Sound speed, density, and absorption parameters were mapped to hue, saturation and value of the photographic data to simulate sound propagation through inhomogeneous tissue using the k-space method. To satisfy the requirements of this method from 1.2 to 1.8 MHz, the grid was adjusted to have 5 points per millimeter in each Cartesian direction. A spherical wave pulse was propagated through the model using tapered absorption boundary conditions. The expected temperature rise due to sound was obtained using the bioheat transfer equation. Optimal insonification parameters that uniformly heat the prostate to 43 °C for 40–60 minutes were determined for use in the construction of a clinical hyperthermia array. [Research supported by the Department of Defense Congressionally Directed Medical Prostate Cancer Research Program.]

2:00

2pBB5. Separating thermal coagulation and cavitation effects in HIFU attenuation measurements. Justin Reed, Michael Bailey, Ajay Anand, and Peter Kaczkowski (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105-6698)

HIFU can be used to destroy tumors. The conversion of acoustic energy into heat causes protein coagulation (Lesion) in tissue. Attenuation measurements have been proposed to monitor the progression of thermal therapy. The goal of this work is to study and separate the effects of cavitation and thermal coagulation in attenuation measurements. A HIFU transducer was used to treat Bovine liver. A receiving transducer mounted across from the transmitting HIFU transducer measured attenuation during the treatment. A pressure chamber provided static pressure greater than the pressure amplitude of the HIFU wave, which suppressed cavitation. rf data from a commercial ultrasound scanner was also obtained. A large increase in attenuation was observed with cavitation present, while a subtle increase in attenuation was observed with cavitation suppressed. Attenuation estimated from the RF data showed an increase in attenuation downstream of the location of the lesion with cavitation present, while a subtle increase in attenuation was observed at the location of the lesion with cavitation suppressed. It has been found that attenuation measurements are greatly affected by the presence of cavitation, and the actual effect of thermal coagulation on attenuation is quite small. [Work supported by NIH, NSF, NSBRI.]

2:15

2pBB6. Numerical investigation of dual-frequency HIFU pulsing for lithotripsy. Wayne Kreider, Michael Bailey, and Lawrence Crum (Ctr. for Industrial and Medical Ultrasound, APL, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@u.washington.edu)

As an alternative to traditional shock-wave lithotripsy, high-intensity focused ultrasound (HIFU) is currently being investigated for its capability to comminute renal calculi. Because current data indicate that cavitation plays a role in both stone comminution as well as collateral tissue damage, the cavitation effects of HIFU treatment strategies are investigated numerically. In particular, numerical simulations are designed to model the response of bubbles to acoustic excitations generated by a prototype, dual-frequency HIFU transducer for lithotripsy. The prototype transducer is capable of producing both high- ($\sim$4-MHz) and low-frequency ($\sim$100-kHz) outputs, while the bubble dynamics are modeled by the Gilmore equation for a single spherical bubble subject to diffusion. Numerical simulations are currently ongoing to investigate the effects of the relative phase between high and low-frequency pulses. Initial results demonstrate that the simultaneous application of high and low-frequency pulses can generate maximum pressures several orders of magnitude higher than high-frequency pulses alone.

2:30

2pBB7. The characterization of the lesion growth in time. Marie Nakazawa, Justin A. Reed, Michael R. Bailey, and Yongmin Kim (Dept. of Elec. Eng., Univ. of Washington, 1400 NE Campus Pkwy., Seattle, WA, nakazawa@ns.cradle.titech.ac.jp)

Thermal heating effects of high intensity focused ultrasound (HIFU) on the dynamics of lesion formation were characterized automatically to assess the role of vapor bubbles in distorting the shape. Tissue mimicking phantom was used in experiments by a 4.2 MHz curve-linear transducer with 44 mm diameter and 44 mm radius of curvature. A variety of HIFU intensities were produced by different amplitudes. Images were acquired by a CCD camera and HDI-1000 ultrasound imager, recorded to VHS, and digitized to measure lesion size and shape. Each image was subtracted with noise reduction in order to detect the HIFU on time and to segment the boundaries of the lesions performed by Matlab programming. Area, length, width, and ratio of lesion area proximal to center line over area distal to center line were calculated along HIFU exposure time. Slight increase in HIFU intensity, means hyperecho forms earlier, and lesion shape change. The data supported the hypothesis that lesion dramatically distorts well after hyperecho with only small increase in HIFU intensity. [Work supported by National Space and Biomedical Research Institute.]

2:45

2pBB8. Optimization of angular compounding in scatterer size estimation. Anthony L. Gerig, Quan Chen, and James A. Zagzebski (Dept. of Medical Phys., Univ. of Wisconsin–Madison, 1300 Univ. Ave., Rm. 1530, Madison, WI 53706, algerig@wisc.edu)

Ultrasonic scatterer size estimates generally have large variances due to the inherent noise of the spectral estimates used to calculate size. Compounding partially correlated size estimates associated with the same tissue, but produced with data acquired from different angles of incidence, is an effective way to reduce the variance without making dramatic sacrifices in spatial resolution. This work derives theoretical approximations for the correlation between these size estimates, and between their associated spectral estimates, as functions of data acquisition and processing parameters, where a Gaussian spatial autocorrelation function is assumed to adequately model scatterer shape. Size results exhibit a fair degree of agreement with those of simulation experiments, while spectral results compare favorably with simulation outcomes. Utilization of the theoretical correlation expressions for data acquisition and processing optimization is discussed. Further simplifying approximations, such as the invariance of phase and amplitude terms with rotation angle, are made in order to obtain closed-form solutions to the derived spectral correlation, and permit an analytical optimization analysis. Results indicate that recommended parameter adjustments for performance improvement depend upon whether, for the system under consideration, the primary source of estimate decorrelation with rotation is scatterer phase change or field separation. [Work supported by NIH T32CA09206.]

2p TUE. PM

3:00

2pBB9. Factors affecting scatterer size estimation using a generalized ultrasound attenuation–compensation function to correct for focusing. Timothy A. Bigelow and William D. O'Brien, Jr. (Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801, bigelow@uiuc.edu)

Over the years many different investigators have attempted to estimate the size of the principle scattering sites in materials by hypothesizing a model for the scattering and then fitting the power spectrum of the backscattered ultrasound signal to the model to obtain an estimate for the scatterer size. Traditionally, these models have assumed that a plane wave is incident on the scattering site limiting the measurement technique to unfocused or weakly focused ultrasound sources. In this investigation, the plane wave assumption was replaced by a model that assumed that the field of a focused source could be modeled as a three-dimensional Gaussian about the focus. Then, strongly focused sources could be used to obtain estimates for the size of the principle scattering sites provided that a generalized attenuation–compensation function was used to correct for the focusing. The improvement provided by the generalized attenuation–compensation function over traditional attenuation–compensation functions that neglected focusing was strongly affected by the type of scatterer being estimated. Also, the wavelength dependence of the axial beamwidth impacted how the improvement provided by the generalized attenuation–compensation function varied with increased focusing (i.e., going from f/1 to f/4). [Work supported by NDSEG Graduate Fellowship and Beckman Institute Graduate Fellowship.]

3:15

2pBB10. Characterization and differentiation of two mammary tumors using parametric imaging with ultrasound. Michael L. Oelze, William D. O'Brien, Jr. (Dept. of Elec. and Computer Eng., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801, oelze@uiuc.edu), and James F. Zachary (Univ. of Illinois, Urbana, IL 61801)

Two kinds of solid tumors were acquired and scanned *in vivo* ultrasonically. The first tumor series (fibroadenoma) was acquired from tumors that developed spontaneously in rats. The second tumor series was acquired by culturing a carcinoma cell line (4T1-MMT) and injecting the cells into Balb/c mice. The scatterer properties (average scatterer diameter and acoustic concentration) were estimated using a Gaussian form factor from the backscattered ultrasound measured from both kinds of tumors. Parametric images of tumors were constructed utilizing estimated scatterer properties for regions of interest inside the tumors and surrounding normal tissues. The average scatterer diameter and acoustic concentration for the fibroadenomas were estimated at 107 ± 14 micrometers and 15.2 ± 5 dB (mm^{-3}), respectively. The average scatterer diameter and acoustic concentration for the carcinomas was estimated at 30 ± 4.6 micrometers and 10.3 ± 6.9 dB (mm^{-3}), respectively. A comparison with light microscopic evaluations of the fibroadenomas showed cellular structures around 100 micrometers in size, and carcinomas showed cell nuclei with an average size of 12.5 micrometers in diameter (the total cellular size ranging from 50% to 200% larger than the nucleus size). [Work supported by NIH F32 CA96419 to MLO and by the University of Illinois Research Board.]

TUESDAY AFTERNOON, 11 NOVEMBER 2003 SAN ANTONIO ROOM, 1:00 TO 2:30 P.M.

Session 2pMUa

Musical Acoustics and Physical Acoustics: Wind Instruments and Nonlinearity II

Peter L. Hoekje, Chair
Department of Physics and Astronomy, Baldwin-Wallace College, Berea, Ohio 44017

Contributed Papers

1:00

2pMUa1. Modes of vibration of air-driven free reeds in transient and steady state oscillation. Ammon Paquette (Augustana College, Rock Island, IL 61201, ammon-paquette@augustana.edu), Justin Vines (Univ. of Arkansas, Fayetteville, AR 72701), and James P. Cottingham (Coe College, Cedar Rapids, IA 52402)

Most treatments of free reed oscillation approximate the reed vibration as a sinusoidal oscillation of a cantilever beam in the fundamental transverse mode, although some evidence of the presence of the second transverse mode has been reported. [Cottingham *et al.*, J. Acoust. Soc. Am. **105**, 940 (1999)]. Some new measurements of the oscillation of a free reed from an American reed organ mounted on a laboratory wind chest show that the second beam mode is present even at low amplitudes of oscillation, and is often observable in the transient period before the oscillation reaches full amplitude. Some evidence of higher frequency modes has also been observed. In addition to steady state oscillation, reed motion during two types of attack transients has been studied. In one case, with full playing pressure in the wind chest and air flowing through the reed, the reed is restrained in its unblown equilibrium position and suddenly released. In another configuration, the reed is provided with a pallet valve mechanism, and reed oscillation is initiated by a sudden rush of air when the valve is opened. [Work supported by the NSF from REU Grant No. 0139096.]

1:15

2pMUa2. Vibrational modes of the reed in a reed organ pipe. T. M. Huber, B. A. Collins (Dept. of Phys., Gustavus Adolphus College, 800 College Ave., St. Peter, MN 56082, huber@gustavus.edu), M. Pineda (Polytec PI, Inc., Tustin, CA 92780), and C. Hendrickson (Hendrickson Organ Co., St. Peter, MN 56082)

We will describe a series of measurements of the vibrational modes of the reed in a reed organ pipe. These measurements were performed using a Polytec PSV-300 scanning vibrometer, which allows the vibrational deflection shape to be determined at any frequency. In addition to blowing the reed pipe in a standard fashion, a mechanical driver was used to excite the reed. Using both excitation sources, a number of deflection shapes were observed including simple cantilever, torsional, and higher-order shapes corresponding to higher-order cantilever and torsional modes. As expected, the observed frequencies of the mechanically driven modes were not integer multiples of the fundamental, and were consistent with theoretical predictions. The reed pipe was also excited in a standard manner using an organ blower. This raised integer multiples of the fundamental frequency as high as 20 kHz within two decades of the velocity amplitude of the fundamental. Torsional and other deflection shapes were present, however nonlinear interactions in the system caused them to be shifted from their mechanically driven frequencies. In some cases, significant vibration was observed in the tuning wire and the section of reed above the tuning wire that was previously considered to be clamped.

1:30

2pMUa3. An inward striking free reed coupled to a cylindrical pipe.
Justin Vines (Univ. of Arkansas, Fayetteville, AR 72701, jvines@uark.edu), Ammon Paquette (Augustana College, Rock Island, IL 61201), and James P. Cottingham (Coe College, Cedar Rapids, IA 52402)

A number of acoustical measurements have made on a reed-pipe combination consisting of a harmonium-type reed from an American reed organ installed at the closed end of a cylindrical pipe. This configuration, which somewhat resembles the configuration of free-reed organ pipes, differs from the reed-pipe combination occurring in the mouth organs of Asia, which use symmetric (outward striking) free reeds and normally operate on both possible directions of airflow. Measurements have been made of the sounding frequency, amplitude of vibration of the reed tongue, and the sound spectrum. Of particular interest is the degree to which the reed frequency can be altered by altering the pipe length, and hence the pipe resonance frequency. In this case the sounding frequency can be pulled considerably below the natural frequency of the reed. These results can be compared with the results of similar measurements on free-reed organ pipes [J. Braasch, C. Ahrens, J. P. Cottingham, and T. D. Rossing, Fortschr. Akust., DAGA (2000)]. In addition, some interesting ''special effects'' have been studied, which can be obtained using unusual pipe lengths and blowing in the ''wrong'' direction. [Work supported by the NSF from REU Grant No. 0139096.]

1:45

2pMUa4. Direct measurement of clarinet air column oscillations.
Jesse JonesIV, Chris Rogers (Mech. Eng. Dept., Tufts Univ., 200 College Ave., Medford, MA 02155), and Chris French (The Selmer Co., Elkhart, IN 46514)

The internal oscillation of a clarinet air column has been directly measured through the implementation of hot-wire anemometry. By taking a series of measurements down the centerline of the bore, velocity and pressure modal shapes of individual harmonics are separated, measured, and plotted. Finally, composite averaged power spectra of the internal oscillation are presented and compared to acoustic measurements acquired outside the clarinet. In many cases, the even harmonics of the internal oscillation dominate over the power found in the odd harmonics. This contradicts the classic model of the clarinet as a cylindrical pipe closed at one end and open at the other (where only odd harmonics are produced). Further, the data from the direct velocity measurements also contradict the externally acquired acoustic data, where odd harmonics generally dominate for the lowest 5–9 harmonics. Thus the clarinet, in theory and practice, is generally considered incapable of generating strong even harmonics. In this research, however, it is seen that dominate even harmonics are generated, but the energy for these frequencies is largely trapped inside the clarinet, whereas the energy associated with the odd harmonics is released to the ambient. [This research was conducted with the support of Selmer Musical Instruments.]

2:00

2pMUa5. Airflow patterns in the vicinity of an air-driven free reed.
Jesse T. Jensen and James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402, jtjensen@coe.edu)

Free reed instruments are characterized by high-volume airflow rates through the oscillating reeds. Measurements have been made of the average volume flow rate through a single American organ reed as a function of blowing pressure. In addition, measurements of average air velocity have been made at a grid of points close to the vibrating air-driven reed, mounted on the surface of a laboratory wind chest. These measurements supplement earlier investigations, which explored relations among the reed motion, airflow, and acoustic pressure associated with the vibration of free reeds. [M. Busha and J. P. Cottingham, J. Acoust. Soc. Am. **111**, 2376 (2002)]. An attempt has been made to determine whether the left–right asymmetry due to the asymmetric shape (twist) of the reed is reflected in the airflow pattern. A graphing program has been developed so that the airflow pattern can be visualized in two or three dimensions.

2:15

2pMUa6. The acoustic effect of cryogenically treating trumpets.
Jesse JonesIV and Chris Rogers (Dept. of Mech. Eng., Tufts Univ., 200 Anderson Hall, 200 College Ave., Medford, MA 02155)

The acoustic effect of cryogenically treating trumpets is investigated. Ten Vincent Bach Stradivarious $B\flat$ trumpets are studied, half of which have been cryogenically treated. The trumpets were played by six players of varying proficiency, with sound samples being recorded directly to disk at a sampling rate of 44.1 kHz. Both the steady-state and initial transient portions of the audio samples are analyzed. When comparing the average power spectra of the treated trumpets to the untreated set, no repeatable, statistically independent differences are observed in the data. Differences observed in player-to-player and trumpet-to-trumpet comparisons overshadow any differences that may have been brought on due to the cryogenic treatment. Qualitatively, players established no clear preference between the treated and untreated trumpets regarding tone and playability, and could not differentiate between the two sets of instruments. All data was collected in a double blind fashion. The treatment itself is a three step process, involving an 8 hour linear cool down period, a 10 hour period of sustained exposure to $-195\,^{\circ}\mathrm{C}$ ($-300\,^{\circ}\mathrm{F}$), and a 20–25 hour period of warming back to room temperature. [Work was completed with the support of Steinway & Sons Pianos and Selmer Musical Instruments.]

TUESDAY AFTERNOON, 11 NOVEMBER 2003 — SAN ANTONIO ROOM, 2:45 TO 4:25 P.M.

Session 2pMUb

Musical Acoustics: Where Are They Now? Current Research by Past Student Paper Award Winners

James P. Cottingham, Chair
Physics Department, Coe College, Cedar Rapids, Iowa 52402

Invited Papers

2:45

2pMUb1. Where they are now: An overview.
James P. Cottingham (Dept. of Phys., Coe College, Cedar Rapids, IA 52402, jcotting@coe.edu)

Since 1997 there have been fifteen award winners in the Best Student Paper competition in Musical Acoustics. Although some of the award winners are still active in musical acoustics, others are now active in other areas of acoustics or in fields outside acoustics altogether. A brief statistical overview will be presented of the history of the competition and past and current interests of those who have been the award winners. Capsule updates on several award winners who are unable to participate in the session will be presented.

3:10

2pMUb2. Sound radiation and phase mapping of Caribbean steelpans. Andrew Morrison (Dept. of Phys., Northern Illinois Univ., DeKalb, IL 60115)

At the 145th ASA meeting [J. Acoust. Soc. Am. **113**, 2315 (2003)] modal analysis and sound intensity plots of two Caribbean steelpans were presented. Since that meeting the relationship of pan dimension to radiated sound has been examined. Sound radiation was explored using a two-microphone probe to gather sound intensity measurements. Preliminary phase maps of vibrating note areas have also been obtained using holographic interferometry with phase imaging.

3:35

2pMUb3. Future evolution of structure in an accelerating universe. Michael T. Busha (Randall Lab., Univ. of Michigan, 500 E. University Ave., Ann Arbor, MI 48109)

Current cosmological data have confirmed the big bang and dark matter hypothesies, and indicate that our universe contains a substantial component of dark vacuum energy that is driving the cosmos to accelerate. We examine the immediate and long-term consequences of this dark energy on large scale structures formed in the universe. Using spherical model solutions and realistic 3D clustering simulations, we present criteria for test bodies to remain bound to existing structures. We show that collapsed halos become spatially isolated and dynamically relax to a particular kinematic profile. From this state, the asymptotic form of the space-time metric is well specified.

4:00

2pMUb4. Three-dimensional sound signals and their relevance to wave energy quantities and sound interference products. Pantelis Vassilakis[a] (Music, De Paul Univ., 2350 N. Kenmore Ave., JTR 307, Chicago, IL 60614, pantelis@acousticslab.com)

Signals are graphic representations of vibrations/waves and, like every representation, capture only selected attributes of the phenomenon they are meant to represent. The often assumed equivalence between signals and sound waves obscures the fact that two-dimensional signals are not fit to (a) represent wave-energy quantities consistently across frequencies, (b) account for the alternating positive/negative amplitude values of modulated waves with AM-depth$>$100%, and (c) represent the energy content of interference. An alternative sound-signal representation is proposed, based on the complex equation of motion describing a wave. It results in spiral sine signals and twisted-spiral complex signals, similar to complex analytic signals. Spiral sine signals offer a consistent measure of sine-wave energy across frequencies, while twisted spiral complex signals account for the negative amplitudes observed in modulated signals and map the modulation parameters onto the twisting parameters. In terms of interference, 3-D signals illustrate that amplitude fluctuations and the signal envelopes that describe them are not just boundary curves but waves that trace changes in the total instantaneous energy of a signal over time, representing the oscillation between potential and kinetic energies within a wave. Examples of 3-D animations illustrating the proposed signals are presented. [a] Work completed while at the Department of Ethnomusicology, University of California, Los Angeles.

TUESDAY AFTERNOON, 11 NOVEMBER 2003 WEDGWOOD ROOM, 5:00 TO 5:50 P.M.

Session 2pMUc

Musical Acoustics: Performance Session: Concert Featuring Wind Players from the University of Texas

James P. Cottingham, Chair
Physics Department, Coe College, Cedar Rapids, Iowa 52402

Chair's Introduction—5:00

This performance technical session will feature chamber music for winds played by musicians selected from the wind ensembles of the University of Texas School of Music. The program has been organized by Dr. Scott S. Hanna. Dr. Hanna serves as Assistant Director of Bands at the university as well as Associate Director of the Longhorn Band and Music Director of the UT Chamber Winds.

Session 2pNSa

Noise and Engineering Acoustics: Sound Quality for Engineered Products and the Environment

Brigitte Schulte-Fortkamp, Cochair
Institute of Technical Acoustics, Technical University Berlin, Einsteinufer 25, 10587 Berlin, Germany

Klaus Genuit, Cochair
HEAD Acoustics GmbH, Eberstrasse 30a, 52134 Herzogenrath, Germany

Patricia Davies, Cochair
Ray W. Herrick Laboratories, Purdue University, West Lafayette, Indiana 47907-2031

Chair's Introduction—1:00

Invited Papers

1:05

2pNSa1. Speech intelligibility index predictions for young and old listeners in automobile noise: Can the index be improved by incorporating factors other than absolute threshold? Meghan Saweikis, Aimée M. Surprenant (Dept. of Psychol. Sci., Purdue Univ., 703 3rd St., West Lafayette, IN 47907), Patricia Davies, and Don Gallant (Purdue Univ., West Lafayette, IN 47907-2031)

While young and old subjects with comparable audiograms tend to perform comparably on speech recognition tasks in quiet environments, the older subjects have more difficulty than the younger subjects with recognition tasks in degraded listening conditions. This suggests that factors other than an absolute threshold may account for some of the difficulty older listeners have on recognition tasks in noisy environments. Many metrics, including the Speech Intelligibility Index (SII), used to measure speech intelligibility, only consider an absolute threshold when accounting for age related hearing loss. Therefore these metrics tend to overestimate the performance for elderly listeners in noisy environments [Tobias *et al.*, J. Acoust. Soc. Am. **83**, 859–895 (1988)]. The present studies examine the predictive capabilities of the SII in an environment with automobile noise present. This is of interest because people's evaluation of the automobile interior sound is closely linked to their ability to carry on conversations with their fellow passengers. The four studies examine whether, for subjects with age related hearing loss, the accuracy of the SII can be improved by incorporating factors other than an absolute threshold into the model. [Work supported by Ford Motor Company.]

1:25

2pNSa2. Prediction of booming sensation and its difference limen for just noticeable change in frequency. Sung-Hwan Shin and Jeong-Guon Ih (ME3075, NOVIC, Dept. of Mech. Eng., KAIST, Taejon, Korea)

Among many auditory feelings for the car interior noise, the booming sensation is considered the most important nuisance to the passengers. Although there are many origins for the booming noise of vehicles in general, the most important one is the engine boom that consists of tonal components related to fundamental engine rotation and its harmonics including the firing frequency. Because the degree of booming sensation is increased when these tonal components are dominating in car interior noise, it is demanded to extract the aurally relevant tonal components only. Although the pitch extraction model based on the place theory enables to find aurally relevant tonal components, there is a difference between booming sensation and pitch perception according to a frequency change of the tonal component. In this study, a subjective listening test using a tracking method is performed to find the difference limen for just a noticeable change of booming sensation in frequency. By applying the resultant data and also the empirical data by Zwicker, the existing pitch extraction model is modified. This refined model and loudness analysis can be used for predicting the degree of booming sensation. [Work supported by the BK21 project and NRL.]

1:45

2pNSa3. Sound quality evaluation of air conditioning sound rating metric. Kathleen K. Hodgdon, Jonathan A. Peters, Russell C. Burkhardt (Appl. Res. Lab, Penn State Univ., University Park, PA 16802), Anthony A. Atchley, and Ingrid M. Blood (Penn State Univ., University Park, PA 16802)

A product's success can depend on its acoustic signature as much as on the product's performance. The consumer's perception can strongly influence their satisfaction with and confidence in the product. A metric that can rate the content of the spectrum, and predict its consumer preference, is a valuable tool for manufacturers. The current method of assessing acoustic signatures from residential air conditioning units is defined in the Air Conditioning and Refrigeration Institute (ARI 270) 1995 Standard for Sound Rating of Outdoor Unitary Equipment. The ARI 270 metric, and modified versions of that metric, were implemented in software with the flexibility to modify the features applied. Numerous product signatures were analyzed to generate a set of synthesized spectra that targeted spectral configurations that challenged the metric's abilities. A subjective jury evaluation was conducted to establish the consumer preference for those spectra. Statistical correlations were conducted to assess the degree of relationship between the subjective preferences and the various metric calculations. Recommendations were made for modifications to improve the current metric's ability to predict subjective preference. [Research supported by the Air Conditioning and Refrigeration Institute.]

2:05

2pNSa4. Methodology for quantifying the tonal prominence of frequency modulated tones. Kyoung Hoon Lee, Patricia Davies (Ray W. Herrick Labs., Purdue Univ., 140 S. Intramural Dr., West Lafayette, IN 47907-2031), and Aimee Surprenant (Purdue Univ., West Lafayette, IN 47907)

The paper is focused on research conducted to develop a metric to quantify the tonal prominence of frequency modulated tones in noise. Frequency modulated tones are commonly encountered in machinery noise because of RPM variations caused by changing loads and poor control in timing and repeatability of combustion events in engines, particularly diesel engines. When tonal components are noticeable they increase annoyance; thus it is important to quantify both their strength and the resulting contribution to annoyance. In preliminary work on tones with randomly varying frequencies, it was found that it is possible to remove the trackable portion of the frequency variation and use established tonal prominence metrics (Aures' Tonality and Tone-to-Noise Ratio) on the modified signal to predict the perceived tonalness of the sounds (Lee, Hastings, Davies and Surprenant, Internoise 2003). The work presented is an extension of this earlier work: deterministic frequency variations are studied along with different levels of background noise. Modifications to the procedure to determine the strength of the tonal feature are described, as well as the issues that must be addressed before using this approach to analyze more complex machinery sounds. [Work sponsored by Caterpillar Inc.]

2:25

2pNSa5. A loudness model in dichotic conditions. Jeong-Guon Ih and Jeong-Ho Cha (Dept. of Mech. Eng., KAIST, Science Town, Taejon 305-701, Korea, ihih@sorak.kaist.ac.kr)

Existing loudness models are specified only to diotic sounds in spite of the fact that normal human beings hear dichotic sounds. The arithmetic mean of loudness values of both ear signals has been suggested for the approximated value of the resultant perceived loudness. In this study, the dependence of overall loudness perception on the interaural level differences was investigated by the subjective tests. It was found that the larger the interaural level difference, the louder the perception than the mean of calculated loudness values at both ears and the lower the critical band rate or the reference level, the louder the perception than the mean value. A modified loudness model was proposed for the application to dichotic sounds by using the equivalent diotic levels.

2:45–3:00 Break

3:00

2pNSa6. Sound preferences in urban open public spaces. Jian Kang and Wei Yang (School of Architecture, Sheffield Univ., Western Bank, Sheffield S10 2TN, UK, j.kang@sheffield.ac.uk)

This paper studies people's perception of sound, based on an intensive questionnaire survey in fourteen urban open public spaces of five European countries. The questionnaire includes identification of recognized sounds, classification of sound preference, and indication of wanted and unwanted sounds. The results indicate three facets to people's sound preferences. First, people generally prefer natural and culture-related sounds rather than artificial sounds. Vehicle sounds and construction sounds are regarded as the most unpopular, whereas sounds from human activities are normally rated as neutral. Second, cultural background and long-term environmental experience play an important role in people's judgment of sound preference. People from a similar environment may show a similar tendency on their sound preferences, which can be defined as macro-preference. Third, personal differences, such as age and gender, further influence people's sound preference, which can be defined as micro-preference. For example, with increasing age, a higher percentage of people are favorable to, or tolerate, sounds relating to nature, culture or human activities. Male and female exhibit only slight differences. [Work supported by the European Commission.]

3:20

2pNSa7. Environmental noise—a challenge for an acoustical engineer. Klaus Genuit (HEAD acoustics GmbH, Ebertstrae 30a, D-52134 Herzogenrath, Germany, klaus.genuit@head-acoustics.de)

People live in a landscape full of noises which are composed of both natural environmental noises and technically created sounds. Regarding environmental noise, more and more people feel heavily annoyed by noises. Noise is defined as an audible sound which either disturbs the silence or an intentional sound listening or leads to annoyance. Thus, it is clearly defined that the assignment of noise cannot be reduced to simple determining objective parameters such as the A-weighted sound pressure level or the equivalent continuous sound pressure level. The question of whether a sound is judged as noise can only be made after the transformation from the sound event into an auditory event has been accomplished. The evaluation of noise depends on the physical characteristics of the sound event, on the psycho-acoustical features of the human ear, as well as on the psychological aspects of man. For the acoustical design of environmental noise and in order to create a better soundscape the acoustical engineer has to consider these aspects. That means a specific challenge for the sound engineering.

3:40

2pNSa8. (Re-)constructed reality and context—When sound quality becomes soundscape. Brigitte Schulte-Fortkamp (ITA/TU–Berlin, Einsteinufer 25, 10587 Berlin, Germany)

In the interior of a vehicle sound quality is a brandname. Bi-aurale measurement and analysis technology is more or less standard, but there are still no general standards and parameters for sound quality. Evaluation of noises is highly sensitive to context, but testing procedures that can include this realization are exceedingly rare. Soundscapes are specific constellations of a general noise volume in clearly defined ambiences; they even can characterize those ambiences and combine the daily recurrent patterns of sound multi-factorial in the process of analysis. The evaluation of noise situations gets interactively modified by their significance for the general residential space. It is contingent upon the respective ponderation of acoustic and nonacoustic modifiers which in turn constitute the soundscape.

Contributed Papers

4:00

2pNSa9. Sound quality descriptors for HVAC equipment from ARI Standards. Stephen J. Lind (Trane, Bldg. 12-1, 3600 Pammel Creek Rd., La Crosse, WI 54601, slind@trane.com)

The Air Conditioning and Refrigeration Institute (ARI) has several standards that provide methods to evaluate the sound quality of heating ventilating and air-conditioning (HVAC) equipment. These include Standard 270 Sound rating of outdoor unitary equipment, Standard 350 Sound rating of non-ducted indoor air-conditioning equipment, and Standard 1140P Procedures for evaluating sound quality of HVAC equipment. The preferred method in these standards is best described in Standard 1140P, which uses one-third octave band sound power levels that are weighted to adjust for the sensitivity to frequency distribution and presence of tones, and are then converted to a single number sound quality indicator. The tone adjustment is based on the projection of a given one-third octave band level relative to the average of the adjacent one-third octave bands. An alternate use of Zwicker method B to determine loudness and loudness level is also provided in ARI Standard 1140P. These standards provide a convenient method by which complex sounds for similar products may be compared.

4:15

2pNSa10. Traffic noise attenuation by scattering, resonance and dispersion. Hasson M. Tavossi (Dept. of Physical and Environ. Sci., Mesa State College, School of Math. & Natural Sci., 1100 North Ave., Grand Junction, CO 81501)

The purpose of this investigation is to analyze various techniques of sound attenuation which can be used to minimize traffic noise from heavy traffic circulation in urban areas, and to improve sound quality in these areas using selective absorption in specific frequency ranges, plus to reduce the degree of annoyance of traffic noise. This study covers uncontrollable noise sources such as trucks, automobiles and trains, in heavy traffic urban areas. Different methods of noise energy reduction are discussed, including sound attenuation by selective absorption, scattering and cavity resonance in the sound barriers. Multiple scattering on sound absorbing road and barriers are considered to increase noise energy dissipation in unwanted frequency ranges. The effects of dispersion and change in the frequency content of the noise with distance are studied. An elimination of the most disturbing frequency components of these uncontrollable noise sources improves sound quality from these sources and increases our overall capabilities in controlling noise pollution in the urban environment.

TUESDAY AFTERNOON, 11 NOVEMBER 2003 CONCHO ROOM, 1:30 TO 3:55 P.M.

Session 2pNSb

Noise, Architectural Acoustics and ASA Committee on Standards: Building Code Noise Compliance

Daniel R. Raichel, Chair
2727 Moore Lane, Fort Collins, Colorado 80526-2192

Chair's Introduction—1:30

Invited Papers

1:35

2pNSb1. Model building codes and acoustical performance: Where are we in 2003? Brandon Tinianov (Johns Manville Tech. Ctr., 10100 W. Ute Ave., Littleton, CO 80127)

The proper acoustical design for multi-family dwellings is an important factor in occupant comfort. Key acoustical design practices are often not mandated by the builder or architect, but by the applicable building codes. In early 2003, the three regional/national building codes agreed to join into a single, unified national building code for residential and commercial construction. The scope and governance of these three codes: the Uniform Building Code (ICBO), the National Building Code (BOCA), the Southern Building Code (SBCCI) are reflected in the International Residential Code (IRC) and the International Building Code (IBC) which was developed by the International Code Council (ICC). With the move to a single code body, those concerned with building acoustical performance welcome the benefit of a single minimum standard. Unfortunately, this new minimum performance requirement does not reflect the state of the science for occupant satisfaction. The acoustical requirements of each of these building codes, the timeline of their development and an overview of the state of the science will be presented. Suggestions for revised performance minimums will also be offered for discussion.

1:55

2pNSb2. Field normalization techniques and practices for determining sound isolation and impact insulation. John J. LoVerde and David W. Dong (Veneklasen Assoc., 1711 Sixteenth St., Santa Monica, CA 90404, jloverde@veneklasen-assoc.com)

The International Building Code (IBC) includes minimum acceptable acoustical performance for floor/ceiling assemblies separating dwelling units. Field tests for determining the Normalized Noise Isolation Class (NNIC) and Field Impact Insulation Class (FIIC) of an assembly require the measurement of the decay rate of the receiving room and the normalization of the sound pressure levels, as set forth in ASTM standards E336 and E1007. The normalization is intended to improve comparability of test results by removing the effects of receiving room absorption. Surprisingly, the normalization procedure is substantially different between these two types of tests. The FIIC normalization is based on the laboratory Impact Insulation Class (IIC) test using a standard room absorption of 10 metric sabins. The NNIC normalization is based on a 0.5-s reverberation time, which is assumed normal in a furnished habitable dwelling space. This difference in normalization procedure for the same receiving room potentially yields different results of NNIC and FIIC as compared with their respective non-normalized values. This will be illustrated with examples of actual field tests. The effectiveness and suitability of the normalization procedures in the field setting are compared and evaluated including the use of the California Building Code modification to FIIC.

2:15

2pNSb3. ANSI S12.60 (2002) testing of modular buildings used as classrooms. Todd A. Busch (Acentech, 1429 E. Thousand Oaks Blvd., Ste. 200, Thousand Oaks, CA 91362)

The acoustics of modular buildings used as classrooms may come under greater scrutiny if schools adopt American National Standards Institute (ANSI) S12.60 (2002) ''Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools'' as required thresholds of performance. To quantify the benefits of series of structural improvements to buildings having an interior volume of less than 10 000 ft^3, both the reverberation time and background noise were measured. Interior spaces were unoccupied as per S12.60, but unfurnished. Reverberation times were 0.4–0.6 s. The background noise level was measured to identify the contributions of exterior sources, interior fluorescent lighting, and AC/Fan systems operating with/without a plenum for the return vent. Targets for the stated room volume are 35 dBA and 55 dBC. For the final configuration, interior noise levels with lights, fan, and cooling active were 43 dBA and 67 dBC; interior noise levels due to exterior sources were 33 dBA and 58 dBC due in part to the surrounding industrial land uses and nearby freeway. For one configuration, with the fan active without AC, pressurized air escaping through a 1/4-in. gap beneath the 36-in.-wide door produced a distinct 35 dBA tone in the 630-Hz third-octave band.

2:35

2pNSb4. Noise codes for residential spaces in Los Angeles. Daniel R. Raichel (Eilar & Assoc., 2727 Moore Ln., Fort Collins, CO 80526, raichel@juno.com)

The State of California is rather unique in using the community noise equivalent level (CNEL) rating to assess the amount of environmental noise transmission into buildings, not just to apply this index to gauge airport noise. When residential housing units in Los Angeles undergo construction or renovation, a certificate of housing compliance must be obtained before tenants or condominium owners are allowed to move in, in accordance with the provisions of Section 91.0318 of the Los Angeles municipal code. As a part of the application procedure for the certificate, a licensed acoustical engineer must prepare an acoustical report that describes the type of construction between dwelling units and the general sound attenuating properties of such construction, including hard numbers for sound attenuation between dwelling units. A field sound transmission class (FSTC) test is mandated, with a minimum required FSTC rating of 45 between the walls of adjacent units. If the FSTC test is not passed, remedial action must be taken so that the proper amount of acoustic insulation is achieved between the dwelling units.

2:55–3:05 Break

3:05

2pNSb5. Evaluation of acoustical parameters of a Brazilian popular housing model. Jos A. C. Ferreira (Acoust. Lab., Dept. of Mech. Eng., UFPR, 81531-990, Curitiba, PR, Brazil), Fabiano B. Diniz, Andressa M. C. Ferreira, and Paulo T. Zannin (Acoust. Lab., UFPR)

This article presents the results obtained from the evaluation of the acoustical insulation parameters determined *in situ* in a popular residence projected to offer an option to combat the housing deficit of the low income Brazilian population. This evaluation has been carried out according to the statements of the standards ISO 140-4 and 140-5, which state about this type of measurement. The results have shown that the surveyed house presents a satisfactory performance if compared to the standard of the Brazilian civil construction, but it is not adequate if compared to the demands of the international standards.

Contributed Papers

3:25

2pNSb6. Residential reuse: Acoustics and making buildings livable. Jesse J. Ehnert (Arpeggio Acoust. Consulting, LLC, 1947 Aspen Dr. NE, Atlanta, GA 30345, jehnert@arpeggioacoustics.com)

Adapting an existing facility, whether it is industrial, educational, or commercial in nature, to residential uses, such as condominiums and lofts, poses many challenges to the design team. Among other things, the process must include maintaining historic character while addressing current code issues in the context of becoming familiar with and utilizing existing architecture and building systems, which may not always be well-documented or obvious. Of the many unique issues encountered in such adaptive reuse projects, acoustic concerns often present themselves, especially in terms of sound and vibration isolation and noise control within and between residential units. Some of the paramount concerns and pitfalls will be discussed in the context of relevant examples and past experiences.

3:40

2pNSb7. Compressor noise control with a two-dimensional enclosure. Ballard W. George (Environtech Consultants, 1367 Bobolink Circle, Sunnyvale, CA 94087, kingstaco@yahoo.com)

This paper is concerned with the use of a two-dimensional enclosure with masonry block wall to mitigate compressor noise in a northern California location for consistency with city standards. Sensitive receptors consist of a duplex and single residences across the street from the source. Background sound is provided primarily by a nearby highway that is elevated. Noise reduction is accomplished by the barrier effect of the wall along with sound absorption realized from a pegboard wall surface in the vicinity of the compressor, unpainted concrete block surface and acoustic treatment on the doors. Design considerations for the pegboard are discussed with the intention of balancing the overall low- and high-frequency noise reduction from the absorption and from the barrier effect and with the intention of reasonably reducing the number of pegboard holes required. Some considerations from the standpoint of ray acoustics are discussed. A complicating factor for the interior acoustics consists of three full height tanks in a row that serve as sound diffusers. Comments on vibration isolation for equipment and pipes are offered.

TUESDAY AFTERNOON, 11 NOVEMBER 2003 TRINITY B ROOM, 1:30 TO 5:00 P.M.

Session 2pPA

Physical Acoustics: Nonlinear Acoustics and Scattering

Kendall R. Waters, Chair
NIST, Materials Reliability Division, 325 Broadway, Boulder, Colorado 80305

Contributed Papers

1:30

2pPA1. Calculation of the acoustic nonlinearity parameter B/A for four linear alkanes by Lee–Kesler equation of state. Zhiqu Lu (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, Coliseum Dr., University, MS 38677)

The acoustic nonlinearity parameter B/A has been calculated for four linear alkanes over the temperature range from 273.15 to 473.15 K and the pressure range from 1 atm to 1000 atm by using the Lee–Kesler equation of state. The calculated results are in agreement with the experimental data in the literature. It is found that under low pressure (1 atm and 20 atm), B/A increases linearly with temperature. Under high pressure, the value of B/A increases with temperature first, however, at a slower rate of rise. After a broad maximum, the ratio B/A decreases slowly as the temperature increases. At the fixed temperature, the B/A value decreases with the number of carbon chain of alkanes. It is shown that the B/A decreases monotonically with pressure over the whole temperature and pressure ranges.

1:45

2pPA2. Nonlinear acoustic acceleration waves in porous media flow. Pedro M. Jordan (Naval Res. Lab., Code 7181, Stennis Space Center, MS 39529, pjordan@nrlssc.navy.mil)

Acoustic acceleration waves are defined as jumps in the first derivatives of the velocity, pressure, or density across a propagating singular surface (or wave front). In this talk, the temporal evolution of the amplitude and the propagation speed of such waves are investigated in the context of finite-amplitude acoustic propagation in inelastic Darcy-type porous media. It is shown that there exists a critical value, $\alpha^*(>0)$, of the initial jump amplitude such that the acceleration wave magnitude either goes to zero, as $t\to\infty$, or infinity, in finite time, depending on whether the given initial jump amplitude is greater than or less than $-\alpha^*$. In addition, a connection to traveling wave solutions is noted and the linearized case is examined. Finally, the numerical solution of a one-dimensional, nonlinear IBVP involving sinusoidal signaling in a fluid-saturated porous slab is used to illustrate the finite-time transition from an acceleration wave to a shock wave that occurs when the given initial jump amplitude is less than $-\alpha^*$. [Work supported by ONR/NRL funding.]

2:00

2pPA3. Feasibility of nonlinear acoustic technique with ultrasonic guided waves for nondestructive defect evaluation. Kyung-il Jung and Suk Wang Yoon (Dept. of Phys., SungKyunKwan Univ., Suwon 440-746, Republic of Korea)

Feasibility of nonlinear acoustic modulation technique [K. I. Jung and S. W. Yoon, Proceedings of Internoise 2000, Nice, France, p. 205 (2000)] was discussed for nondestructive defect evaluation. Such nonlinear acoustic technique can be applied to localize a defect in a solid rod and also extended to layer-structured media. In this study, nonlinear acoustic technique with ultrasonic guided waves is introduced. Nonlinear acoustic responses from a defect are observed with guided waves in a 2-D structure. The defect was oriented in the direction of guided wave propagation. The experimental results show that the nonlinear acoustic technique with ultrasonic guided waves seems feasible for nondestructive evaluation of 2-D structure. [Work supported by BK21 Program in Korea and by the Basic Research Program of KOSEF (R01-2000-000-00014-0).]

2:15

2pPA4. Nonlinear modal methods for crack localization. Alexander Sutin (Stevens Inst. of Technol., Hoboken, NJ 07030), Lev Ostrovsky (Zel Technologies/NOAA ETL, Boulder, CO 80305), and Andrey Lebedev (Inst. of Appl. Phys., Nizhnny Novgorod 603095, Russia)

A nonlinear method for locating defects in solid materials is discussed that is relevant to nonlinear modal tomography based on the signal cross-modulation. The scheme is illustrated by a theoretical model in which a thin plate or bar with a single crack is excited by a strong low-frequency wave and a high-frequency probing wave (ultrasound). A crack is considered as a small contact-type defect which does not perturb the modal structure of sound in linear approximation but creates combinational-frequency components whose amplitudes depend on their closeness to a resonance and crack position. Using different crack models, including the hysteretic ones, the nonlinear part of its volume variations under the given stress and then the combinational wave components in the bar can be determined. Evidently, their amplitude depends strongly on the crack position with respect to the peaks or nodes of the corresponding linear signals which can be used for localization of the crack position. Exciting the sample by sweeping ultrasound frequencies through several resonances (modes) reduces the ambiguity in the localization. Some aspects of inverse problem solution are also discussed, and preliminary experimental results are presented.

2:30

2pPA5. Parametric wave phase conjugation of nonlinear ultrasound waves. Andrew Brysev, Vladislav Mikhalevich, and Vladimir Streltsov (Wave Res. Ctr. of General Phys. Inst., RAS, 38 Vavilov str., 119991 Moscow, Russia, brysev@orc.ru)

Real time acoustic wave phase conjugation (WPC), based on parametric self-consistent physical mechanisms, was realized up to the present time only for the monochromatic waves [A. P. Brysev *et al.*, Phys.-Usp. **41**, 793 (1998)]. Here the possibility of WPC of nonmonochromatic ultrasound waves is considered. For simultaneous WPC of the entire series of spectral components generated by nonlinear propagation of the incident wave we propose the use of phonon-plasmon interaction in piezosemiconductors. WPC of nonlinear acoustic waves can be accomplished by modulation of the electron density provided by a sequence of short laser pulses pumping the sample. If the periodicity of the optical pulses is half the period of the fundamental component of the acoustic wave, such wideband, excitation leads to self-synchronized parametric conjugation of each spectral component in the incident wave. The conjugation efficiency depends sharply on relations between acoustical frequency content, laser pulse duration, and interband relaxation time. It is shown that under certain conditions the time profile of the conjugate wave may be efficiently controlled by varying the duration of the laser pulses. The time profile of the conjugate wave is investigated for some physical conditions of practical interest.

2:45

2pPA6. Radiation force on a compressible cylinder in a standing wave: Long-wavelength approximation. Wei Wei, David B. Thiessen, and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

For situations where compressible cylinders are manipulated using acoustic standing waves, it is desirable to have a simple long wavelength approximation of the radiation force analogous to the sphere result published by Yosioka and Kawasima in 1955. That result has also been derived from the Rayleigh scattering approximation of the monopole and dipole scattering contributions. The analogous result for a rigid-movable cylinder (that may have a different density than the surroundings) was given by the first line of Eq. (24) of Wu *et al.* [J. Wu, G. Du, S. S. Work, and D. M. Warshaw, J. Acoust. Soc. Am. **87**, 581–586 (1990)]. The cylinders axis is parallel to the standing wave nodal planes. To generalize that result to a compressible cylinder it is only necessary to replace a monopole term [the final unity term in the curly brackets in the first line of Eq. (24)] by $(l-R)$ where R denotes the ratio of the adiabatic compressibility of the inner fluid to that of the outer fluid. This modification agrees with the form of the monopole scattering contribution for cylinders in the Rayleigh approximation and recovers numerically the low frequency limit for the radiation force-per-length based on the full partial-wave series. [Work supported by NASA.]

3:00–3:15 Break

3:15

2pPA7. Acoustics spectral gaps and transmission in periodic stub tuners. M. S. Kushwaha (Inst. of Phys., Univ. of Puebla, Puebla, Mexico), P. Vasilopoulos, and X. F. Wang (Concordia Univ., Montreal, Canada)

Periodic binary systems can give rise to genuine acoustic stop bands within which sound and vibrations remain forbidden. Extensive band structure and transmission are computed for a periodically stubbed waveguide. In general the waveguide segments and stubs are made of different materials. The acoustic wave in such a system has two independent polarizations: out-of-plane and in-plane modes. The band structure and transmission spectrum are studied for diverse geometries using a simple and efficient version of the transfer-matrix method [Phys. Rev. B **65**, 035107 (2002)]. For the same material between the waveguide and symmetric stubs the width of some gaps can change, upon varying the stub length or width, by more than one order of magnitude. A further modulation can be achieved for different materials between the stubs and the main waveguide or if the stubs are asymmetric. The spectral gaps in the band structure of an infinitely long system correspond to those in the transmission spectrum of the same system but with a *finite* number n of units. For n finite (i) there exist *pseudogaps* that gradually turn into *complete* gaps with increasing n and (ii) the introduction of defects gives rise to states inside the gaps and leads to transmission resonances.

3:30

2pPA8. Acoustic propagation using the direct simulation Monte Carlo method. Amanda L. Danforth and Lyle N. Long (Penn State Univ., 202 Appl. Sci. Bldg., P.O. Box 30, State College, PA 16804, ald227@psu.edu)

In the simulation of fluid dynamics, one can either treat the fluid as a continuum or as discrete particles. Although popular for acoustics, the continuum model is limited to small Knudsen numbers (the ratio of mean free path to a length scale). Particle methods are necessary for, but not limited to, problems with Knudsen numbers greater than 0.1, which can occur in shockwaves, microdevices, or rarefied gases. Some well known particle methods include Molecular Dynamics, Monte Carlo methods, and the Lattice Boltzmann Method. The Direct Simulation Monte Carlo (DSMC) method describes gas flows through direct physical modeling of particle motions and collisions. DSMC can model linear and nonlinear acoustics, as well as the details of viscous dissipation. DSMC results will be shown and compared with continuum theory. A DSMC method has been implemented for 1-dimensional linear and nonlinear acoustics problems on parallel computers using object-oriented C++ and the message passing interface (MPI). The code has been run on large Beowulf clusters. The results have been compared to solutions to the linear wave equation. The complete paper will include two-dimensional solutions as well as comparisons to nonlinear continuum results. [Work supported by Consortium for Education in Many-Body Applications, Grant No. NSF-DGE-9987589.]

3:45

2pPA9. Modeling of ultrasonic backscatter using ultrasonic radiative transfer theory. Goutam Ghoshal and Joseph A. Turner (Dept. of Eng. Mech., W317.4 Nebraska Hall, Univ. of Nebraska–Lincoln, Lincoln, NE 68588-0526)

The scattering of elastic waves in polycrystalline media is primarily due to the elastic inhomogeneities present between grains. This scattering may be used to extract the microstructural parameters of the material such as grain size and grain texture. In particular, ultrasonic backscatter measurements have been especially useful for extracting microstructural information. From the theoretical perspective, derivations related to ultrasonic radiative transfer equations (URTE) govern the propagation of diffuse intensities and include all multiple scattering effects. In this presentation, a rigorous connection between the URTE theory and the backscatter experiments is discussed. The general URTE formulation is first reduced to the singly scattered problem. The backscattered ultrasonic flux is shown to be the main quantity of interest. The flux is obtained in terms of the relevant scattering parameters. Specific solutions are obtained for a specimen excited by a normally incident longitudinal wave. The backscatter model for a normally incidental longitudinal wave and its mode converted shear wave are discussed. The theoretical results are compared with numerical simulations based on Voronoi polycrystal finite-element calculations. Results are also compared with previous backscattering theories. Relevant applications for materials of common interest are also discussed. [Work supported by DOE.]

4:00

2pPA10. Bragg backscatter and band structure in a sinusoidal waveguide model. Dan Valente (Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804, dpv110@psu.edu) and David C. Swanson (Penn State Univ., State College, PA 16804)

The unique characteristics of wave propagation through periodic acoustic structures have been well known for many years. These structures allow only certain bands of frequencies to propagate through the system, resulting in a distinct filter response. The purpose of this work is to investigate the properties of a periodic waveguide whose walls vary sinusoidally along its length. The waveguide is modeled using a transmission line/transfer matrix approach to obtain the power reflection and transmission coefficients at the waveguide input. Typically in the literature, waveguides with distinct periodic scatterers or abrupt area changes have been examined, and a distinct band structure is evident. It is shown that for small amplitude wall variations of the sinusoidal waveguide, the system denies the transmission of only one frequency. This frequency fits the condition for Bragg backscatter, a phenomenon that is analogous to what is observed in the acousto-optic (AO) effect. As the amplitude of the wall variations increase, however, a band structure similar to what is expected for periodic systems begins to arise. [Work supported by FRA.]

4:15

2pPA11. Further results on acoustic scattering from the edge of a wedge. Ronald Hughes, Jan M. Niemiec, Yuri Stoyanov, and Herbert Überall (Naval Surface Weapons Ctr., Carderock Division)

Acoustic monostatic and bistatic scattering from the edge of a solid wedge immersed in water is analyzed further, always assuming a line source parallel to the edge of a wedge. Our previous work on the subject is extended by emphasizing the dependence of the scattering amplitude on the wedge material, which may be uniform or layered, and which can be characterized by surface impedance. Impedance values are obtained for multiple layers by the method of Brekhovskihk. Our numerical analysis is based upon Macdonald functions [A. Erdelyi, *Higher Transcendental Functions* (McGraw–Hill, New York, 1953)] via an integral representation that we were able to render convergent. The scattering amplitude is integral bilinear in the Macdonald functions and dependent upon the wedge impedance, and has been evaluated numerically. [Work supported by NAVSEA.]

4:30

2pPA12. Measurements of the total scattering and absorption cross-sections of the human body. Stephane G. Conti (Southwest Fisheries Sci. Ctr., 8604 La Jolla Shores Dr., La Jolla, CA 92037), Philippe Roux (Scripps Inst. of Oceanogr., Univ. of California–San Diego, La Jolla, CA 92093-0238), David A. Demer (Southwest Fisheries Sci. Ctr., La Jolla, CA 92037), and Julien de Rosny (Laboratoire Ondes et Acoustique, ESPCI, Universit Paris VII, 75005 Paris, France)

Presented here are the first absolute measurements of the total acoustic energy scattered and absorbed by humans. The total scattering and absorption cross-sections were obtained from long-duration acoustic reverberation, for individual humans walking randomly in a room. An extension of the technique is proposed to estimate the absorption cross-section for the human body. Within the audible range, the sound scattering spectra of the human body is similar to that of a hard ellipsoid with the same volume (dimensions proportional to the weight to the one-third power). Moreover, we show that increasing amounts of clothing have little effect on scattering while absorption is greatly increased.

4:45

2pPA13. Sound sources of the screech tone radiated from circular supersonic jet oscillating in the helical mode. Yoshikuni Umeda (Dept. of Aeronautic & Astronautics, Grad. School of Eng., Kyoto Univ., Kyoto 606-8501, Japan) and Ryuji Ishii (Kyoto Univ., Kyoto 606-8501, Japan)

The generation mechanism of the screech tone in the helical oscillation mode is investigated using a series of instantaneous photographs. From the analysis of the photographs, six evanescent sound sources are observed as prominent points along the jet axis. These sound sources move along circular orbits in the planes perpendicular to the jet axis and slightly downstream of the rear edges of each shock cell. The speed of the moving sound sources is supersonic and moving Mach cones generated behind the moving sound sources form the helical-shaped wave fronts of the screech tone. The existence of the moving Mach cones about the jet axis was confirmed by comparing Schlieren photographs and drawings of the envelopes of the moving Mach cones. [Work supported by The Mitsubishi Foundation.]

Session 2pPP

Psychological and Physiological Acoustics: Normal and Pathological Perception

James D. Miller, Chair
Central Institute for the Deaf, 818 South Euclid, St. Louis, Missouri 63110

Chair's Introduction—1:25

Contributed Papers

1:30

2pPP1. Psychoacoustic processing of test signals. Frantisek Kadlec (Dept. of Radioelectronics, Czech Tech. Univ., Technicka 2, 166 27 Prague, Czech Republic, kadlec@fel.cvut.cz)

For the quantitative evaluation of electroacoustic system properties and for psychoacoustic testing it is possible to utilize harmonic signals with fixed frequency, sweeping signals, random signals or their combination. This contribution deals with the design of various test signals with emphasis on audible perception. During the digital generation of signals, some additional undesirable frequency components and noise are produced, which are dependent on signal amplitude and sampling frequency. A mathematical analysis describes the origin of this distortion. By proper selection of signal frequency and amplitude it is possible to minimize those undesirable components. An additional step is to minimize the audible perception of this signal distortion by the application of additional noise (dither). For signals intended for listening tests a dither with triangular or Gaussian probability density function was found to be most effective. Signals modified this way may be further improved by the application of noise shaping, which transposes those undesirable products into frequency regions where they are perceived less, according to psychoacoustic principles. The efficiency of individual processing steps was confirmed both by measurements and by listening tests. [Work supported by the Czech Science Foundation.]

1:45

2pPP2. On temporal integration versus frequency-based discrimination of direct-to-reverberant energy ratio. Erik Larsen, Nandini Iyer, Albert Feng, and Charissa Lansing (Beckman Inst. for Adv. Sci. and Technol., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801, elarsen@uiuc.edu)

The direct-to-reverberant (D/R) energy ratio is considered to be an important cue for sound source distance judgments (Zahorik, 2002; Bronkhorst, 2001). The current experiments investigated the processes that might be involved in D/R energy ratio discrimination. Two basic processes may be responsible: temporal integration and frequency domain changes. Temporal integration cues are available only if the on- and offset times of the stimulus are sufficiently rapid relative to the room reverberation time such that distinct direct and reverberant portions of the stimulus are perceived. Therefore, in the first experiment we compared JNDs for signals that either did or did not contain temporal integration cues by varying on- and offset time. We propose that the frequency-based method relies on the standard deviation of the spectral response in a reverberant room, which increases monotonically from 0 dB to 5.57 dB (Schroeder limit) as the D/R energy ratio decreases from approx. 20 dB to 0 dB. Outside this range, the standard deviation asymptotes and thus should not offer any cues for discrimination. Therefore, in the second experiment we compared JNDs for stimuli with D/R energy ratios inside and outside the range of 0–20 dB. The results from both these experiments will provide evidence for the processes of source distance coding based on the D/R energy ratio by the auditory system.

2:00

2pPP3. Detection of high-frequency spectral notches as a function of level. Ana Alves-Pinto and Enrique A. Lopez-Poveda (Instituto de Neurociencias de Castilla y Leon, Universidad de Salamanca, Alfonso X El Sabio s/n, 37007 Salamanca, Spain)

In experiment I, the threshold depth for detecting a spectral notch centered at 8 kHz in an otherwise flat-spectrum noise was measured as a function of the noise spectrum level. Threshold depths increased gradually with spectrum level. This result is explained in terms of the broadening of cochlear filters with level and of the narrow dynamic range of most auditory nerve (AN) fibers. Experiment II investigated the extent that notch detection is affected by stimulus duration. On average, threshold depths were larger for shorter (25-ms) than for longer (220-ms) stimuli, particularly at higher sensation levels. This suggests that the adapted AN response is important for encoding spectral information, despite the fact that spectral features may be more clearly encoded at the onset, where AN fibers have a wider dynamic range. In experiment III, the steady spectral notch was replaced by a dynamic one whose center frequency changed abruptly after 110 ms from 7 to 9 kHz. Depths in this condition improved considerably at high levels. Therefore, at high levels, transient spectral information is detected more easily. This relates to the wide dynamic range of AN fibers for sudden increases in level. [Work supported by FIS PI020343, G03/203, and MCYT.]

2:15

2pPP4. A two level model of the masking property of human ear. Jack Xin (Dept. of Mathematics, Univ. of Texas, Austin, TX 78712, jxin@math.utexas.edu) and Yingyong Qi (Qualcomm, Inc., San Diego, CA 92121)

The absolute hearing threshold and masking are two fundamental phenomena in psychoacoustics. The former is the minimum intensity for the ear to detect sound at a given frequency in quiet. The latter is described by the nonlinearly raised threshold for the ear to detect sound in the vicinity of an existing signal or the masker. A two level model is developed to compute the latter given the former and the masker. The first level is a partial differential equation (PDE) model of the inner ear (cochlea), and the second level is a similarity transform, accounting for the functions of the remaining high level processes of audition. The model has a solid ground on first principles and is adaptive to nonlinearities when compared with existing data-driven empirical models. Modeled masking thresholds of banded noise by tonal signals agree well with existing hearing data. [Work partially supported by ARO Grant No. DAAD 19-00-1-0524 and NSF No. ITR-0219004.]

2:30

2pPP5. Spiral model of pitch. James D. Miller (Dept. of Speech and Hearing, Central Inst. for the Deaf and Washington Univ., 4560 Clayton Ave., St. Louis, MO 63110, jdmiller@artsci.wustl.edu)

A spiral model of pitch interrelates tone chroma, tone height, equal temperament scales, and a cochlear map. Donkin suggested in 1870 that the pitch of tones could be well represented by an equiangular spiral. More recently, the cylindrical helix has been popular for representing tone

chroma and tone height. Here it is shown that tone chroma, tone height, and cochlear position can be conveniently related to tone frequency via a planar spiral. For this "equal-temperament spiral," (ET Spiral) tone chroma is conceived as a circular array with semitones at 30° intervals. The frequency of sound on the cent scale (*re* 16.351 Hz) is represented by the radius of the spiral defined by $r=(1200/2\pi)\theta_r$, where θ_r is in radians. By these definitions, one revolution represents one octave, 1200 cents, 30° represents a semitone, the radius relates θ to cents in accordance with equal temperament (ET) tuning, and the arclength of the spiral matches the mapping of sound frequency to the basilar membrane. Thus, the ET Spiral gives tone chroma as θ, tone height as the cent scale, and the cochlear map as the arclength. The possible implications and directions for further work are discussed.

2:45

2pPP6. On perception of missing fundamental. Takahide Matsuoka and KeNichi Ito (Utsunomiya Univ., Utsunomiya 321-8585, Japan, matsuoka@cc.utsunomiya-u.ac.jp)

The frequency band of one channel of telephone line is from 300 Hz to 3400 Hz. Although the pitch frequency of speech is not in the frequency band, the frequency can be perceived through telephone. So, the psycho acoustic experiments of missing fundamental f0 were carried out at hearing a compound sound of f1 and f2, where $\mathrm{f1}=n\mathrm{f0}$, $\mathrm{f2}=(n+k)\mathrm{f0}$, n and k: natural numbers. It has been confirmed, by a significant difference test, that the missing fundamental can be heard. Then, it was investigated how the phase difference and amplitude difference between f1 and f2 influence the production of f0. The results are the following. The phase difference does not influence on the perception of f0. The amplitude difference in some range does not influence it. Those mechanisms have been made clear by consideration based on the experimental results using cochlea models. Now, the perception of missing fundamental for higher harmonics is being investigated.

3:00

2pPP7. Discrimination of dynamic frequency sounds and a multichannel detector model. Vivek Rajendran, Ashok K. Krishnamurthy (Dept. of Elec. Eng., Ohio State Univ., 2015 Neil Ave., Columbus, OH 43210, rajendrv@ee.eng.ohio-state.edu), and Lawrence L. Feth (Columbus, OH 43210)

A virtual frequency (VF) glide signal refers to the frequency transition perceived in an amplitude-modulated two-tone complex. For example, when the amplitude of one tone increases with duration, while that of the other decreases, the listener hears a rising pitch. This virtual glide is easily distinguished from a linear frequency-modulated (FM) glide because of envelope fluctuations in the virtual glide. They can be rendered more difficult to discriminate by extracting the envelope from the VF glide and imposing it on the FM glide. Discrimination of common envelope VF and FM glides was studied through an adaptive 2Q, 2-AFC task. VF and FM glides (250 ms) were centered on 500-, 1000-, 2000-, and 4000-kHz. Frequency sweeps began at approximately 1/7 octave and were adjusted using the 2 up, 1 Down rule (Levitt, 1971). Center frequency roved over a range of approximately 1/2 octave for each interval. We show that a multichannel detector model (Durlach, 1986) operating on intensity weighted average of instantaneous frequency (IWAIF) values computed by a short-term, multichannel IWAIF model is able to predict listener performance in the common-envelope VF versus FM glide discrimination task. The noise variance required by the model is derived from measured FMDL data.

3:15

2pPP8. Synthetic vowel categorization by hearing-impaired listeners. Michelle R. Molis and Marjorie R. Leek (Army Audiol. and Speech Ctr., Walter Reed Army Medical Ctr., 6900 Georgia Ave. NW, Washington, DC 20307-5001)

Vowel identification performance can be predicted reasonably well for normal-hearing (NH) listeners based on principal components derived from excitation patterns [M. R. Molis, J. Acoust. Soc. Am. **111**, 2433–2434 (2002)]. In this study, vowel categorization was measured in listeners with mild-to-moderate hearing loss. Stimuli were 54 synthesized vowels that varied orthogonally in $F2$ (1081–2120 Hz) and $F3$ (1268–2783 Hz) frequency in equal 0.8-bark steps. Fundamental frequency contour, $F1$ (455 Hz), $F4$ (3250 Hz), $F5$ (3700 Hz), and duration (225 ms) were held constant. Hearing-impaired (HI) listeners categorized the stimuli as the vowels /ɪ/, /ʊ/, or /ɝ/. Estimates of frequency resolution were also obtained, and excitation patterns were constructed for each listener. Categorization performance was more variable for HI listeners relative to NH listeners, both within and between listeners. Many subjects appeared to rely solely on $F2$ frequency and had particular difficulty with the /ʊ/ vs /ɝ/ distinction. Excitation patterns suggested a rather imprecise internal spectral representation of the stimuli. HI response patterns may reflect decreased audibility, spectral smearing of formant structure due to poor frequency resolution, or both. Models of vowel perception based on the impaired excitation patterns will be compared with formant-based models. [Work supported by NIH (DC00626).]

3:30

2pPP9. Dichotic presentation to overcome the effect of increased spectral masking and frequency dependent hearing threshold shifts in persons with bilateral sensorineural impairment. Alice N. Cheeran (BME Group, IIT Bombay, Powai Mumbai-400 076, India) and Prem C. Pandey (IIT Bombay, Powai Mumbai-400 076, India)

A binaural dichotic presentation scheme for reducing the effect of increased spectral masking in persons with bilateral sensorineural loss, using spectral splitting with complementary comb filters based on auditory critical bands, has been earlier reported [Cheeran *et al.*, J. Acoust. Soc. Am. **110**, 2705 (2001)]. The 256-coefficient linear phase FIR filters designed using frequency sampling technique had transition crossovers adjusted within −6 to −4 dB for perceptual balance, and had 78–117 Hz transition, 1 dB passband ripple, and 30 dB stopband attenuation. We evaluated the scheme by conducting listening tests on 5 normal hearing subjects with simulated loss, using a closed set identification of 12 vowel-consonant-vowel syllables. Based on significant improvement, further tests were conducted on 5 hearing-impaired persons with moderate bilateral sensorineural loss. Significant improvement in response time, recognition scores, and transmission of consonantal features, particularly place and duration, was obtained, indicating reduction in the effect of spectral masking. In order to partly compensate for frequency dependent hearing threshold shifts, a pair of filters with frequency response adjusted within a 6 dB range, based on the audiogram for the corresponding ear, was cascaded with the comb filters. These filters resulted in additional improvement, particularly for persons with relatively uniform loss.

3:45

2pPP10. The effect of compression and attention allocation on speech intelligibility. Sangsook Choi and Thomas Carrell (Univ. of Nebraska–Lincoln, 253 Barkley Memorial Ctr., Lincoln, NE 68583, schoi6@bigred.unl.edu)

Research investigating the effects of amplitude compression on speech intelligibility for individuals with sensorineural hearing loss has demonstrated contradictory results [Souza and Turner (1999)]. Because percent-correct measures may not be the best indicator of compression effectiveness, a speech intelligibility and motor coordination task was developed to provide data that may more thoroughly explain the perception of compressed speech signals. In the present study, a pursuit rotor task [Dlhopolsky (2000)] was employed along with word identification task to measure the amount of attention required to perceive compressed and noncompressed words in noise. Monosyllabic words were mixed with speech-shaped noise at a fixed signal-to-noise ratio and compressed using a wide dynamic range compression scheme. Participants with normal hearing identified each word with or without a simultaneous pursuit-rotor task. Also, participants completed the pursuit-rotor task without simultaneous word presentation. It was expected that the performance on the additional motor task would reflect effect of the compression better than simple word-accuracy measures. Results were complex. For example, in some

conditions an irrelevant task actually improved performance on a simultaneous listening task. This suggests there might be an optimal level of attention required for recognition of monosyllabic words.

4:00

2pPP11. A speech compression/expansion method based on subband filtering of the signal envelope. Aslak Bjerkvik, Asbjorn Krokstad, and Barbara Resch[a] (NTNU Acoust. group, Trondheim, Norway)

Compression algorithms are used in hearing aids for hearing-impaired listeners with recruitment, due to these listeners' loss of dynamic range. We propose a method for both compressing and expanding speech signals with the goal to improve speech intelligibility in reverberant conditions where the important rapid variations of the signal are inherently reduced. The method is based on dividing the envelope of the speech signal into two subbands with a division around modulation frequencies of 2 Hz, and compressing the signal based on the lower envelope subband, while expanding the signal based on the higher envelope subband. The sub-band division is accomplished by filtering the envelope of the signal and also by computing separate envelopes for each subband. A secondary goal is to develop low-delay algorithms. Various methods for calculating the envelope of the signal are evaluated, with a focus on the delay. [a] Currently at KTH, Speech processing group, Stockholm, Sweden.

4:15

2pPP12. Adaptive modeling of compression hearing aids: Convergence and tracking issues. Vijay Parsa and Donald Jamieson (Natl. Ctr. for Audiol., Elborn College, Univ. of Western Ontario, London, ON N6G 1H1, Canada, parsa@nca.uwo.ca)

Typical measurements of electroacoustic performance of hearing aids include frequency response, compression ratio, threshold and time constants, equivalent input noise, and total harmonic distortion. These measurements employ artificial test signals and do not relate well to perceptual indices of hearing aid performance. Speech-based electroacoustic measures provide means to quantify the real world performance of hearing aids and have been shown to correlate better with perceptual data. This paper investigates the application of system identification paradigm for deriving the speech-based measures, where the hearing aid is modeled as a linear time-varying system and its response to speech stimuli is predicted using a linear adaptive filter. The performance of three adaptive filtering algorithms, viz. the Least Mean Square (LMS), Normalized LMS, and the Affine Projection Algorithm (APA) was investigated using simulated and real digital hearing aids. In particular, the convergence and tracking behavior of these algorithms in modeling compression hearing aids was thoroughly investigated for a range of compression ratio and threshold parameters, and attack and release time constants. Our results show that the NLMS and APA algorithms are capable of modeling digital hearing aids under a variety of compression conditions, and are suitable for deriving speech-based metrics of hearing aid performance.

4:30

2pPP13. Evaluation of noise reduction techniques for digital hearing aids. Vijay Parsa and Karthikeyan Umapathy (Natl. Ctr. for Audiol., Elborn College, Univ. of Western Ontario, 1201 Western Rd., London, ON N6G 1H1, Canada, parsa@nca.uwo.ca)

Individuals with sensorineural hearing loss have increased difficulty in understanding speech in noisy backgrounds. To combat this issue, there has been a major thrust in recent years toward the development of noise reduction algorithms. The goals of this paper are to quantify the relative benefits of different single-microphone noise reduction algorithms, and to investigate the interaction between the noise reduction and dynamic range compression algorithms. Noise reduction techniques evaluated in this paper include spectral subtraction-based techniques, a wavelet-packet-based technique and a matching pursuit-based technique. All algorithms were tested with HINT signals with SNR levels ranging from -5 to 15 dB, and two different noise types viz. the speech-shaped noise and multi-talker babble. Performance was quantified using the ITU standardized PESQ measure which computes the perceptual similarity between the enhanced signal and the original signal. Initial PESQ results showed that the spectral subtraction-based techniques perform superior to that of the wavelet-packet and matching pursuit-based approaches and that the compression time constants have an impact on the overall performance. Perceptual data collected from hearing impaired listeners on sound quality and noise reduction performance will be presented and their correlation with the objective measurements will be discussed.

4:45

2pPP14. Perception of synthetic speech in quiet and in noise by adults with cochlear implants. Benjamin Munson and Peggy B. Nelson (Dept. of Commun. Disord., Univ. of Minnesota, 115 Shevlin Hall, 165 Pillsbury Dr. SE, Minneapolis, MN 55455)

Recent research has demonstrated that the ability of adults with cochlear implants (CI) to perceive speech in noise is highly variable. There are many potential reasons why this may be so, including poor signal/noise segregation, and poor perception of phonetic features in noise. We examined the CI listeners' perception of four synthetic speech continua in quiet and in noise. In two continua, dynamic spectral information was manipulated (/ra/-/la/ and /wa/-/ja/); in one, static spectral information was manipulated (/i/-/u/); and in one, only temporal information was manipulated (/sei/-/stei/). These were presented to 12 listeners with CIs and 13 normal-hearing listeners in quiet and concurrent with speech-shaped noise, at a $+10$ dB SNR. Noise affected identification (endpoints, boundaries, and slopes) of CI listeners more than NH listeners. Significant group-by-SNR interactions were found for endpoint identification of the /ra/-/la/, /wa/-/ja/, and /sei/-/stei/ continua. CI listeners had significantly shallower identification slopes for the /ra/-/la/ continuum; this was exaggerated in the $+10$ dB SNR condition. In addition, CI listeners showed more /u/ percepts than the NH listeners in the /i/-/u/ continuum at $+10$ dB SNR. Results are discussed with respect to the relative vulnerability of temporal and spectral features to misperception in noisy conditions.

5:00

2pPP15. Cochlear nonlinearity between 500 and 8000 Hz in listeners with impaired hearing. Enrique A. Lopez-Poveda (Instituto de Neurociencias de Castilla y Leon, Universidad de Salamanca, Alfonso X El Sabio s/n, 37007 Salamanca, Spain), Christopher J. Plack, Ray Meddis (Dept. of Psych., Univ. of Essex, Colchester CO4 3SQ, UK), and Jose L. Blanco (Oticon Spain SA, Ctra. Fuencarral 24, Ed. Europa, 28108 Madrid, Spain)

Cochlear nonlinearity was estimated in listeners with impaired hearing using a forward-masking method. For a fixed low-level probe, the masker level required to mask the probe was measured as a function of the masker-probe interval, to produce a temporal masking curve (TMC). TMCs were measured for probe frequencies from 500 to 8000 Hz, and for masker frequencies from 0.5 to 1.6 times the probe frequency. Unlike what happens for normal-hearing (NH) listeners [Lopez-Poveda *et al.*, J. Acoust. Soc. Am. **113**, 951–960 (2003)], TMCs for on-frequency maskers sometimes show a single slope, suggesting linear responses for tones at CF. Sometimes, however, two distinct slopes are visible, suggesting remaining compression. Both patterns are uncorrelated with the absolute threshold and are consistent with selective damage to outer or inner hair cells (IHC), respectively. Remarkably, the slope of the TMCs for very low-frequency maskers is shallower for the impaired ears than for normal ones. This result implies that for NH listeners, the slope of the TMCs for very low-frequency maskers reflects some kind of compression, even at high CFs. It is discussed that this compression is likely to occur at the IHC rather than at the basilar membrane. [Work supported by FIS PI020343, G03/203, and Oticon Spain.]

5:15

2pPP16. Vowel formant discrimination in high-fidelity speech by hearing-impaired listeners. Diane Kewley-Port (Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405, kewley@indiana.edu), Chang Liu (Univ. at Buffalo, Buffalo, NY 14214), and T. Zachary Burkle (Indiana Univ., Bloomington, IN 47405)

The ability to discriminate differences in vowel formant frequency under a more ordinary listening condition has been reported recently by Kewley-Port and Zheng (1999) for synthetic speech and Liu and Kewley-Port (2003) for high-fidelity speech. Results for normal-hearing (YNH) listeners showed that a longer phonetic context (sentences versus words) degraded formant discrimination while adding a sentence identification task did not. The present study used the same high-fidelity stimuli but employed listeners with moderate hearing impairment (YHI). Experimental factors manipulated included phonetic context (isolated vowels, words and sentences), level [70-dB SPL (partially audible) and 95-dB SPL (fully audible)], and task (discrimination with and without the sentence identification task). Thresholds for F1 and F2 of the vowels /I, ε, æ ʌ/ were estimated using adaptive tracking for 71% correct discrimination. The anticipated degrading effects of higher formant frequency and longer phonetic context were obtained. Unexpectedly formant thresholds for high-fidelity vowels at the higher (95-dB SPL) were slightly elevated compared to the 70-dB SPL level, the opposite of results for synthetic vowels. No effect of the added sentence task was seen, similar to the YNH listeners. Details of the differences in vowel processing attributable to moderate hearing impairment will be discussed. [Research supported by NIH-NIDCD.]

TUESDAY AFTERNOON, 11 NOVEMBER 2003 NUECES ROOM, 2:00 TO 4:45 P.M.

Session 2pSA

Structural Acoustics and Vibration, Engineering Acoustics and Physical Acoustics: Measurements of Particle Velocities

Sean F. Wu, Chair

Department of Mechanical Engineering, Wayne State University, 5050 Anthony Wayne Drive, Detroit, Michigan 48202

Invited Papers

2:00

2pSA1. Measurement of acoustic and streaming velocities in fluids using laser Doppler anemometry. Michael W. Thompson and Anthony A. Atchley (Grad. Prog. in Acoust., The Penn State Univ., University Park, PA 16802, mwt126@psu.edu)

The optical technique of laser Doppler anemometry (LDA) can be used to measure both acoustic and streaming velocities in a fluid. In a typical LDA system, two coherent laser beams intersect within the fluid to form a series of interference fringes. Whenever a small tracer particle suspended in the fluid passes through these fringes, light is scattered with an intensity that fluctuates at a frequency proportional to the particle velocity. This intensity is measured using a photodetector, and the resultant signal is processed to yield the fluid velocity. An overview of several signal processing methods will be given, and the inherent strengths and weaknesses of the LDA technique will be discussed. The authors have used LDA with burst spectrum analysis and recursive Fourier averaging to measure standing-wave acoustic velocities in the range of 0.02–15 m/s, and to measure Rayleigh streaming velocities in the range of 0.1–20 cm/s in the presence of standing-wave acoustic velocities that were 2–3 orders of magnitude larger. [Work supported by ONR.]

2:30

2pSA2. Particle image velocimetry measurements in thermoacoustic refrigerators. Philippe Blanc-Benon (Ecole Centrale de Lyon, Ctr. Acoustique, LMFA UMR CNRS 5509, 69134 Ecully Cedex, France)

The knowledge of flow fields in the microchannels and at the edges of the stack plates becomes an increasingly important issue in the design of heat exchangers for thermoacoustic engines. We have developed numerical simulations and conducted experiments in a resonant standing wave thermoacoustic refrigerator model. Recent computational evidence indicates that near the edges of the plates the flow field is dominated by concentrated eddies, whose complex motion significantly affects the performance of the device. Consequently, the effective design and optimization of thermoacoustic refrigerators necessitates a fundamental understanding of these vortical motions, and their dependence on geometric parameters and operating conditions. We present experimental data obtained using Particle Image Velocimetry: velocity profiles across the microchannels, 2D velocity maps including a zoom for the edges of the stack, and vorticity fields calculated with a criterion based on a normalized angular momentum. Results are obtained for two distinct configurations, involving thin and thick stack plates. In the first case, the flow field around the edge of the stack exhibits elongated vorticity layers, while in the latter it is dominated by the shedding and impingement of concentrated vortices. Time-resolved PIV measurements of the velocity and vorticity fields are compared with computational predictions.

3:00

2pSA3. Flow-induced noise on a pressure-acceleration underwater acoustic intensity probe. James A. McConnell (Acoustech Corp., P.O. Box 139, State College, PA 16804) and Gerald C. Lauchle (Grad. Prog. in Acoust., Penn State Univ., 218 Appl. Sci. Bldg., University Park, PA 16802)

The results of an experiment to determine the effect of flow-induced noise on a pressure-acceleration intensity probe are presented. The sensor is a bluff body that conforms to the geometry of a right circular cylinder having a diameter of 10.16 cm and an aspect ratio of unity. A flexural mode hydrophone is used to measure the acoustic pressure and a flexural mode accelerometer is used to measure the acoustic particle acceleration. The hydrophone has a nominal sensitivity of about −185 dB *re*: 1 V/uPa and the accelerometer has an in-water acoustic sensitivity of nearly 1 V/g. The bandwidth of the device covers the 10-Hz to 1-kHz frequency range and the principle axis of sensitivity is coincident with the axis of the cylinder. The sensor was towed in cross-flow at speeds ranging from 0.1 to 0.5 cm/s and data were collected over the 10-Hz to 100-Hz frequency range. Of particular interest is the decomposition of the intensity spectrum into real and reactive parts along with an assessment of any filtering that can be accomplished using the intensity technique to measure far-field sound in the presence of near-field noise. [Work supported by various grants from ONR and NAVAIR.]

3:30–3:45 Break

3:45

2pSA4. Measurements in acoustic boundary layers. G. Huelsz, A. A. Castrejon-Pita, J. R. Castrejon-Pita, F. Lopez-Alquicira, E. Ramos, and R. Tovar (Centro de Investigacion en Energia, UNAM, Mexico)

A summary of the results from experimental measurements of temperature and velocity in the oscillatory boundary layers produced by acoustic waves (acoustic boundary layers) are presented. Temperature measurements were done using the named cold wire anemometer and velocity measurements were done using hot wire and laser Doppler anemometers and particle image velocimetry. Experimental results are compared to theoretical results of a linear theory. The advantages and the limitations of these experimental techniques are discussed, especially for their use in acoustic boundary layers. [Partial support for this work has been provided by DGAPA-UNAM IN104702-2 and CONACYT 32707-U projects.]

Contributed Papers

4:15

2pSA5. A laser-based acoustic velocity sensor. Peter R. Stepanishen (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882-1197) and Lynn Antonelli (Naval Undersea Warfare Ctr., Newport, RI 02841)

A linear systems model is presented for a Laser Doppler Vibrometer (LDV), which is used to detect the acoustically induced normal surface velocity at an air-water interface. Previous acoustic test tank results for signal reception are interpreted using the model. Noise performance is addressed using the model for a commercially available LDV where thermal noise in the fluid is shown to be of lesser importance than the internal noise of the sensor. In addition to the internal sensor noise that can be viewed as an additive noise which is uncorrelated to the signal, it is also noted that sidebands of the signal spectrum can be generated for large displacements of the air-water interface. For the case of harmonic signals the spectral distortion increases as kY increases where k is the acoustic wavenumber and Y is the amplitude of the surface displacement. Results from a series of experiments in-air and in-water using a piezo-ceramic disk with a variable voltage excitation are discussed in light of the model. The in-air experiments use the vibrating face of the disk as the reflecting surface whereas the in-water experiments use the air/water interface in a wave tube excited by the disk.

4:30

2pSA6. Prediction of acoustic radiation via 3D particle velocity measurements. Zhi Ni, Huancai Lu, Sean Wu (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI 48202), and Yang Zhao (Dept. of Elec. & Computer Eng., Wayne State Univ., Detroit, MI 48202)

It has been shown [Wu and Hu, J. Acoust. Soc. Am. **103**, 1763–1774 (1998)] that one can use an explicit integral formulation to predict acoustic radiation from a vibrating object based on a particle velocity distribution over a hypothetical surface enclosing this object. The advantages of this integral formulation are that solutions thus obtained are unique and the efficiency of numerical computations is high. The challenge is how to acquire 3D particle velocities when measurements are taking place in the air, in which seeding particles are hard to control and the signal to noise ratio is low. Here we use laser Doppler Anemometry (LDA) to accomplish this task. Two LDA probes are utilized to measure two components of the particle velocities simultaneously, and the third component is obtained by rotating one probe 90° and measuring again. A traverse system is adopted to sweep the LDA probes over a measurement plane to enhance efficiency. Once particle velocities are specified on all nodes of the enclosing measurement surface, the sound pressure anywhere in an exterior region can be predicted. An interface program for commercial hypermesh software is developed, the Dijkstra algorithm and boundary element method are adopted, and error analyses are presented. [Work supported by NSF.]

Session 2pSCa

Speech Communication: Cross Linguistic Issues (Poster Session)

Katsura Aoyama, Chair

Communication Disorders, Texas Tech University Health Science Center, 3601 4th Street, Lubbock, Texas 79430-6073

Contributed Papers

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and contributors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

2pSCa1. Cross-linguistic preference and the phonetics of geminates: Place of articulation and duration of phonologically short and long consonants in Guinaang Bontok. Katsura Aoyama (Dept. of Speech-Lang. & Hearing Sci., Texas Tech Univ., Health Sci. Ctr., 3601 4th St., Stop 6073, Lubbock, TX 79430-6073, katsura.aoyama@ttuhsc.edu)

It is reported that there is a preference for the place-of-articulation in geminates among world languages; alveolar, labial, velar geminate consonants always appear in that order with respect to one another (G. Thurgood, *Papers in Honor of Frederick H. Brengelman*, 1993, pp. 129–137). This study investigated whether the phonetic contrasts in duration are larger in cross-linguistically favored place-of-articulation (e.g., alveolar) than in the less favored place-of-articulation (e.g., velar). The data is from a language called Guinaang Bontok. Four speakers of Guinaang Bontok produced thirty-five words that have either a singleton or a geminate of /p t k m n ŋ/. Eighteen words included a singleton and the other seventeen words included a geminate. The participants produced the target word in isolation first, and then repeated it in a frame sentence twice. The recordings were digitized at 22.05 Hz, and the durations of the consonants were measured. A total of 420 tokens (35 words $\times 3$ repetitions $\times 4$ participants) were analyzed. The results partially supported the hypothesis; the durational contrasts were indeed larger in alveolar consonants (/t n/) than in labial consonants (/p m/) and velar consonants (/ŋ/). However, contrasts in velar stop (/k/) were as large as in the alveolar stop (/t/).

2pSCa2. Contrast enhancement of vowels in Modern Standard German. Amy E. Coren and Cheryl L. Heckmann (Dept. of Psych., Univ. of Texas, 1 University Station A8000, Austin, TX 78712-0187)

In an earlier cross-language study of vowels in and out of utterance focus [Hay *et al.*, J. Acoust. Soc. Am. **111**, 2367 (2002)], it was reported that languages tend to make differential use of three possible means of contrast enhancement: an enlarged vowel space, greater use of vowel inherent spectral change (VISC), and greater systematic variation in duration across vowel categories. In the current study, these same means of vowel contrast enhancement were examined in Modern Standard German. Relative to speakers of American English, German speakers make less use of VISC and greater use of durational variation to enhance vowel distinctions. It appears that phonological properties of a language (e.g., the use of phonemic vowel length differences) are predictive of the particular means used by the language to enhance vowel contrasts in the utterance focus. [Work supported by NIDCD.]

2pSCa3. Perceptual illusion overrides reality: A phoneme detection study of epenthetic vowels in Japanese. Yuki Hirose (The Univ. of Electro-Commun., 1-5-1 Chofugaoka, Chofu 182-8585, Japan, hirose@cs.uec.ac.jp)

Vowel epenthesis is a known phenomenon in Japanese where speakers insert a vowel ([u], by default) inside consonant clusters (e.g., in a VCCV sequence), due to the phonotactic constraint of the language. Dupoux *et al.* (1999) argue that vowel epenthesis occurs at a perceptual level. In their study, Japanese listeners perceived the epenthetic vowel [u] when the interconsonantal vowel [u] in the original stimuli was removed. In the present study, we investigate further whether (1) the perceptual system in Japanese listeners that causes vowel epenthesis is not only constrained by the phonotactic constraint (*VCCV) but also by the quality of the vowel that can be inserted, and (2) whether this constraint overrides existing acoustic information present in the input, e.g., the formant information of some other vowel. In our phoneme-detection experiment, we recorded a set of nonword stimuli (VCVCV) and gradually removed the interconsonantal vowel ([u] or other vowels). Some Japanese listeners perceived [u] not only when the originally present [u] was removed (as found in Dupoux *et al.*), but also when a vowel other than [u] was substituted in the interconsonantal position and that substituted vowel was partially removed, leaving only some remnant information.

2pSCa4. Learning to perceive Mandarin tones: The role of acoustic information. Connie K. So (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC V5A 1S6, Canada, klso@sfu.ca)

Studies show that auditory-training improves non-native listeners' tonal identification; however, persistently perceptual confusion of several tone pairs is still observed. This may imply that learners have not yet mastered/acquired the lexical tones. Their confusion may be reduced further if the essential acoustic information of tones (e.g., duration and pitch contour) could be implemented and emphasized during training. To verify the assumption, the present study examines the impact of employing acoustic information of lexical tones as feedback on non-native listeners' performance during a computer-based perception training of Mandarin tones. Non-native speakers of Mandarin were randomly assigned to one of two groups. Listeners in the control group were merely shown that the answer was right or wrong. In contrast, those in the experimental group received acoustic information by means of both visual and auditory feedback when the response was incorrect (i.e., showing pitch graphs and presenting the audio files for the contrastive tonal pairs). Results indicated a significant improvement in the tonal identifications for listeners who received detailed acoustic information during training. This suggests that

training with acoustic information of lexical tones assists non-native listeners in distinguishing the tone pairs more effectively. [Work supported by SSHRC.]

2pSCa5. Word-final nasals in Romanian. Benjamin Tucker (Univ. of Arizona, 1100 E. University Blvd., Tucson, AZ 85715)

In this study Romanian word-final nasals, e.g., /basm/ "fairytale," are examined acoustically. Acoustic descriptions of Romanian are few, but early research indicated that word-final nasals are devoiced or partially devoiced. In this study an acoustic analysis of cross-linguistically unusual voiceless nasals, as well as information about a language which has not been phonetically well-documented is provided. In the current work the word-final /sm, mn, Vm, Vn/ are examined. The words were placed in various phonological environments (e.g., utterance final, before vowel-initial following word, etc.). Eight Romanian speakers were recorded. A judgment of voicing was made from the spectrogram. The duration of the gap between the /s/ and /m/, the total duration of voiced nasals and the duration of the preceding vowel were measured. The data was also analyzed for the presence of a release noise at the end of the nasal, which indicates that there is an oral closure during the nasal, and hence that the nasal is devoiced rather than deleted. Preliminary data shows that the average duration for /mn/ (0.148 ms) is longer than the duration of the /n/ or /m/ (0.105 ms) further indicating that deletion is not occurring. The preliminary data also shows that 93% of nasals in the /sm/ sequences are devoiced.

2pSCa6. Acoustic comparisons of Japanese and English vowels produced by native speakers of Japanese. Kanae Nishi (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405), Reiko Akahane-Yamada, Rieko Kubo (ATR Intl.-Human Information Sci. Labs., Kyoto 619-0288, Japan), and Winifred Strange (City Univ. of New York–Grad. Ctr., New York, NY 10016)

This study explored acoustic similarities/differences between Japanese (J) and American English (AE) vowels produced by native J speakers and compared production patterns to their perceptual assimilation of AE vowels [Strange *et al.*, J. Phonetics **26**, 311–344 (1998)]. Eight male native J speakers who had served as listeners in Strange *et al.* produced 18 Japanese (J) vowels (5 long-short pairs, 2 double vowels, and 3 long-short palatalized pairs) and 11 American English (AE) vowels in /hVbopena/ disyllables embedded in a carrier sentence. Acoustical parameters included formant frequencies at syllable midpoint ($F1/F2/F3$), formant change from 25% to 75% points in syllable (formant change), and vocalic duration. Results of linear discriminant analyses showed rather poor acoustic differentiation of J vowel categories when $F1/F2/F3$ served as input variables (60% correct classification), which greatly improved when duration and formant change were added. In contrast, correct classification of J speakers' AE vowels using $F1/F2/F3$ was very poor (66%) and did not improve much when duration and dynamic information were added. J speakers used duration to differentiate long/short AE vowel contrasts except for mid-to-low back vowels; these vowels were perceptually assimilated to a single Japanese vowel, and are very difficult for Japanese listeners to identify.

2pSCa7. Discriminability and identification of English vowels by native Japanese speakers in different consonantal contexts. Takeshi Nozawa (Kansai Univ. of Intl. Studies, 1-18 Aoyama Shijimi-cho Miki Hyogo 673-0521, Japan, nozawa@kuins.ac.jp), Elaina M. Frieda (Auburn Univ., Auburn, AL), and Ratree Wayland (Univ. of Florida–Gainesville, Gainesville, FL)

The purpose of the present experiment was to examine the effects of consonantal context on discrimination and identification of English vowels by native Japanese speakers learning English in Japan. A number of studies have assessed the effects of consonantal contexts on the perception of nonnative vowels. For instance, Strange *et al.* (1996, 2001) found that perceptual assimilation of nonnative vowels is affected by consonantal contexts, and Morrison (2002) has shown that Japanese speakers use durational cues to perceive English /i/–/ɪ/. The present study revealed that consonantal context affects discriminability and identification of each English vowel differently. Of all the six vowel contrasts tested, /i/–/ɪ/ was the most likely to be affected by voicing status of the surrounding consonants with it being easier to discriminate in voiceless consonantal contexts. Moreover, /ɪ/ is more likely to be equated with the Japanese short vowel /i/ in a voiceless consonantal context which is in keeping with Morrison (2002). /æ/–/ɑ/, on the other hand, is the most strongly affected by the place of articulation of the preceding consonants. [Work supported by Grant-in-Aid for Scientific Research (C)(1)(1410635).]

2pSCa8. Accent, intelligibility, and comprehensibility in the perception of foreign-accented Lombard speech. Chi-nin Li (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC V5A 1S6, Canada, clia@sfu.ca)

Speech produced in noise (Lombard speech) has been reported to be more intelligible than speech produced in quiet (normal speech). This study examined the perception of non-native Lombard speech in terms of intelligibility, comprehensibility, and degree of foreign accent. Twelve Cantonese speakers and a comparison group of English speakers read simple true and false English statements in quiet and in 70 dB of masking noise. Lombard and normal utterances were mixed with noise at a constant signal-to-noise ratio, and presented along with noise-free stimuli to eight new English listeners who provided transcription scores, comprehensibility ratings, and accent ratings. Analyses showed that, as expected, utterances presented in noise were less well perceived than were noise-free sentences, and that the Cantonese speakers' productions were more accented, but less intelligible and less comprehensible than those of the English speakers. For both groups of speakers, the Lombard sentences were correctly transcribed more often than their normal utterances in noisy conditions. However, the Cantonese-accented Lombard sentences were not rated as easier to understand than was the normal speech in all conditions. The assigned accent ratings were similar throughout all listening conditions. Implications of these findings will be discussed.

2pSCa9. Intonational phonology description of Porteno Spanish. John P. Barjam (Dept. of Linguist., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095-1543, ungaucho@humnet.ucla.edu)

In this study the intonational patterns of Porteno Spanish (PS) are described within an Autosegmental Metrical (AM) (Ladd, 1996) approach. Porteno Spanish is the Spanish spoken in Buenos Aires, Argentina, and because of the influence from Italian and other languages, it differs markedly from varieties of Spanish that have been previously described. Six monolingual PS speakers were recorded saying 99 sentences which include declaratives, interrogatives, imperatives, and focused declarative and interrogative sentences. A ToBI framework, based on the F0 contour, was developed for labeling the sentences intonation. Preliminary results suggest that each phrase contains a somewhat regular pattern: (1) the stressed syllables of content words carry one of five pre-nuclear pitch accents, (2) nuclear pitch accents are more restrictive and tend to vary depending on the sentence type, and (3) boundary tones, both intermediate and intonational, are regular but also vary depending on sentence type.

2pSCa10. Age effects on acquisition of word stress in Spanish–English bilinguals. Susan G. Guion, J. J. Clark (Dept. of Linguist., Univ. of Oregon, Eugene, OR 97403-1290, guion@uoregon.edu), and Tetsuo Harada (Univ. of Oregon)

Based on studies of syntactic and semantic learning, it has been proposed that certain aspects of second language learning may be more adversely affected by delays in language learning than others. Here, this proposal is extended to the phonological domain in which the acquisition of English word stress patterns by early (AOA <6 years) and late (AOA >14 years) Spanish–English bilinguals is investigated. The knowledge of English word stress was investigated by three behavioral tasks. In a production task, participants produced two syllable nonwords in both noun and verb sentence frames. In a perception task, participants indicated a preference for first or last syllable stress on the nonwords. Real words that were phonologically similar to the test items were also collected from each participant. Regression analyses and ANOVAs were conducted to determine the effect of syllable structure, lexical class, and stress pattern of phonologically similar words on the data from the production and perception tasks. Early bilinguals patterned similarly to the native English participants. Late bilinguals showed little evidence of learning prosodically based stress patterns but did show evidence of application of distributional patterns based on lexical class and analogy in stress assignment. [Research supported by NIH.]

2pSCa11. The effect of the dialect on perception of words with different pitch accents in Japanese. Kyoko Okamura (Dept. of Linguist., Indiana Univ., Memorial Hall 322, 1021 E. 3rd St., Bloomington, IN 47405)

Eastern and western dialects of Japanese have been noted to have different pitch accent systems. This study explores how Japanese words with and without accent are perceived by speakers of different dialects. Otake and Cutler (J. Phonet. **27**, 229–253) found little difference in identification between Tokyo and the other variety of speakers. In current study, a speaker of Tokyo Japanese produced words with and without an accent. Using the STRAIGHT resynthesis system, the $F0$ of the accented mora was lowered in 7 steps to the level of the unaccented form, and raised in the unaccented mora to the level of the accented form. Continua were presented to listeners from western and eastern Japan. Listeners exhibit consistent category boundaries according to the difference in F0 between the target mora and the following one. In addition, reaction times are greater near the category boundary, and subjective measures of confidence decrease near the boundary. Speakers from different dialects showed similar identification patterns. Reaction times and confidence measures were different overall, but exhibited the same pattern for the two groups. These results indicate that the accent is a uniform phonetic entity for the two dialects, despite numerous differences in the accentual systems.

2pSCa12. Is there a correlation between Japanese $L2$ learner's perception of English stressed words and acoustic features? Keiko Asano and Toshiko Isei-Jakkola (Yokohama Natl. Univ., 79-1 Tokiwadai Hodogaya-ku Yokohama-city, Kanagawa-prefecture 240-8501, Japan, ll-ed@ynu.ac.jp)

Is there a correlation between Japanese $L2$ learner's perception of English stressed words and acoustic features? [Keiko Asano (Yokohama National University, ll-ed@ynu.ac.jp) and Toshiko Isei-jaakkola (University of Helsinki)]. It is well known that the Japanese have weakness in listening to unstressed words in English, but there are less data on their perception of stressed words. Thus, the listening tests and the acoustic experiments were conducted in terms of (1) relevancy of difficulites depending on part of speech and their English proficiency, (2) the relationship between pitch and intensity of stressed words, and (3) if there is a correlation between their perception and experimental data. In the listening test, an English prose read by an American male speaker was used. The 150 Japanese $L2$ learners were assigned to mark the primary stressed words. The statistical results showed that there was a variance depending on part of speech and more markedly the comparative rating scores of correct words were highly correlated to the learner's English proficiency in any part of speech. In the acoustic experiments, pitch and intensity were measured. It was confirmed that (1) both $F0$ and dB carried the cue to perceive a stressed-word but they were not necessarily correlated, and (2) the relationship between $F0$ and dB might be compared only by relative movement. By further analyzing these acoustic data, prosodic combination of $F0$ and dB might be relevant to the correct ratios of part of speech.

2pSCa13. A note on the acoustic-phonetic characteristics of nonnative English vowels produced in noise. Chi-nin Li and Murray J. Munro (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC V5A 1S6, Canada)

The Lombard reflex occurs when people unconsciously raise their vocal levels in the presence of loud background noise. Previous work has established that utterances produced in noisy environments exhibit increases in vowel duration and fundamental frequency ($F0$), and a shift in formant center frequencies for $F1$ and $F2$. Most studies of the Lombard reflex have been conducted with native speakers; research with second-language speakers is much less common. The present study examined the effects of the Lombard reflex on foreign-accented English vowel productions. Seven female Cantonese speakers and a comparison group of English speakers were recorded producing three vowels (/i u a/) in /bVt/ context in quiet and in 70 dB of masking noise. Vowel durations, $F0$, and the first two formants for each of the three vowels were measured. Analyses revealed that vowel durations and $F0$ were greater in the vowels produced in noise than those produced in quiet in most cases. First formants, but not $F2$, were consistently higher in Lombard speech than in normal speech. The findings suggest that non-native English speakers exhibit acoustic-phonetic patterns similar to those of native speakers when producing English vowels in noisy conditions.

2pSCa14. Allophonic variation of Japanese /z/. Yuka Matsugu (Dept. of East Asian Studies, Univ. of Arizona, P.O. Box 210105, Tucson, AZ 85721-0105) and Timothy J. Vance (Univ. of Arizona, Tucson, AZ 85721)

According to impressionistic descriptions, the Japanese phoneme /z/ is pronounced sometimes as a fricative [z] and sometimes as an affricate [dz]. There is no consensus about how the two allophones are distributed, but one claim is that they are in complementary distribution: [dz] word initially or after a nasal and [z] intervocalically. Native speakers of "standard" Japanese pronounced words containing /z/ in a carrier sentence. Using spectrograms and waveforms, two (necessarily approximate) measurements were taken for each token of /z/: total duration and closure duration. The results did not show a categorical affricate/fricative distinction. Some word-initial and post-nasal tokens had no measurable closure, while many intervocalic tokens had clear closure. Nonetheless, the average closure duration was significantly longer for word-initial and post-nasal tokens of /z/ than for intervocalic tokens, and a much higher proportion of intervocalic tokens had zero closure. These results partially support the distributional claim above but make it clear that the allophonic variation involves two statistically different ranges of realizations rather than a categorical distinction between two discrete alternatives.

Session 2pSCb

Speech Communication and ASA Committee on Standards: Speech Intelligibility and AAC Devices

Fredericka Bell-Berti, Cochair
Speech Communication Sciences and Theater, St. John's University, 8000 Utopia Parkway, Jamaica, New York 11439

Susan B. Blaeser, Cochair
Acoustical Society of America Standards Secretariat, 35 Pinelawn Road, Suite 114E, Melville, New York 11747

Chair's Introduction—2:00

Invited Papers

2:05

2pSCb1. Advancements in text-to-speech technology and implications for AAC applications. Ann K. Syrdal (AT&T Labs—Res., 180 Park Ave., Rm. D159, Florham Park, NJ 07932-0971)

Intelligibility was the initial focus in text-to-speech (TTS) research, since it is clearly a necessary condition for the application of the technology. Sufficiently high intelligibility (approximating human speech) has been achieved in the last decade by the better formant-based and concatenative TTS systems. This led to commercially available TTS systems for highly motivated users, particularly the blind and vocally impaired. Some unnatural qualities of TTS were exploited by these users, such as very fast speaking rates and altered pitch ranges for flagging relevant information. Recently, the focus in TTS research has turned to improving naturalness, so that synthetic speech sounds more human and less robotic. Unit selection approaches to concatenative synthesis have dramatically improved TTS quality, although at the cost of larger and more complex systems. This advancement in naturalness has made TTS technology more acceptable to the general public. The vocally impaired appreciate a more natural voice with which to represent themselves when communicating with others. Unit selection TTS does not achieve such high speaking rates as the earlier TTS systems, however, which is a disadvantage to some AAC device users. An important new research emphasis is to improve and increase the range of emotional expressiveness of TTS.

2:35

2pSCb2. Assessment of variation between and within speakers. Ruth Huntley Bahr (Dept. of Commun. Sci. Disord., 4202 E. Fowler Ave., PCD 1017, Univ. of South Florida, Tampa, FL 33620)

While few individuals would argue that vocal cues can signal a person's identity, it is difficult to specify exactly which parameter(s) provide the most salient information for speaker identification. Previous literature has suggested that speaking fundamental frequency, long-term spectra, vowel formant frequencies, and speech tempo can provide speaker-specific information. However, investigations focused on automatic speaker identification have provided less than satisfactory results. These findings could be related to how each acoustic parameter is measured or, more probably, to the idea that these acoustic parameters interact in specific ways that may be more obvious in the perceptual realm and may vary across speaking situations. To further complicate matters, individuals may speak more than one language or use multiple dialects. Little is known about the effect of code switching on voice production and identification. The purpose of this presentation is to present some of the relevant literature on voice recognition and factors related to misidentification. The role of intraspeaker variability will be discussed with a special emphasis on bilingualism and bidialectalism. Implications for voice production in augmentative and alternative communication devices will be described.

3:05

2pSCb3. The effects of four variables on the intelligibility of synthesized sentences. Carol Conroy (New York City Board of Education, Technol. Solutions, District 75, 400 First Ave., New York, NY 10010 and Dept. of Speech, Commun. Sci. & Theatre, St. John's Univ., Jamaica, NY 11439), Lawrence J. Raphael (Adelphi Univ., Garden City, NY 11530), and Fredericka Bell-Berti (St. John's Univ., Jamaica, NY 11439)

The experiments reported here examined the effects of four variables on the intelligibilty of synthetic speech: (1) listener age, (2) listener experience, (3) speech rate, and (4) the presence versus absence of interword pauses. The stimuli, eighty IEEE–Harvard Sentences, were generated by a DynaVox augmentative/alternative communication device equipped with a DECtalk synthesizer. The sentences were presented to four groups of 12 listeners each (children (9–11 years), teens (14–16 years), young adults (20–25 years), and adults (38–45 years). In the first experiment the sentences were heard at four rates: 105, 135, 165, and 195 wpm; in the second experiment half of the sentences (presented at two rates: 135 and 165 wpm), contained 250 ms interword pauses. Conditions in both experiments were counterbalanced and no sentence was presented twice. Results indicated a consistent decrease in error rates with increased exposure to the synthesized speech for all age groups. Error rates also varied inversely with listener age. Effects of rate variation were inconsistent across listener groups and between experiments. The presences versus absences of pauses affected listener groups differently: The youngest listeners had higher error rates, and the older listeners lower error rates when interword pauses were included in the stimuli. [Work supported by St. John's University and New York City Board of Education, Technology Solutions, District 75.]

Session 2pSPa

Signal Processing in Acoustics: Time Reversal and Array Processing

Geoffrey S. Edelson, Chair

BAE Systems, P.O. Box 868, Nashua, New Hampshire 03061-0868

Contributed Papers

1:00

2pSPa1. Time-reversal communications in a hostile, reverberative environment. James V. Candy, Alan W. Meyer, Andrew J. Poggio, and Brian L. Guidry (Univ. of California, Lawrence Livermore Natl. Lab., P.O. Box 808, L-156, Livermore, CA 94551)

Time-reversal (T/R) communications is a new application area motivated by the recent advances in T/R theory. T/R receivers offer an interesting solution to the communications problem for a reverberant channel. This paper describes the performance of various realizations of the T/R receiver for an acoustic communications experiment in air. An experiment is developed to evaluate the performance of point-to-point T/R receivers designed to extract a transmitted information sequence propagating in a hostile, highly reverberant environment. It is demonstrated that T/R receivers are capable of extracting the transmitted coded sequence from noisy microphone sensor measurements with reasonable success. The processing required to validate these experimental results is discussed. These results are also compared with those produced by an equivalent linear equalizer or inverse filter, which provides the optimal solution when it incorporates all of the reverberations.

1:15

2pSPa2. Experimental design and processing for time-reversal communications in a highly reverberant environment. Brian L. Guidry, James V. Candy, Andrew J. Poggio, and Alan W. Meyer (Lawrence Livermore Natl. Lab., 7000 East Ave. L-154, Livermore, CA 94550)

A suite of experiments has recently been conducted to validate the utility of using time-reversal (T/R) theory to solve a communications problem for highly reverberant environments. This paper discusses the design and layout of those experiments as well as experimental equipment criteria and selection. Solutions to problems arising from equipment limitations encountered during the experimental design process are examined. The signal processing used to extract information from gathered data is described and it is shown that communications receivers utilizing T/R theory can be used to accurately reproduce messages broadcast through hostile, reverberant communications channels.

1:30

2pSPa3. Deconvolutional beamforming for air and underwater acoustic sensor arrays. Geoffrey S. Edelson, Shelby F. Sullivan, Jr. (BAE Systems, P.O. Box 868, Nashua, NH 03061-0868, geoffrey.s.edelson@baesystems.com), James M. Alsup (AETC, Inc., San Diego, CA 92122), and Harper J. Whitehouse (Linear Measurements, Inc., San Diego, CA 92121)

Autonomous acoustic sensor arrays are often plagued with poor spatial resolution capabilities, especially at the lowest frequencies. Beamforming techniques that are data-adaptive can improve upon the resolution capability of conventional beamformers under certain conditions. An alternate nonadaptive technique that is simply implemented and that achieves sidelobe reduction while retaining or narrowing main lobe beamwidth is presented. The method for weight matrix construction is based on regularized, constrained deconvolution. These deconvolutional beamformer (DBF) beamspace weights, which are precomputed off-line using this nonlinear process, are applied to the data using a linear process. The physical array may be smaller when using the DBF since this beamformer simultaneously reduces the width of the main lobe at low frequencies while reducing the integrated sidelobe level. This is something even "optimal" beamformers have difficulty doing since the main lobe generally increases as the sidelobes are reduced. One additional DBF attribute, when compared to most traditional weighting functions, is its compatibility with nonuniformly spaced arrays, including the special case of receive elements that are known to be missing. The automated procedure for determining these beamspace weights is discussed and results are shown for both air and underwater acoustic array data.

1:45

2pSPa4. Time-reversal super-resolution and stability in a weak scattering medium. Christopher D. Jones (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, cjones@apl.washington.edu) and James G. Berryman (Univ. of California, Lawrence Livermore National Lab., Livermore, CA 94551)

The properties of super-resolution and stability in time-reversal focusing of an acoustic wave are investigated using perturbation theory in a finite scattering medium with weak sound-speed fluctuations. Although super-resolution is obscured by weak scattering of the coherent field, the incoherent field still shows time-reversal enhanced focusing and the analytically tractable results may provide physical insight into time-reversal in stronger scattering media. Using first-order perturbation theory, the correlation scales of focused time-reversed field are defined by the scales of the fluctuations in the medium, the scale of effective aperture of the time-reversal array due to scattering and boundary conditions, the bandwidth of the system, and the array geometry. In this weak scattering example analytic expressions for the first and second moments of the time-reversal Green's function operator are found and provide physical insight into the mechanism of time-reversal focusing and the role of bandwidth in super-resolution and stability. Numerical simulation using discrete point scatterers are performed to compare with the analytic results and to investigate the transitional behavior of time-reversal focusing from weak to stronger scattering.

2:00

2pSPa5. Observations of severe storms with infrasound. John M. Noble and Stephen M. Tenney (Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, jnoble@arl.army.mil)

We report on data recorded from small aperture (20-m) infrasonic microphone arrays. The array monitors man-made and natural sources of infrasound from 0.1 to 25 Hz. The array runs 24/7 and stores the data into hourly blocks. Each block of data is passed through a quick-look algorithm to determine if an interesting event occurred. It was noted about a year after operation began that severe thunderstorms would show up as detections and the array would track the passage of the storm system. The data collected during the operation of the array will be presented along with possible ideas on what the array is detecting within the storm. Also the preliminary results of a field study will be presented where a portable infrasound system was placed near severe storm cells in the Midwest during June 2003.

2:15

2pSPa6. A method for making acoustically bright intensity zone. Joung-Woo Choi and Yang-Hann Kim (Ctr. for Noise and Vib. Control, Dept. of Mech. Eng., KAIST, Science Town, Daejeon-shi 305-701, Korea, yanghannkim@kaist.ac.kr)

Acoustic variables on a selected zone can be manipulated by controlling input signals of the fixed sound sources. In the previous work [J.-W. Choi and Y.-H. Kim, J. Acoust. Soc. Am. **111**, 1695 (2002)], acoustic potential energy was controlled to make a bright and dark sound zone in space. Extending this work, this paper addresses a method to manipulate acoustic intensity in the zone of interest. This inevitably has to do with a magnitude as well as direction control, simply because intensity is a vector quantity. To accomplish this objective, it is required to define the acoustic intensity that can represent acoustic intensity distribution in the selected zone. This has been attempted by defining mean intensity projected to the desired direction. This mean intensity is maximized provided that the input power is kept constant. [Work supported by NRL project of KISTEP and the BK21 project initiated by the Ministry of Education and Human Resources Development of Korea.]

2:30

2pSPa7. Target sound source extraction using spectral subtraction with a two-channel microphone array. Takahiro Murakami and Yoshihisa Ishida (Dept. of Electron. and Commun., Meiji Univ., 1-1-1, Higashi-Mita, Tama-Ku, Kawasaki 214-8571, Japan)

A novel method for extracting a target sound source from the measurements of a two-channel microphone array is presented. In this paper, it is assumed that there is a single target sound source located directly in front of the array and a single noise source located in the other direction. The proposed method exploits the power spectrum of the difference signal between the signals observed at two microphones. Since the arrival times of the target sound are equal and those of the noise source are different, the difference signal contains only the noise signal. In the proposed method, the power spectrum of the difference signal is linearly modified on the assumption that the two microphones are located sufficiently close to each other, and then, the target source is extracted by using a spectral subtraction in which the power spectrum of the difference signal is subtracted from that of the observed signal. The proposed method requires neither delay estimation nor direction-of-arrival estimation, which are generally used for modification of the spectrum of the difference signal. Therefore, the proposed method is performed by a simple algebraic calculation.

2:45

2pSPa8. Software-centered implementation of 128 channel huge speaker array with stock PC. Hiroshi Mizoguchi, Yuki Tamai, Koichi Nagashima (Dept. of Mech. Eng., Tokyo Univ. of Sci., 2641 Yamazaki, Noda 278-8510, Japan), Satoshi Kagami, Tachio Takano (Digital Human Res. Ctr., AIST, Japan), and Koichi Nagashima (R-lab., Inc., Japan)

A huge speaker array system of 128 loudspeakers was constructed and experimented. It was implemented as "software-centered" style utilizing stock loudspeakers and a PC. No dedicated hardware nor DSP was utilized. Spot forming, instead of beam forming, could be realized by 32 by 4 square layout of the array. Spot means small area of higher sound pressure level. Number of the spot was not limited to one. In the experiment, within 3 m by 3 m area, four spots of different sounds could be simultaneously formed. This spot forming was confirmed by actually measured spatial distribution of sound pressure level. The effect of the spot was also confirmed auditorily. Since the system was software-centered, it was dynamic. By simply changing software parameters, location of the spot can be easily moved even while the system was running. This movability of the spot was intended to be basis for visual steering. To realize the system, a simultaneous 128 channel 14-bit DA converter PCI board was developed. 44.1 kHz sampling rate was achieved by 2.4-GHz Intel Xeon-based PC utilizing the DA board and a real-time OS, named ART-Linux. Approximately 23-μs loop could be realized by software. It was the world's fastest software loop.

3:00

2pSPa9. Development of a handheld bistatic imaging sonar system for underwater search and survey. Alice Chiang, Steven Broadstone, and John Impagliazzo (Teratech Corp., 77–79 Terrace Hall Ave., Burlington, MA 01803)

A high resolution, handheld imaging sonar system is under development by Teratech Corporation for the U.S. Navy. This is a 192 channel, dual frequency bistatic sonar for Navy divers performing search and survey missions for underwater explosives. Our goal is to provide the most compact and energy efficient imaging system for the divers. The system consists of a self-contained handheld unit and a head mounted display integrated into the divers mask. The low power and small volume are a result of the development of Teratechs Charge Domain Processing (CDP) technology. This technology has led to the development of a low power 64-channel beamformer chip. As a result, only three beamformer chips will be required for the 192 channels. Until now, the implementation of small, low power sonar systems containing this many elements and forming enough beams to create an image was considered impossible. Progress in the development of this product will be presented. In-water testing is planned for late summer 2003. Experimental results and test images available will be presented at the conference. [Work sponsored by ONR and OSD Small Business Innovative Research Program, Program manager, Mr. Bruce Johnson, Naval Explosive Ordnance Disposal Technology Division.]

Session 2pSPb

Signal Processing in Acoustics and Speech Communication: Novel and Hybrid Methods (Poster Session)

Ashok Krishnamurthy, Chair

Department of Electrical Engineering, The Ohio State University, Columbus, Ohio 43210

Contributed Papers

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and contributors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

2pSPb1. Speech enhancement using parametric spectral subtraction combined with generalized sidelobe canceller. Jaeyoun Cho and Ashok Krishnamurthy (Dept. of Elec. Eng., The Ohio State Univ., 2015 Neil Ave., Columbus, OH 43210, cho.163@osu.edu)

Speech enhancement is an important problem with applications in hearing aid design, speech recognition, speech coding, etc. Parametric spectral subtraction is a common method for speech enhancement when only a single channel of data is available. On the other hand, beamforming methods can be used when multiple channels of spatially separated data are available, such as from a microphone array. In previous work, we have shown that spectral subtraction combined with spatial averaging from multiple microphones leads to improvements in speech SNR and reduction of musical noise compared with either method used alone. In this talk, we extend the previous work to combine parametric spectral subtraction with adaptive beamforming, specifically the generalized sidelobe canceller. The proposed parametric spectral subtraction method determines the parameters adaptively so as to minimize speech distortion. In addition, it is shown that the major drawback of spectral subtraction, so-called musical noise, can be diminished by adaptive beamforming process. We show that the method leads to a reduction of musical noise and results in the enhanced speech having better quality and intelligibility.

2pSPb2. A speech enhancement method using the sliding DFT. Hiroyuki Ono, Takahiro Murakami, and Yoshihisa Ishida (Dept. of Electron. and Commun., Meiji Univ., 1-1-1, Higashi-mita, Tama-ku, Kawasaki 214-8571, Japan)

In this paper, we propose a novel noise reduction of speech signals using sliding DFT (SDFT) which is based on the spectral subtraction method. The spectral subtraction method reduces noise by subtracting the spectrum of noise estimated from an input signal and is a relatively simple and effective technique. We use SDFT to obtain the frequency spectra of speech signals. As is well known, to obtain the successive N-points output, SDFT requires only N times of complex multiplication, although the Fast Fourier Transform (FFT) requires $N\log 2N$ times that. Thus SDFT is computationally simple as compared with FFT. The frame length, N, is determined based on the pitch period. In this paper, the pitch period is estimated by using an ideal low-pass filter and the analytic signal that is obtained from speech signals by the Hilbert converter. Finally, it is shown that the proposed method has good performance for speech enhancement.

2pSPb3. Spoken digits recognition using DP matching combined with a subspace decomposition method. Ken Kusakari, Kurihara Kiyoshi, Takahiro Murakami, and Yoshihisa Ishida (Dept. of Electron. and Commun., Meiji Univ., 1-1-1, Higashi-mita, Tama-ku Kawasaki 214-8571, Japan)

In this paper, we propose a method for spoken digits recognition using DP Matching combined with subspace decomposition that linearly separates into phonetic information from speaker information based on principle component analysis [M. Nishida and Y. Ariki, IEICE Trans. Japan **J85-D-II**, No. 4 (2002)]. This method allows for more robust speech recognition of less standard speech patterns. The use of the spectral envelope by LPC in speech recognition is unable to avoid errors in recognition due to the uncertainty of personalities, the dynamic variation of features, and so on. By using the subspace method, the proposed method eliminates these problems and enables good recognition results of less standard speech patterns. We use DP matching in recognizing, because it allows for more efficient pattern matching by normalizing the length of vowels. Simulation results show that the proposed method, using orthonormal projection to phonetic subspace with less speaker information, is superior to the conventional method using LPC spectra and DP Matching.

2pSPb4. Voice conversion using the radial basis function network and the frequency–domain pitch modification technique. Tomomi Watanabe, Takahiro Murakami, and Yoshihisa Ishida (Dept. of Electron. and Commun., Meiji Univ., 1-1-1, Higashi-mita, Tama-ku, Kawasaki 214-8571, Japan)

This paper presents a novel algorithm for developing a voice conversion system that modifies the speech uttered by a speaker to sound as if produced by another target speaker. And an improved technique for a frequency–domain pitch modification is proposed. The approach is based on modifying spectral envelopes and the pitch of voiced sounds separately. The conversion rules for the spectral envelopes are constructed by the radial basis function (RBF) network, which is one of the well-known artificial neural networks. An excitation signal containing prosodic information is modified to match the average pitch using the proposed technique which proves that the speech quality deteriorates when the pitch is modulated by the Frequency–Domain Pitch-Synchronous Overlap and Add (FD-PSOLA) algorithm. The simulation results show that the proposed method achieves nearly optimal spectral conversion performance.

2p TUE. PM

Moreover, average cepstrum distance to the target speech is reduced by 87%, and the listening tests prove the proposed pitch modification technique maintains higher speech quality than the original FD-PSOLA.

2pSPb5. Musical noise reduction using an adaptive filter. Takeshi Hanada, Takahiro Murakami, Yoshihisa Ishida (Dept. of Electron. and Commun., Meiji Univ., 1-1-1, Higashi-mita, Tama-ku, Kawasaki 214-8571, Japan), and Tetsuya Hoya (Lab. for Adv. Brain Signal Processing BSI-RIKEN 2-1, Hirosawa, Wakoh, Saitama 351-0198, Japan)

This paper presents a method for reducing a particular noise (musical noise). The musical noise is artificially produced by Spectral Subtraction (SS), which is one of the most conventional methods for speech enhancement. The musical noise is the tin-like sound and annoying in human auditory. We know that the duration of the musical noise is considerably short in comparison with that of speech, and that the frequency components of the musical noise are random and isolated. In the ordinary SS-based methods, the musical noise is removed by the post-processing. However, the output of the ordinary post-processing is delayed since the post-processing uses the succeeding frames. In order to improve this problem, we propose a novel method using an adaptive filter. In the proposed system, the observed noisy signal is used as the input signal to the adaptive filter and the output of SS is used as the reference signal. In this paper we exploit the normalized LMS (Least Mean Square) algorithm for the adaptive filter. Simulation results show that the proposed method has improved the intelligibility of the enhanced speech in comparison with the conventional method.

TUESDAY AFTERNOON, 11 NOVEMBER 2003 SABINE ROOM, 1:00 TO 5:00 P.M.

Session 2pUW

Underwater Acoustics, Acoustical Oceanography and Physical Acoustics: The Acoustics of Bubbles in the Marine Boundary Layer

Grant B. Deane, Chair
Marine Physical Laboratory, University of California–San Diego, La Jolla, California 92093-0238

Chair's Introduction—1:00

Invited Papers

1:05

2pUW1. Scattering and attenuation from near-surface bubbles. Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

Scattering from bubbles in the vicinity of the air–sea interface can often be the dominant source of apparent sea surface backscatter, particularly for lower grazing angles and for frequencies of O(10) kHz and above. Such scattering, and by inference, the near-surface bubble concentration, is observed in field data to be highly correlated with a nonlinear function of wind speed, such as a power or exponential law. A hysteresis effect is also observed, wherein for a given wind speed there is a tendency for scattering level to be higher if prior winds had been falling. The situation changes for acquisition geometries far removed from monostatic or bistatic backscattering; for example, in the case of bistatic forward scattering, the scattering level from bubbles rarely reaches the strength of that from the air–sea interface. Here, though, a bubble-mediated attenuation can be observed in field data, which is also wind-speed dependent with hysteresis. However, linking together field measurements of scattering and attenuation associated with near-surface bubbles has long been a vexing problem. This topic is discussed using examples from various field experiments, the most recent of which is ASIAEX East China Sea (2001), and from well-controlled laboratory experiments. [Research supported by ONR.]

1:25

2pUW2. The impact of bubbles on underwater acoustic communications in shallow water environments. James Preisig (Dept. of Appl. Ocean Phys. and Eng., WHOI, Woods Hole, MA 02543)

Two characteristics of the acoustic channel that significantly impact the performance of underwater acoustic communications systems are the total energy throughput of the channel and the delay spread of the channel impulse response. These characteristics impact both coherent (e.g., phase shift keyed) and incoherent (e.g., frequency shift keyed) systems. Experimental data are presented from the surf zone and shallow-water environments demonstrating the impact of bubbles on these characteristics. A novel statistical analysis utilizing a *variance reduction ratio* is used to identify the measurable environmental variables (e.g., significant wave height) that can be used to best predict the impact of the bubbles. [Work supported by ONR Ocean Acoustics.]

1:45

2pUW3. The noise spectral parameters and the energy of breaking waves experimental study. Jaroslaw Tegowski (Inst. of Oceanology, Polish Acad. of Sci., Sopot, Poland, tegowski@iopan.gda.pl)

The process of wave breaking and whitecap creation is one of the most important and least understood phenomena associated with the evolution of the surface gravity waves in the open sea. This process is the main way of energy and momentum transfer between ocean and atmosphere. However, it is very difficult to estimate, under real sea conditions, the frequency of breaking wave events or

the fraction of sea surface covered by whitecaps and the amount of dissipated energy produced by wave breaking. A controlled experiment was carried out in the Ocean Basin Laboratory at MARINTEK, Trondheim (Norway). The simulation of random waves of the prescribed spectra provided a very realistic pattern of the sea surface. The number of breaking waves was estimated using photography method and wave staff recording. Acoustic measurements during the experiments were conducted in order to examine the relationship between the noise spectral parameters and both the whitecap coverage and dissipation energy of breaking waves for different types of waves. A comparison of simultaneous video observations, wave staff records of the surface wave above the hydrophones, with the spectral parameters of acoustical signals made it possible to find physical links between processes.

2:05

2pUW4. Propagation and scattering in bubbly liquids near resonance: A review of recent laboratory measurements. Preston S. Wilson, Eun-Joo Park, William M. Carey, and Ronald A. Roy (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

Acoustic waves propagating through bubbly liquid experience dispersion and attenuation when the excitation frequency is near the resonance frequency region of the bubble distribution. Due to high attenuation in this regime, traditional standing wave and time-of-flight measurement techniques fail and existing theories remain unverified above void fractions of about 10^{-4}. An impedance tube technique was developed in order to investigate this regime by inferring phase speed and attenuation within a bubbly layer from its complex reflection coefficient. Results of experiments using this technique, as well as its limitations, will be reviewed. We found that existing theory described the measured results near the resonance region within the uncertainty of the measured parameters of the bubble population. In a second class of experiments, scattering from laboratory bubble clouds was investigated. Here, the bubbles were much smaller than resonance size and the bubbly liquid was treated as an effective medium, with density given by a mixture law and sound speed given by Wood's equation. Experimental results and the predictions of an effective fluid scattering model will be reviewed for spherical and cylindrical bubbly-liquid targets. [Work supported by U.S. Navy ONR.]

2:25

2pUW5. Laboratory measurements of the 1st and 2nd moments of propagation through bubbles in a flow. Thomas Weber, David Bradley, R. Lee Culver, and Anthony Lyons (Appl. Res. Lab. and the Grad. Prog. in Acoust., P.O. Box 30, State College, PA 16804, tcw141@psu.edu)

The classic theory describing the change in soundspeed and attenuation for waves propagating in bubbly fluids (i.e., the 1st moment of the multiple scattering solution) can be viewed as a statement about the average pressure observed over all possible configurations of bubble position and size. The 2nd moment can also be predicted using the multiple scattering formulation, and both moments can be found independent of any knowledge of the fluid dynamic conditions controlling the mixing of bubbles. Characteristics of the fluid dynamics do appear in the time scales associated with this mixing, however, and this should be reflected in the time scales associated with convergence to the 1st and 2nd moments. This was explored by conducting propagation experiments in a tank filled with bubble-laden water that was subjected to varying flow conditions. Frequency dependent attenuation measurements were made and inverted for the bubble size distribution so that predictions of the 2nd moment could be compared to the observed quantities. High ping rates and long data records were used to observe the flow-dependent time scales associated with the measurements. The results of these experiments will be presented and discussed. [Work supported by ONR Code 321US. First author supported by a National Defense Science and Engineering Graduate Fellowhip.]

2:45

2pUW6. Listening for ambient bubbles in the marine boundary layer. Jeffrey A. Nystuen (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Bubbles created by breaking waves and raindrop splashes are responsible for much of the high-frequency (500 Hz–50 000 Hz) ambient sound in the ocean. Under high sea state and during heavy rainfall conditions clouds and layers of ambient bubbles in the marine boundary layer absorb sound from bubbles being newly created at the ocean surface. This absorption changes the shape of the ambient sound spectrum below the bubbly surface marine layer, allowing passive detection of the ambient bubbles. This change in the shape of the ambient sound spectrum is especially apparent in the situation of rainfall during high wind conditions, presumably because existing wind-generated turbulent from breaking waves is available to transport rainfall-generated bubbles downward. Estimates of vertically integrated ambient bubble populations can be made from the presumed frequency-dependent attenuation in the ambient sound spectrum. Ambient sound data from many months of deep ocean mooring measurements will be used to document this phenomenon. [Work sponsored by ONR Ocean Acoustics.]

3:05–3:15 Break

Contributed Papers

3:15

2pUW7. Signal propagation and the dispersion relations of bubbly water. Gregory J. Orris, Michael Nicholas, and Dalcio Dacol (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375)

The physics behind the extreme variations of the phase velocity of acoustic wave propagation in a bubbly liquid as a function of frequency have been addressed theoretically. Until recently there has not been a concerted experimental effort to validate this work over the rather large parameter landscape covered by the theories. To this end there there have been several sets of experiments conducted at the Naval Research Laboratory's Salt-Water Tank Facility aimed in part at validating these theories. In order to properly measure phase velocities in a semi-free field environment, one must be able to accurately measure the arrival of a short acoustic pulse. This leads naturally to the question of what one measures as the velocity of sound in a highly dispersive medium where the group velocity

can indeed have infinite, zero, or negative values. We discuss this and its relationship to work in other fields of physics. [Work supported by ONR.]

3:30

2pUW8. Measurements of the attenuation and phase velocities in bubbly fresh and salt water. Michael Nicholas and Gregory J. Orris (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375)

Several types of bubblers have been used to create a dense bubble cloud that fills a 44-m^3 tank, wherein phase velocity and attenuation measurements have been made over a wide range of frequencies. Initial experimental results on the measurement of the phase velocity and attenuation have been previously reported [J. Acoust. Soc. Am. **112**, 2269]. We present in this paper how this work has been extended with respect to the environmental parameter landscape. Paramount among these parameters are the bubble size distribution, void fraction, temperature, and surface tension. In these sets of experiments, void fractions from 0.5% to roughly 4% are investigated over a wide set of temperatures to give a much larger experimental picture of acoustic propagation through near-surface bubble clouds. Current techniques for measuring these quantities will be discussed and results presented within the context of current theories with implications on ocean acoustic experiments. Additionally, initial work using data collected using various concentrations of salt water will also be presented. [Work supported by ONR.]

3:45

2pUW9. Pulse distortion of a signal passing through a bubble cloud. Ralph R. Goodman and Jerald W. Caruthers (Dept. of Marine Sci., The Univ. of Southern Mississippi, Stennis Space Center, MS 39529)

At the 145th Meeting of the Acoustical Society the authors presented a paper on the dispersion and absorption of sound in an idealized bubble field that had a bubble radius size distribution $n(a)$ decreasing as the fourth power [J. Acoust. Soc. Am. **113**, 1278 (2003)]. It was shown that, for this case, exact, relatively simple, analytic solutions exist both for dispersion and absorption. The same model is used here to calculate the effects of dispersion and absorption on pulsed signals as they propagate through a bubbly mixture. The approach is essentially the same as that used by Frank Henyey (unpublished) to analyze the propagation of signals through clouds that were observed in the well-known 1997 Scripps Pier experiment. The effects on signal shapes as a function of range and bubble density and size distribution will be presented.

4:00

2pUW10. Scaling laws for bubble acoustic radiation strength in whitecaps. Grant B. Deane (Scripps Inst. of Oceanogr., Mail Code 0238, La Jolla, CA 92093-0238, grant@mpl.ucsd.edu)

The past decade has seen significant advances in our understanding of open-ocean bubble plumes, including scaling laws for whitecap properties [W. K. Melville and P. Matusov, Nature (London) **417**, 58–63 (2002)] and the physics of bubble creation [Garrett *et al.*, J. Phys. Oceanogr. **30**, 2163–2171 (2000), and G. B. Deane and M. D. Stokes, Nature (London) **418**, 839–844 (2002)]. Despite this progress, we do not yet have a complete understanding of the relationship between noise radiated by whitecaps and the bubble creation processes occurring within the wave crest. Understanding this relationship is important: it would allow bubble-mediated gas transfer rates to be estimated from wind speed, for example, and provide a model for the Knudsen spectrum of underwater ambient noise. An important piece of this puzzle is the dependence of bubble acoustic radiation strength on bubble radius. We will present estimates of the scaling laws for this relationship based on open-ocean measurements of whitecap noise and laboratory measurements of bubble creation rates in plunging breakers. [Work supported by ONR and NSF.]

4:15

2pUW11. An experiment on near-shore sonar performance. Nicolas Le Dantec and Grant B. Deane (Scripps Inst. of Oceanogr., Univ. of California–San Diego, 9500 Gilman Dr., La Jolla, CA 92093)

In order to further investigate the use of sonar systems in the near-shore environment, a simple, short range sonar system has been deployed along with additional devices to characterize the environmental conditions. The goal of the experiment is to determine the effects of volume and boundary conditions on the performance limits of sonar in the near-shore. A 10 kHz source transmitted a series of acoustic pulses with Barker code phase modulation along a 40 m propagation path to spherical targets just outside the surf zone north of Scripps Pier. Both forward and backscatter acoustic signals were recorded. A pressure array mounted on the seafloor measured the shoaling surface waves along the transmission path and the seawater conductivity, temperature and horizontal velocity was measured mid-way between the source and targets. Meteorological data were also collected. The effects of surface gravity waves focusing and multi-path arrivals on target discrimination will be addressed, along with scattering from artificially created bubble clouds. [Work supported by ONR.]

4:30

2pUW12. Bubble acoustics near surfaces. Dalcio K. Dacol and Gregory J. Orris (Acoust. Div., Naval Res. Lab., Washington, DC 20375)

Boundary effects were included in Foldy's effective medium theory of acoustic propagation in a bubbly fluid. This allowed for the discussion of bubble effects on propagation near interfaces and in waveguides such as the ones encountered in ocean acoustics. The implications of this model were examined in a variety of situations from laboratory experiments to acoustic propagation in shallow waters.

4:45

2pUW13. Evaluation of bubble–saturated subsurface sea layer by means of optoacoustics. Sergey V. Egerev (Sci. Council on Acoust., Russian Acad. of Sci., 4 Shvernika St., Moscow 117036, Russia)

Laser sound generation has proved to be a useful tool for probing of various media. The pressure signal outgoing from the point of optoacoustic conversion provides information on the medium properties. The results are presented concerning optoacoustic conversion in the gas-containing bubbly subsurface sea layer. The sea surface is irradiated by a laser pulse. The subsequent acoustic signal, very susceptible to the subsurface gas content and propagating along the vertical axis z is recorded by a submerged hydrophone and interpreted. A model is developed to predict the features of signal, recorded during full–scale experiment conditions, making clear the formation of signal in nonuniform bubble–saturated subsurface layer, characterized by strong dispersion and additional acoustical absorption. The sound velocity increment $\Delta c(z)$ along with the additional attenuation factor $\alpha(z)$ are parameters of a dispersion environment dependent upon the distribution of numerical bubble concentration $n(R,z)$ upon their radii R at a current depth z. Laboratory test results of the phenomena along with the White Sea full-scale experimental results are presented. The equipment issues necessary for such measurements are discussed.