



Sound pressure distribution in a long, narrow hallway: Measurements versus results from a computer model with scattering from surface roughness and diffraction

Wang, Lily M.; Rathsam, Jonathan; Christensen, Claus Lyngge; Rindel, Jens Holger

Published in:
Journal of the Acoustical Society of America

Publication date:
2005

Document Version
Publisher's PDF, also known as Version of record

[Link back to DTU Orbit](#)

Citation (APA):
Wang, L. M., Rathsam, J., Christensen, C. L., & Rindel, J. H. (2005). Sound pressure distribution in a long, narrow hallway: Measurements versus results from a computer model with scattering from surface roughness and diffraction. In *Journal of the Acoustical Society of America* (Vol. 118/3, pp. 4aAA3). Acoustical Society of America.

General rights

Copyright and moral rights for the publications made accessible in the public portal are retained by the authors and/or other copyright owners and it is a condition of accessing publications that users recognise and abide by the legal requirements associated with these rights.

- Users may download and print one copy of any publication from the public portal for the purpose of private study or research.
- You may not further distribute the material or use it for any profit-making activity or commercial gain
- You may freely distribute the URL identifying the publication in the public portal

If you believe that this document breaches copyright please contact us providing details, and we will remove access to the work immediately and investigate your claim.

Session 4AA

Architectural Acoustics: Early Reflections on Reflections

Peter D'Antonio, Cochair

RPG Diffusor Systems Inc., 651C Commerce Dr., Upper Marlboro, MD 20774

Trevor Cox, Cochair

RPG Diffusor Systems Inc., 651C Commerce Dr., Upper Marlboro, MD 20774

Chair's Introduction—9:00

Contributed Papers

9:05

4aAA1. Creative reflections—the strategic use of reflections in multitrack music production. Alexander Case (Dept. of Music, Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Lowell, MA 01854, alex_case@uml.edu)

There is a long tradition of deliberately capturing and even synthesizing early reflections to enhance the music intended for loudspeaker playback. The desire to improve or at least alter the quality, audibility, intelligibility, stereo width, and/or uniqueness of the audio signal guides the recording engineer's use of the recording space, influences their microphone selection and placement, and inspires countless signal-processing approaches. This paper reviews contemporary multitrack production techniques that specifically take advantage of reflected sound energy for musical benefit.

9:20

4aAA2. Simulating the acoustics of classical sound recording techniques in rooms. Jonas Braasch, William L. Martens, and Atsushi Marui (CIRMMT, Faculty of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada)

Recently, a method to position sound sources in 3-D space using virtual microphone control has been proposed [J. Acoust. Soc. Am. **117**, 2391]. In this computer-generated environment, gains and delays between a virtual sound source and virtual microphones are calculated according to their distances and the axis orientations of the microphone directivity patterns. In the follow-up study reported here, it was investigated how to best simulate the influence of a room on the virtual microphone recording. For this purpose, a virtual rectangular room was created using the mirror-image technique (up to second-order reflections). The room dimensions were copied from an existing concert space at McGill University. Late reverb was created using a multiple feedback delay network with a time-variant architecture to enable modulation. To evaluate the system, measured impulse responses between a sound source and a five-channel microphone setup were compared to the virtual impulse responses for the same room/microphone-placement configuration. Among the tested parameters was the signal ratio between direct sound source and reflections as a function of the microphone placement and choice of microphone directivity patterns. The coherence between the simulated microphone signals was adjusted as well from the measured data. [Work supported by NSERC and VRQ.]

9:35

4aAA3. Sound-pressure level distribution in a long, narrow hallway: Measurements versus results from a computer model with scattering from surface roughness and diffraction. Lily M. Wang, Jonathan Rathsam (Architectural Eng. Program., Univ. of Nebraska-Lincoln, Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0681, lwang4@unl.edu), Claus L. Christensen, and Jens H. Rindel (Tech. Univ. of Denmark, DK-2800 Kgs. Lyngby, Denmark)

The sound-pressure level distribution down a long, narrow hallway due to a sound source at one end does not decrease linearly along the length of the hall. This characteristic may be due to the changing behavior of scattering that occurs down the length of the hallway, which is distance- and angle-dependent. A new scattering method that incorporates these effects has been implemented in the room acoustic computer modeling program, ODEON [C. L. Christensen and J. H. Rindel, Forum Acusticum, Budapest (2005)]. A comparison of the results from an ODEON model with real-world measurements along a long, narrow hallway on the campus of the Technical University of Denmark is provided in this paper. [Work supported by the NSF.]

9:50

4aAA4. Surface characterization from the point of view of an acoustical consultant. Scott D. Pfeiffer, Jacob Ament, Zackary Belanger, Molly Norris (Kirkegaard Assoc., 801 West Adams St., 8th Fl., Chicago, IL 60607), and Tim Gulsrud (Kirkegaard Assoc., Boulder, CO 80302)

The practical characterization of the acoustical properties of absorption, diffusion, and transparency of surfaces is an area where the application of physical acoustic techniques may be more applicable than typical architectural acoustical parameters. Understanding the diffusive behavior of surfaces is crucial to the design of spaces, but cumbersome to quantify, requiring different techniques depending on the question being asked. Past studies conducted by Kirkegaard Associates on products for partial reflection, diffusion, nominal transparency, and the results of a recent implementation of the current proposed Audio Engineering Society standard AES-4id-2001 are discussed with respect to the need to make design decisions and implement products for fulfillment of an acoustic design process. The viability of using a single-valued diffusion coefficient is discussed with respect to the needs of the acoustical consulting community, while still exploring the possibility of alternative characterization methods.

10:05

4aAA5. A narrow-band analysis of reflected magnitude and phase from six reflector panel arrays. Jonathan Rathsam, Lily M. Wang (Architectural Eng. Program, Univ. of Nebraska-Lincoln, 247 PKL, 1110 S. 67th St., Omaha, NE 68182-0681, jrathsam@mail.unomaha.edu), and Rendell R. Torres (Rensselaer Polytechnic Inst., Troy, NY 12180-3590)

This investigation analyzes measurements of the reflected sound field from various reflector panel arrays. A previous study by the third author [R. R. Torres and M. Vorlaender, "Scale-model MLS-measurements of scattering from overhead panel arrays," *Acta Acustica* (in press)] measured impulse responses using the maximum length sequence method at various receiver positions from six scale-model panel arrays of different sizes and densities. This study included an octave band analysis of reflected magnitude as a function of receiver position. The current authors have extended the research by conducting a narrow-band analysis of magnitude and phase of the reflected sound fields for three of the panel arrays [J. Rathsam, L. M. Wang, and R. R. Torres, *J. Acoust. Soc. Am.* **117**, 2499(A) (2005)]. This study revealed a close relationship between phase and magnitude. The current work continues the analysis of reflected phase and magnitude for the three additional reflector panel arrays. The newer arrays are more complex and may reveal the effects of multiple diffraction more clearly than the previous three arrays.

10:20–10:35 Break

10:35

4aAA6. A stable transient boundary element method (BEM) for diffuser scattering. Jonathan A. Hargreaves and Trevor J. Cox (Univ. of Salford, M5 4WT, UK)

Boundary element methods (BEM) may be used to model scattering from hard rigid surfaces such as diffusers. They have the advantage over volumetric methods that only the surface need be meshed and the surface velocity potential found. Unlike the more widely used single frequency methodology, transient BEM discretizes integral equations to produce an iterative system that is marched on in time from known initial conditions to calculate how the velocity potential varies over time. This iterative process can be unstable, and this is one reason why transient BEM is not more widely used. Previous works on transient BEMs have focused on idealized surfaces, such as spheres and plates. However, little is published on the performance of these methods for more complex surfaces of interest, such as Schröder diffusers. Consequently, this paper presents an implicit scheme suitable for a surface comprising thin and solid sections. Such an implicit scheme has the benefits of not constraining time-step duration to the smallest surface detail, and brings stability benefits. Numerical integration is carried out efficiently and accurately by conversion to contour integrals. Accuracy and stability is investigated by comparison to a verified single frequency BEM.

10:50

4aAA7. Wavelet transform use for reflection analysis in architectural acoustics. Zackery Belanger (Kirkegaard Assoc., 801 West Adams St. 8th Fl., Chicago, IL 60607)

The use of wavelet transforms in the realm of architectural acoustics is discussed as an alternative to more traditional frequency-analysis techniques. Wavelet transforms allow the time and frequency information of a signal to be viewed simultaneously, providing a valuable way to study the frequency content of individual reflections and the frequency structure of impulse responses. Discussion will focus on the application of wavelet transforms to project work at Kirkegaard Associates.

11:05

4aAA8. Acoustics of a planetarium. Micah Shepherd, Timothy W. Leishman, and Sentagi Utami (Acoust. Res. Group, Dept. of Phys. and Astron., Brigham Young Univ., C110 ESC BYU, Provo, UT 84602)

Brigham Young University has recently constructed a planetarium with a 38-ft.-diameter dome. The facility also serves as a classroom. Since planetariums typically have poor acoustics due to their domed ceiling structures, acoustical recommendations were requested before its construction. The recommendations were made in an attempt to create an acceptable listening environment for lectures and other listening events. They were based in part on computer models and auralizations intended to predict the effectiveness of several acoustical treatments on the outer walls and on the dome itself. The recommendations were accepted and the planetarium was completed accordingly. A series of acoustical measurements was subsequently made in the room and the resulting acoustical parameters were mapped over the floor plan. This paper discusses these results and compares them with the predictions of the computer models.

11:20

4aAA9. A novel approach to achieving significant reverberation control in performance halls. David A. Conant and William Chu (McKay Conant Brook, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, DConant@MCBinc.com)

Conventional methods for achieving broadband, variable sound absorption in large halls normally include heavy application of sound-absorptive drapery and/or thick fibrous panels, applied near available surfaces below, at, and in volumes above the catwalk plane. Occasionally, direct adjustments to room air volume are also provided to effect double-sloped decays. The novel method described here combines carefully located, broad scattering and absorption in singular architectural elements and was applied to a new, 1200-seat concert hall. A change of 0.70 s RT60 in midfrequency is achieved in a visually dramatic manner while neither materially changing room volume nor introducing often-maligned drapery. The aggregate of reverberation control methodologies employed reduces the unoccupied RT60 at midfrequencies from about 3.2 to 1.7 s in this space programed principally for music, including pipe organ. Results of MLS measurements including binaural measurements and binaural recordings of anechoic material and CATT-acoustic modeling and auralizations are discussed.

11:35

4aAA10. Development of a new fabric canopy material. Zackery Belanger (Kirkegaard Assoc., 801 West Adams St. 8th Fl., Chicago, IL 60607)

The unique requirements of two concert halls, one existing and one new, have driven the development of a new fabric canopy material with beneficial reflection and transmission spectra. The development process is highlighted, including motivation, material choice, testing, and challenges such as absorption issues and architectural requirements. The potential of the fabric for use in other acoustic environments is also discussed.

11:50

4aAA11. Evaluation of diffuse reflections in a 1:10 scale model hall. Jin Yong Jeon, Jong Kwan Ryu, and Yong Hee Kim (School of Architectural Eng., Hanyang Univ., Seoul 133-791, Korea, jyjeon@hanyang.ac.kr)

The acoustical properties of a multipurpose hall (opera mode) were measured in a 1:10 scale model to investigate the effect of diffuse reflections. Measurements were made in stall areas with and without diffusers on the walls adjacent to the pit and on the side walls. The diffusers consisted of 15- to 25-cm-high hemispheres and 25-cm high polygons. A mix of these diffusers was used to cover about 40% of the area of the side walls. It was found that 1-IACCE3 increased at positions close to the walls

4a THU. AM

adjacent to the pit and the first reflection with maximum amplitude increased in some seats. The possibility of diffuse reflections either hitting the proscenium wall or soffit close to the side wall first, as well as reflections coming from both sides of the hall explains the reason for the bunch

of early reflections in the impulse responses. The effect of diffusers at each receiver position was investigated via subjective evaluation. The results from the subjective evaluation indicated that diffuse reflections were preferred to specular reflections.

THURSDAY MORNING, 20 OCTOBER 2005

HENNEPIN ROOM, 9:00 TO 11:00 A.M.

Session 4aAB

Animal Bioacoustics: Topics in Bioacoustics

Michael J. Ferragamo, Chair

Gustavus Adolphus College, Dept. of Biology, St. Peter, MN 56082

Contributed Papers

9:00

4aAB1. Whistle repertoire of bottlenose dolphins (*Tursiops truncatus*) in the Mississippi Sound. Erica N. Hernandez, Stan Kuczaj (Univ. of Southern Mississippi, 118 College Dr., Box 5025, Hattiesburg, MS 39402), and Moby Solangi (Inst. for Marine Mammal Studies, Gulfport, MS 39502)

The whistle repertoire of wild bottlenose dolphins (*Tursiops truncatus*) in the Mississippi Sound, part of the northern Gulf of Mexico, was investigated. There is a large population of dolphins in this area, and many dolphins that are now housed in zoos and aquariums were captured in the Mississippi Sound. This paper reports the types of whistles that are predominant in this area, and how these whistles are used in the context of concurrent surface behavior. Over the course of 1 year (April 2004–March 2005), dolphin whistles were recorded as part of an ongoing study of the effects of human activity on wild bottlenose dolphins. The surface behavior of the focal group was categorized at 1-min intervals as follows: mill, travel, mill/travel, feed, social, with boat, or with shrimp boat. Whistles were then categorized as one of the following: upsweep, downsweep, convex, concave, sine, or constant frequency. Preliminary analysis of the data suggests that both the rate of whistling and the types of whistles produced vary as a function of dolphin behavior. Further analysis of the data will reveal if different types of whistles are associated with specific surface behavior categories. [Research supported by Department of Commerce.]

9:15

4aAB2. Sound-conducting mechanisms for echolocation hearing of a dolphin. Vyacheslav A. Ryabov (Karadag Natural Reserve Natl. Acad. of Sci. of Ukraine, Kurortnoe, Feodosiya 98188, Crimea, Ukraine, ryaboff@ukr.net)

The morphological study of the lower jaw of a dolphin (*Tursiops truncatus p.*), and the modeling and calculation of its structures from the acoustic point of view have been conducted. It was determined that the cross-sectional area of the mandibular canal (MC) increases exponentially. The MC represents the acoustical horn. The mental foramens (MFs) is positioned in the horn throat, representing the nonequidistant array of waveguide delay lines (NAWDL). The acoustical horn ensures the traveling wave conditions inside the MC and intensifies sonar echoes up to 1514 times. This “ideal” traveling wave antenna is created by nature, representing the combination of the NAWDL and the acoustical horn. The dimensions and sequence of morphological structures of the lower jaw are optimal both for reception and forming the beam pattern, and for the amplification and transmission of sonar echoes up to the bulla tympani. Morphological structures of the lower jaw are considered as components of the peripheral section of the dolphin echolocation hearing.

9:30

4aAB3. Automatic recognition of harmonic bird sounds. Jason R. Heller (Dept. of Appl. Math, Stony Brook Univ., Stony Brook, NY 11794, hellerjr@ams.sunysb.edu) and John D. Pinezich (Adv. Acoust. Concepts, Hauppauge, NY 11788)

The method of sound recognition relies on a transformation of a sound into a spectrogram followed by extraction of the harmonics as curves. The extracted curves are called *frequency tracks*. A procedure called *find_feasible_sets* is used to extract sets of tracks that may correspond to harmonic sounds. If a set of tracks overlap each other sufficiently in time, then the set is designated a *feasible set*. Following the extraction of the feasible sets, the procedure *find_maximal_subsets* is applied to each feasible set. This procedure uses a function called *harmonic_relate* that determines if two tracks are harmonically related. All tracks that are not harmonically related to any other tracks in the feasible set are discarded. Furthermore, the feasible set is divided into *maximal subsets*. A maximal subset is a subset of the feasible set in which every track is harmonically related to one fixed track in the set called the *reference track* but no other tracks in the feasible set are related to the reference track. Each frequency track in a track set is transformed into a *feature vector* whose components describe the frequency, slope, and shape of the track. The species of birds analyzed are bluejay and herring gull.

9:45

4aAB4. An optimized toolchain for predicting directivity patterns from digital representations of biological shapes. Rolf Müller (School of Phys. and Microelectronics, Shandong Univ., Hongjia Lou 5, Jinan, China, mueller@sdu.edu.cn)

Animals have evolved intricate shapes which diffract emitted or received sound and thereby generate a specific directivity pattern. Computer-tomographic methods can generate high-resolution digital representations of these morphological structures in the form of three-dimensional voxel arrays. However, predicting acoustic directivity patterns from these representations with numerical methods can incur high computational cost, e.g., for large structures with fine detail and/or high wave numbers (as in bats and dolphins). Here, the design of a toolchain is described which can handle all steps of deriving a directivity prediction from a voxel representation: generation of a finite-element mesh, assembly of the system matrix, computation of an approximate solution, forward projection into the far field. All individual operations are performed by self-contained tools, which communicate through files. This gives access to intermediate results and limits re-execution upon parameter changes to downstream steps. At each stage, optimizations can be made based on the specifics of the problem such as the regular structure of the voxel array and the distance independence of the directivity. Use of these optimizations has resulted in a highly efficient performance, which is documented by measures for execution speed, memory usage, and accuracy.

10:00–10:15 Break

10:15

4aAB5. Equine acoustics: Anatomy of a whinny. David G. Browning (Dept. of Phys., Univ. of Rhode Island, 2 Lippitt Rd., Kingston, RI 02881, decibeldb@aol.com) and Peter M. Scheifele (Dept. of Animal Sci., Univ. of Connecticut, Storrs, CT 06269)

Of the roughly nine different vocalizations of a horse, the whinny appears to be the most interesting. A whinny is a horse's primary means of long range vocal communication; the bandwidth and variability offer the possibility of expression, at least at a primitive level. Acoustic analysis of a whinny shows three distinct domains: the initial frequency ramp-up running from 1 to 2 kilohertz in roughly 1 second, matched by a similar response in the second and third harmonics; secondly, this is followed by a nasal tremolo of a longer duration with generally a slight downslope in frequency (this is perhaps the stage most associated with the human perception of a whinny); and, lastly, a guttural tremolo, essentially the same sound as a nicker. In the samples analyzed, each domain seems to vary independently both in strength and duration. Attempts to link an aspect of a whinny with a particular behavior is still in a formative stage, complicated by the fact that a horse's behavior is usually based primarily on visual rather than acoustic inputs.

10:30

4aAB6. Identifying individual sperm whales acoustically using self-organizing maps. Juliette W. Ioup and George E. Ioup (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148)

The Littoral Acoustic Demonstration Center (LADC) is a consortium at Stennis Space Center comprising the University of New Orleans, the University of Southern Mississippi, the Naval Research Laboratory, and the University of Louisiana at Lafayette. LADC deployed three Environmental Acoustic Recording System (EARS) buoys in the northern Gulf of Mexico during the summer of 2001 to study ambient noise and marine mammals. Each LADC EARS was an autonomous, self-recording buoy capable of 36 days of continuous recording of a single channel at an

11.7-kHz sampling rate (bandwidth to 5859 Hz). The hydrophone selected for this analysis was approximately 50 m from the bottom in a water depth of 800 m on the continental slope off the Mississippi River delta. This paper contains recent analysis results for sperm whale codas recorded during a 3-min period. Results are presented for the identification of individual sperm whales from their codas, using the acoustic properties of the clicks within each coda. The recorded time series, the Fourier transform magnitude, and the wavelet transform coefficients are each used separately with a self-organizing map procedure for 43 codas. All show the codas as coming from four or five individual whales. [Research supported by ONR.]

10:45

4aAB7. Spectral identification of sperm whales from Littoral Acoustic Demonstration Center passive acoustic recordings. Natalia A. Sidorovskaia, Blake Richard (Phys. Dept., UL Lafayette, Lafayette, LA 70504, nsidorovskaia@louisiana.edu), George E. Ioup, and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA 70148)

The Littoral Acoustic Demonstration Center (LADC) made a series of passive broadband acoustic recordings in the Gulf of Mexico and Ligurian Sea to study noise and marine mammal phonations. The collected data contain a large amount of various types of sperm whale phonations, such as isolated clicks and communication codas. It was previously reported that the spectrograms of the extracted clicks and codas contain well-defined null patterns that seem to be unique for individuals. The null pattern is formed due to individual features of the sound production organs of an animal. These observations motivated the present studies of adapting human speech identification techniques for deep-diving marine mammal phonations. A three-state trained hidden Markov model (HMM) was used with the phonation spectra of sperm whales. The HMM-algorithm gave 75% accuracy in identifying individuals when it had been initially tested for the acoustic data set correlated with visual observations of sperm whales. A comparison of the identification accuracy based on null-pattern similarity analysis and the HMM-algorithm is presented. The results can establish the foundation for developing an acoustic identification database for sperm whales and possibly other deep-diving marine mammals that would be difficult to observe visually. [Research supported by ONR.]

Session 4aAO

Acoustical Oceanography and Underwater Acoustics: Environmental Variability and Sonar Systems

James F. Lynch, Cochair

Woods Hole Oceanographic Inst., 203 Bigelow Bldg., Woods Hole, MA 02543

William M. Carey, Cochair

Boston Univ., College of Engineering, Boston, MA 02215

Chair's Introduction—8:25

Contributed Papers

8:30

4aAO1. Time and frequency analysis of seismic airgun calibration data from an environmental acoustic recording system (EARS) buoy.

George E. Ioup, Juliette W. Ioup, Sean R. Chapin, Arslan M. Tashmukhambetov (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148), Joal J. Newcomb, William M. Sanders (Naval Res. Lab., Stennis Space Center, MS), Christopher D. Walker, Benjamin Brack, Grayson H. Rayborn, James M. Stephens (Univ. of Southern Mississippi, Hattiesburg, MS 39406), and Natalia A. Sidorovskaia (Univ. of Louisiana at Lafayette, Lafayette, LA 70504)

In the summer of 2003 two Environmental Acoustic Recording System (EARS) buoys were deployed in the northern Gulf of Mexico by the Littoral Acoustic Demonstration Center. The buoys were collocated and recorded ambient noise and seismic airgun array shots up to approximately 25 kHz. The gains and hydrophone sensitivities were set such that one EARS buoy could record the seismic shots without clipping and the other could record ambient noise. The M/V Kondor towed an airgun array on parallel linear tracks with horizontal closest points of approach to the buoy of 0, 500, 1000, 2000, and 5000 m, giving experimental data for a wide range of horizontal distances (up to 7 km) and arrival angles. The raw data were calibrated using the EARS system parameters to produce calibrated pressure time series for each shot. These data are analyzed in both the time and frequency domains. Maximum pressures for each shot as well as sound exposure levels (pressure squared integrated over time for 200 ms in this case) are presented. Also presented is a spectrogram analysis. The maximum time-domain peak pressure recorded is 200 dB *re*: 1 μ Pa. The maximum sound exposure level is 177 dB *re*: 1 μ Pa² s. [Research supported by Industry Research Funding Coalition.]

8:45

4aAO2. Littoral Acoustic Demonstration Center 2003 seismic calibration experiment: Modeling acoustic energy distribution from seismic exploration array.

Natalia A. Sidorovskaia (Phys. Dept., UL Lafayette, Lafayette, LA, 70504, nsidorovskaia@louisiana.edu), Arslan Tashmukhambetov, George E. Ioup, and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA 70148)

In the summer of 2003 the Littoral Acoustic Demonstration Center (LADC) conducted a calibration experiment to measure the acoustic energy output of an industrial seismic exploration array in a frequency range up to 25 kHz. A standard acoustic propagation model, RAM (Range-dependent Acoustic Model by M. Collins), was adapted to model the broadband (up to 1000 Hz) waveguide transfer function for a 21-element moving seismic exploration source array generating acoustic data. Notional source signatures for each airgun in an array were generated by the calibrated airgun modeling package GUNDALF, which is based on original theoretical work by Ziolkowski, Hatton, Laws, and others. The package allows modeling of the very close bubble interactions from each airgun in an array. Experimental and simulated data demonstrate good agreement in

the frequency range up to 300 Hz. Factors responsible for the discrepancies between measured and modeled data in the higher frequency region are discussed. Environmental implications of variations in the acoustic energy distribution with seasonal and geographic changes in the propagation environment are addressed. [Research sponsored by the Industry Research Funding Coalition through the International Association of Geophysical Contractors.]

9:00

4aAO3. Arrival time analysis for seismic airgun data classification.

James Stephens (Dept. of Phys. & Astron., Univ. of Southern Mississippi, 6971 Lincoln Rd., Hattiesburg, MS 39402)

Survey data were collected in order to characterize the output of a marine seismic airgun array for various propagation geometries. Near uniform sound speed in the survey area allowed for classification of first-break and secondary arrivals as direct arrival and either reflected or refracted path arrival, respectively, using simple propagation geometries. Plots of initial arrival versus horizontal offset between the airgun and receiver allow classification of the arrival as via direct, reflected, or refracted path in a manner similar to that employed in the analysis of seismic reflection data in the shot domain. Beyond a certain horizontal range (approximately 4.5 km), the critically refracted arrival overtakes the reflected arrival, affording an estimate of the propagation speed in the sea-floor layer. Analysis of this sort allows separation of the direct arrival for analysis purposes.

9:15

4aAO4. Performance variability of matched-field processors on a shelf-break region in the Mid-Atlantic Bight.

Young-Nam Na (Agency for Defense Development, Chinhae, Kyungnam, ROK 645-600), Peter C. Mignerey, and Bruce H. Pasewark (Naval Res. Lab., Washington, DC 20375-5350)

An acoustics experiment (RAGS) was conducted by NRL in December 2003 in the Mid-Atlantic Bight. The deployed equipment included two sources broadcasting 300- and 500-Hz cw signals and three vertical arrays distributed at 10-km intervals over a 30-km range. The water depth varied between 63-125 m. Temperature profiles were obtained from sensors attached to the vertical arrays. Fourier analysis of temperature and vertical array data show that the region is dominated by the M2 tide, the effects being strongest up slope. Several storms significantly influenced the received acoustic fields. Simulations, using several matched-field processors, show significant mismatches between replica and data fields are introduced by changes in the water depth and the sound speed profiles. Just 1 m of tidal depth changes cause significant phase differences between replica and data fields that induce nearly 20-dB variation in matched-field processor output. The study will apply several existing matched-field processors to the measured data with modeled replicas and discuss source

localization results. The discussion will focus on the inevitable degradation of processors that utilize coherence between replica and measured data in shallow waters. [Work supported by the U.S. ONR and the R.O.K. Agency for Defense Development.]

9:30

4aAO5. Littoral tidally induced environmental influences on horizontal array performance. Roger Oba, Steven Finette (Acoust. Div. Naval Res. Lab., Washington, DC 20375), Colin Shen, and Thomas Evans (Naval Res. Lab., Washington, DC 20375)

A modeled continental shelf-break region is used to assess the combined and relative impact of bathymetry and internal wave/bore influences on horizontal array performance in a littoral environment. Tidal forcing over a smooth break varying from a maximum depth of 100-m flat bathymetry to 50 m is used in conjunction with a submesoscale hydrodynamic model to produce internal wave activity in a fully stratified water column. A continuous wave source (100 Hz, for example) is sited at different aspects relative to the shelf break, and the acoustic field is propagated to broadside arrays that are placed at various bearings. A three-dimensional parabolic equation code propagated the field to the arrays. Time-variable three-dimensional acoustic effects such as beam wander due to horizontal refraction, resulting from either slope or internal waves, are both illustrated. The time-dependent bathymetric effects can be either enhanced or reduced by the internal tidal activity, depending upon the relative geometry of propagation, bathymetry, and internal tide. The influence of bathymetry versus the full evolving environment due to hydrodynamic action can be compared using array correlation calculations, signal gain degradation, and beam wander. [This research is sponsored by the ONR.]

9:45

4aAO6. Environmentally associated signal-to-interference variability in low-frequency active sonar in Korea Strait. P. G. Cable (BBN Technologies, 11 Main St., Mystic, CT 06355) and W. M. Carey (Boston Univ., Boston, MA 02215)

An analysis has been conducted of spatial and temporal variability of target echo-to-interference measurements made during Area Characterization Test III (ACT III) in 1995 in the Korea Strait. The measurements were made over a 5-day period for five fixed bistatic geometries (spatial scale of order 200 square km) using explosive sources, bottom-mounted horizontal receiving hydrophone arrays, and passive reflector targets in 100-m-depth water under downward-refracting acoustic conditions. The bottom at the ACT III site was nearly flat, of sand-silt composition. Meteorological conditions were relatively calm over the conduction of the measurements, but the main proximate oceanographic feature, the South Korean Coastal Front, did result in a relatively varied and complex sound-speed environment. The signal-to-interference fluctuation statistics associated with a single ping was determined to give a standard deviation of order 1 dB, while the observed variability of signal-to-interference over the spatial and temporal scales of the measurements gave a standard deviation of about 2 dB. The observed variability of received signal-to-interference will be discussed in terms of physical causes and measurement error. [Work supported by ONR Code OA321.]

10:00–10:20 Break

10:20

4aAO7. A review of spatial coherence and array signal gain results. William M. Carey (College of Eng., Boston Univ., Boston, MA 02215)

A question discussed over the last three decades is: "how does scattering in the deep and shallow-water-sound channels limit coherent array processing?" In particular, for the horizontal array, what is the limitation of resolution in terms of the mutual coherence function and its coherence length? For frequencies (400 Hz) the measurement of the magnitude-squared coherence in a multipath environment with a partly coherent-dynamic noise field is difficult. C. Carter has determined that the measurement of phase coherence in the presence of noise is a poor statistical

estimator since it is a function of the signal-to-noise ratio with large confidence bounds [IEEE Trans. Audio Electroacoust. **AU21**, 388–389 (1973)]. However, array signal gain and a theoretical coherence functional form can also provide a measure of the horizontal coherence length. This paper reviews single path coherence results and those derived from array measurements and shows that representative deep-water lengths for a frequency of 400 Hz are 100 wavelengths at 400 km with a frequency scaling of the 5/2 power while for shallow water they are 30 wavelengths at 40 km. These numbers will be discussed in terms of internal wave scattering theories.

10:35

4aAO8. Calculation of fine-scale sound-field structure based on towed CTD (conductivity, temperature, and depth) chain measurements. Joy E. Lyons, David L. Bradley, and R. Lee Culver (Grad. Program in Acoust., Penn. State Univ., State College, PA 16804, jel246@psu.edu)

In July 2002, an experiment was conducted off the coast of southern California in which the two-dimensional temperature and salinity field was measured directly using a towed CTD chain. The sensor spacing was smaller (2 to 8 m) near the surface (above 30 m) in order to better resolve the mixed layer and thermocline; the depth of the lowest sensor was about 150 m. The path of each tow was back and forth across the same section of ocean such that ocean dynamic features observed in the CTD data were recorded in expanded and compressed states depending on the ship's heading. The range of each run was a few kilometers, with data taken in 1-s increments. This yielded a high-resolution field suitable for analyzing the propagation behavior of high frequencies. Using the fine-scale measurements of the sound-speed field and an acoustic propagation model based on an FFT-realized parabolic equation, the sound field was calculated reflecting the observed ocean dynamic features. Modeling of scintillation in acoustic intensity will be discussed in the context of spic theory. [Work supported by ONR Code 321US.]

10:50

4aAO9. A hydro-acoustic source model in calculating noise field generated by breaking waves. Xuemei Chen, Steven L. Means, William G. Szymczak (Acoust. Div. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, xchen@wave.nrl.navy.mil), and Joel C. W. Rogers (Polytechnic Univ., Brooklyn, NY 11201)

To develop a complete model for the breaking wave noise, it is necessary to relate the source quantities to the physical parameters of the wave-breaking and noise-generation processes. In this paper, the source structure of an individual breaking wave is simulated using a coupled hydro-acoustic model, which incorporates the physical processes underlying the mechanisms of the generation of the noise. The physical processes of wave formation and breaking are modeled using a generalized hydrodynamics formulation, providing the hydrodynamic parameters, such as pressure variations and air cavity shapes, etc., for the acoustic calculation. In the acoustic simulation, an algorithm has been developed in handling wave propagation in irregular regions, such as the bubbly liquid generated by wave-breaking. For the noise field modeling, the locations and occurrence times for the individual breaking waves are specified as stochastic quantities using Poisson simulations and the total noise field is calculated as the superposition of the noise contributed by those breaking waves. [Work supported by ONR.]

11:05

4aAO10. Sound field temporal fluctuations due to mode coupling in shallow water with internal soliton. Mohsen Badiy (Univ. of Delaware, Newark, DE 19716), Valery Grigorev, Boris Katsnelson (Voronezh Univ., Voronezh, 394006, Russia), and James Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Sound field temporal fluctuations due to mode coupling, resulting from traveling internal solitons (IS) approximately along the acoustic track is addressed. The spectrum of received signal amplitude fluctuations in the range of very low frequency ($\sim F = 0.001-0.01$ Hz) is obtained. Maxi-

mums in this spectrum, correspond to the times which IS passes the sound field ray cycles. A relationship between F-spectrum and acoustic frequency is illustrated by showing Frequency-frequency ($F-f$) diagram. It is shown that depending on the sound field frequency (f), different mode pairs give the most pronounced contribution to the sound intensity fluctuations. At the same time invariant frequency, F , of maximal fluctuations remains approximately the same for all sound frequency (f) bands. For example in the case of the SWARM95 experiment, F is about 0.0015 Hz for sound frequency, f in the range of 30–150 Hz. Generally, F depends on waveguide parameters and velocity of IS. In this paper, the physical meaning of the invariant frequency and its relationship to the sound frequency is discussed. In addition, the experimental setup for observation of F in future shallow-water acoustic experiments is proposed. [Work supported by CRDF, PFBR and ONR-3210A.]

11:20

4aAO11. Environmental influence on acoustic impulse responses across the New Jersey Shelf in the winter. Peter C. Mignerey (Acoust. Div. 7120, Naval Res. Lab., Washington, DC 20375-5350)

In December 2003 as part of the RAGS03 experiment, the Naval Research Laboratory moored three vertical arrays at ranges of 10, 20, and 30 km distant from fixed 300- and 500-Hz LFM acoustic sources. Data were recorded continuously for approximately 20 days, which covered periods of severe storms and various phases of internal tides. Associated temperature profile data show the internal tide arriving as either an internal jump or series of elevation waves superimposed on a bottom layer associated with the foot of a shelf-break front. Impulse responses derived from acoustic

data received by the vertical arrays provide insight into the acoustic modal structure within the ocean for various sea states and phases of the internal tide. Results to date show significant loss of high modes during those time periods when either the internal tide is present or the sea state is high in comparison with periods of low sea state and absent internal tide. The extent of mode stripping increases with down-slope range. Apparently high acoustic modes are being scattered and stripped by near-boundary interactions. Evidence for these interactions will be presented along with a discussion of stripping processes. [Work supported by ONR.]

11:35

4aAO12. Acoustic sampling volume of a bistatic sonar. Kenneth G. Foote (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

The expression for the acoustic sampling volume of a monostatic sonar [K. G. Foote, J. Acoust. Soc. Am. **90**, 959 (1991)] is generalized to the case of a bistatic sonar, with separate, non-collocated transmitting and receiving transducers. The bistatic sampling volume is the integral of a counting function over the physically accessible space. The argument of the counting function, a Heaviside step function, is the difference between the scattering amplitude, or other measure of scattering strength, and a noise-dependent threshold. When this difference is positive, the integrand is unity; when zero, one-half; when negative, zero. Both the sampling volume and a related quantity, the equivalent beam angle, are evaluated numerically for several bistatic geometries, transducer shapes, and narrow-band frequencies for the cases of point scatterers and directional scatterers. The application of the bistatic acoustic sampling volume to the quantification of suspended particulate matter and aquatic organisms is discussed.

THURSDAY MORNING, 20 OCTOBER 2005

CARVER ROOM, 8:15 A.M. TO 12:00 NOON

Session 4aBB

Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Acoustic Radiation Force Methods for Medical Imaging and Tissue Evaluation

Mostafa Fatemi, Chair

Mayo Clinic, Physiology Biophysics, 200 First St., SW, Rochester, MN 55905

Chair's Introduction—8:15

Invited Papers

8:20

4aBB1. Principles of radiation force. Oleg Rudenko (Blekinge Inst. of Technol., Karlskrona, Sweden and Moscow State Univ., Russia, rudenko@acs366.phys.msu.ru)

A brief review of general problems of radiation force (RF) is given. RF was discovered by Lord Rayleigh in 1902 and further fundamental contributions to the RF phenomenon were made by Brillouin and Langevin. In spite of century-old history, RF has not been studied exhaustively and there are many grey areas. The difference between the Rayleigh and Langevin RF appearing in fluids, the difference between averaging procedure in Lagrange and Euler representations, the connections between mean characteristics of the RF-induced streaming (velocity, pressure, mass transfer) and acoustic parameters are still under discussion. Everyone knows that RF is produced by a change in the density of ultrasonic wave energy due to the absorption or due to the reflection from obstacles and inhomogeneous inclusions. However, can RF appear in lossless homogeneous medium or not? The pumping effect of sound beam caused by hydrodynamic nonlinearity was first discussed more than 50 years ago, and similar effects in biomedical applications RF produced by high-intensity focused ultrasound are being discussed now. The talk reviews recent progress in investigation of the problems concerning RF in fluids, biological tissues, and solids with account for specific properties of wave field like nonlinearity, diffraction, absorption, focusing, and scattering.

4aBB2. Generation of ultrasound radiation force with the use of time reversal acoustics principles. Armen Sarvazyan and Alexander Sutin (Artann Labs., Lambertville, NJ 08530, armen@artannlabs.com)

There are numerous medical applications of ultrasound radiation force (RF) which could be made more effective using the time reversal acoustics (TRA) principles. This paper gives an overview of research into physical and technical bases of RF generation in heterogeneous biological media using TRA focusing systems. A custom-designed compact multichannel TRA system for receiving, digitizing, storing, time reversing, and transmitting acoustic signals in a wide frequency range from 0.01 to 10 MHz has been developed and extensively tested in model systems and *ex vivo* tissues and bones. Shear strain and shear waves remotely induced in soft tissues and bones by radiation force were detected using various acoustical and optical means. Experimental studies fully confirmed the feasibility of TRA generation of RF and demonstrated several advantages over conventional means of remotely inducing shear stress in biological media. These advantages include a possibility to create highly localized (close to diffraction limit) shear stress in heterogeneous media stir focused ultrasound beam in 3-D volume using very simple hardware. [Work supported by NIH grant.]

9:10

4aBB3. *In vivo* noninvasive method for measuring local wave velocity in femoral arteries of pig. Xiaoming Zhang, Randall Kinnick, Cristina Pislaru, Mostafa Fatemi, and James Greenleaf (Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905)

We have proposed generating a bending wave in the arterial wall using ultrasound radiation force and measuring the wave velocity along the arterial wall [Zhang *et al.*, IEEE Trans. Ultrason. Ferroelectr. Freq. Control **52**, 642–652 (2005)]. Here, we report the results of *in vivo* studies on pigs. The pig was anesthetized, and a micromanometer tip catheter was inserted into the femoral artery to measure luminal pressure. A water bath was created on the animal's groin to allow unimpeded access of the ultrasound beams to the femoral artery. The femoral artery was first located using a 13-MHz linear-array transducer. Then, a vibro-acoustography image was obtained to ensure precise positioning of the excitation force relative to the artery. The artery was excited by the force transducer and the resulting vibration of the arterial wall was measured by a sensing Doppler transceiver. Measured wave velocity was 3.1 m/s at 300 Hz. With this new method wave velocity over a distance of 5 mm, and therefore stiffness of arteries, can be measured locally and non-invasively. Measurement time is short in a few tens of milliseconds, which allows pressure dependence and pharmacological effect on the wall properties to be measured at different cardiac times.

9:35

4aBB4. Vibro-acoustography: Recent advances and future promises. Mostafa Fatemi, Azra Alizad, and James F. Greenleaf (Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, fatemi@mayo.edu)

Vibro-acoustography is an imaging modality that uses ultrasound as the source of energy to explore objects at two ends of the spectrum. Employing two ultrasound beams at slightly different frequencies, this technique produces a highly localized radiation stress field that vibrates the object at the difference frequency. In response, the object produces a secondary acoustic field. This sound, which contains information about the dynamics of the object at the low difference frequency as well as objects characteristics at the ultrasound range, is used to produce an image of the object. This combination makes vibro-acoustography a unique tool to study biological tissues at low and high frequencies simultaneously. Utility of this method has already been demonstrated for a wide range of applications *in vitro*. *In vivo* applications of vibro-acoustography have been demonstrated in a recent study on human breast. It has been shown that this technique can produce high quality *in vivo* images at ultrasound intensities within the recommended FDA limits. Vibro-acoustography may also be used for imaging solids, such as metallic implants. This paper describes recent achievements in vibro-acoustography as well as possible future role of this technology in medicine. [This abstract includes materials patented by MF and JFG.]

10:00–10:30 Break

Contributed Papers

10:30

4aBB5. A bioreactor for the dynamic mechanical stimulation of vocal-fold fibroblasts based on vibro-acoustography. Roger W. Chan (Otolaryngol. Head & Neck Surgery, Grad. Program in Biomed. Eng., Univ. of Texas Southwestern Medical Ctr., Dallas, TX 75390-9035, roger.chan@utsouthwestern.edu) and Maritza Rodriguez (UT Southwestern Medical Ctr., Dallas, TX 75390)

During voice production, the vocal folds undergo airflow-induced self-sustained oscillation at a fundamental frequency of around 100–1000 Hz, with an amplitude of around 1–3 mm. The vocal-fold extracellular matrix (ECM), with appropriate tissue viscoelastic properties, is optimally tuned for such vibration. Vocal-fold fibroblasts regulate the gene expressions for key ECM proteins (e.g., collagen, fibronectin, fibromodulin, and hyaluronic acid), and these expressions are affected by the stress fields experi-

enced by the fibroblasts. This study attempts to develop a bioreactor for cultivating cells under a micromechanical environment similar to that *in vivo*, based on the principle of vibro-acoustography. Vocal-fold fibroblasts from primary culture were grown in 3D, biodegradable scaffolds, and were excited dynamically by the radiation force generated by amplitude modulation of two confocal ultrasound beams of slightly different frequencies. Low-frequency acoustic radiation force was applied to the scaffold surface, and its vibratory response was imaged by videostroboscopy. A phantom tissue (standard viscoelastic material) with known elastic modulus was also excited and its vibratory frequency and amplitude were measured by videostroboscopy. Results showed that the bioreactor was capable of delivering mechanical stimuli to the tissue constructs in a physiological frequency range (100–1000 Hz), supporting its potential for vocal-fold tissue engineering applications. [Work supported by NIH Grant R01 DC006101.]

10:45

4aBB6. Modulated acoustic radiation pressure and stress-coupling projections. Philip L. Marston and David B. Thiessen (Phys. and Astron. Dept., Washington State Univ., Pullman, WA 99164-2814)

Low-frequency deformation can be induced at a single frequency using radiation stress oscillations of double-sideband suppressed-carrier ultrasound [P. L. Marston and R. E. Apfel, *J. Acoust. Soc. Am.* **67**, 27 (1980)]. The transducer voltage is proportional to a product of low- and high-frequency sine waves. To anticipate the shape and magnitude of induced deformations, it is helpful to expand the distribution of the radiation stress on the object to be deformed as a series of projections [P. L. Marston, *J. Acoust. Soc. Am.* **67**, 15 (1980)]. Stress projections are also useful for unmodulated waves: the radiation force is an example. In addition to spherical and nearly spherical objects, recent experiments and calculations have concerned cylindrical objects [S. F. Morse, D. B. Thiessen, and P. L. Marston, *Phys. Fluids* **8**, 3 (1996); W. Wei, D. B. Thiessen, and P. L. Marston, *J. Acoust. Soc. Am.* **116**, 202 (2004)]. In standing waves the following projections are nonvanishing in the low acoustic frequency limit for appropriately situated dense objects: radial projection [M. J. Marry-Lyon, D. B. Thiessen, and P. L. Marston, *Phys. Rev. Lett.* **86**, 2293 (2001)] and quadrupole projection [P. L. Marston *et al.*, *J. Acoust. Soc. Am.* **69**, 1499 (1981)].

11:00

4aBB7. Non-dissipative mechanisms of radiation force and shear wave generation. Lev Ostrovsky (Zel Technologies/NOAA ETL, Boulder, CO 80305), Yurii Ilinskii (Univ. of Texas, Austin, TX 78713), Armen Sarvazyan, and Alexander Sutin (Artann Labs., Lambertville, NJ 08530)

We describe new mechanisms of shear stress generation in tissue by radiation force (RF) that are not related to attenuation or reflection of ultrasound waves. The suggested theoretical model is based on the five-constant theory for elastic solids that is extended to waterlike biological media. It is shown that in the absence of dissipation, a potential and linear sound beam in a homogeneous medium creates a potential radiation pressure rather than the shear stress. However, in inhomogeneous non-dissipative media, that is in a lossless medium with spatially varied linear and/or nonlinear parameters, the shear stress generation by this RF is possible. Additional non-dissipative mechanism is associated with the anharmonicity of the primary beam. Although the corresponding effect is proportional to the third power of the beam amplitude, it can be significant in the focal area. Estimates demonstrate that the contribution of the above mechanisms to the value of the generated RF can be comparable with and even exceed the classical dissipative radiation force in realistic situations. Experiments with inhomogeneous tissue mimicking phantoms are presented and experimental results are interpreted in the framework of the above theory. [Work partly supported by NIH grant.]

11:15

4aBB8. Radiation force of ultrasound as shear wave source in microscopic magnetic resonance elastography. Shadi F. Othman, M. Bulent Ozer, Huihui Xu, Thomas J. Royston (Univ. of Illinois at Chicago, 842 West Taylor St., MC 251, Chicago, IL 60607, troyston@uic.edu), and Richard L. Magin (Univ. of Illinois at Chicago, Chicago, IL 60607)

Microscopic magnetic resonance elastography (micro-MRE) is a high-resolution imaging technique for measuring the viscoelastic properties of small synthetic and biological samples. Taking MRE to the microscopic scale requires stronger static fields, stronger magnetic field gradients, higher performance RF coils, and more compact, higher frequency shear wave actuators. Prior work by our group has been conducted at 11.74 T. A

needle attached to a vibrating cantilever beam was placed in contact with the surface of the sample to generate shear waves up to 800 Hz. At higher frequencies, the excited shear waves attenuate within an extremely short distance such that only a very small region in the vicinity of the actuator can be studied due to inherent dynamic range limitations. In principle, modulated radiation force of US should be able to create a localized shear wave source within the test sample at a distance from the US transducer, thereby enabling micro-MRE probing of the sample at very high frequencies (up to 5 kHz). A confocal US transducer was fabricated to create such a source within the working constraints of the micro-MRE system. Initial feasibility studies are reviewed in this presentation. [Research supported by NIH Grant No. EB004885-01.]

11:30

4aBB9. Finite-difference time-domain approach to acoustic radiation force problems. Glauber T. Silva (Departamento de Tecnologia da Informacao, Universidade Federal de Alagoas, Maceio, AL, Brazil 57072-970)

Acoustic radiation force plays a major role in elastography methods such as vibro-acoustography, acoustic radiation force, shear wave elasticity, and supersonic shear wave imaging. The radiation force (dynamic or static) exerted on an object by an incident wave can be obtained by solving the acoustic scattering problem for the object. However, only in rather simple cases the scattering of waves can be described by exact analytical expressions. In this work, we developed an algorithm based on the finite-difference time-domain (FDTD) method to compute the radiation force exerted on arbitrary shaped objects. The algorithm simulates the wave propagation in a finite extended medium with an embedded object. The radiation force is obtained by numerically calculating a surface integral of the momentum flux, which depends on the incident and scattered fields. Absorbing boundary conditions are used to truncate the medium. We compute the radiation force exerted on a rigid and soft cylinder by a plane wave. Results are in agreement with the theoretical predictions. Discrepancies due to numerical dispersion in the algorithm are under investigation. The presented method might be used to calculate the radiation force on complex objects present in elastography techniques. [Work supported by FAPEAL/CNPq, Brazil.]

11:45

4aBB10. Stress field formation for multifrequency vibro-acoustography: A simulation study. Matthew Urban, Mostafa Fatemi, and James Greenleaf (Mayo Clinic and Foundation, 200 First St. SW, Rochester, MN 55905)

Vibro-acoustography is a method that uses the dynamic radiation force (stress) of ultrasound to locally excite an object at low frequency [Fatemi and Greenleaf, *Science* **280**, 82]. A multifrequency vibro-acoustography method is proposed that uses a multifrequency radiation stress produced by an array transducer driven with N ultrasound frequencies. A multifrequency image at the different $N(N-1)/2$ frequencies is produced with one scan of the region of interest, increasing the information yield by a factor of $N(N-1)/2$. Image formation theory is presented for multifrequency vibro-acoustography using an annular array transducer. Simulations of the stress field were performed for a 20-element, 3.5-MHz annular array transducer. Four ultrasound frequencies produce six unique low-frequency components in the point spread function. Computer phantoms with small spheres are used to demonstrate the usefulness of multifrequency vibro-acoustography for microcalcification detection in breast imaging. The lateral and axial width of the main lobe at -6 dB is 0.68 and 6.47 mm, respectively. The lateral and axial sidelobe levels are -27.8 and -16.8 dB, respectively. The proposed method holds the potential for large gains of information with good spatial resolution and no increase in scanning time. [Work supported by Grant EB00535-03 from NIH.]

Session 4aID**Interdisciplinary: Electronic Tools and Services for ASA Authors and Readers**

Allan D. Pierce, Chair

*Boston Univ., Aerospace and Mechanical Engineering, 110 Cummington St., Boston, MA 02215***Chair's Introduction—9:00**

Many powerful tools and services are available to authors and readers who publish in and use *The Journal of the Acoustical Society of America* and *Acoustics Research Letters Online*. This seminar will feature presenters from the American Institute of Physics who host the online versions of Acoustical Society journals.

9:05–10:00**Tools and Services for Authors**

A brief overview of the online manuscript submittal process for the ASA journals using the all-electronic paper management system, Peer X-press.

10:00–11:00**Tools and Services for Readers**

A demonstration of research tools available through AIP's online platform, "Scitation," including searchable databases, reference linking features, article collection folders, citation exports, email alerts, and RSS data feeds to assist readers who access ASA and AIP journals.

Session 4aMU**Musical Acoustics and Architectural Acoustics: Acoustics of Choir Singing I**

Sten O. Ternstrom, Cochair

Kungliga Tekniska Hogskolan, Speech Music and Hearing, Lindstedtsvagen 24, S-10044 Stockholm, Sweden

Thomas D. Rossing, Cochair

*Northern Illinois Univ., Physics Dept., DeKalb, IL 60115***Chair's Introduction—10:00*****Invited Papers*****10:05**

4aMU1. Multitrack analysis of amateur and professional choirs. Harald Jers (Franzstrasse 33, D-50931 Cologne, Germany, harald.jers@gmx.de)

Many choir singers in the world know the fascinating phenomenon of choir sound, which is the result of multiple voices singing in an ensemble. This so-called chorus effect, where the normal mechanisms of auditory localization of the single voices are disrupted, may be caused by complex interactions between the choir singers, but has not been researched in detail. Each singer of an amateur and a professional vocal ensemble of 16 singers was recorded on separate tracks while singing in the choir. The evaluation of different choir pieces and exercises provided information and predictions about F_0 , SPL, timing/synchronization, vibrato behavior, and the produced choir sound. The results reveal differences between amateur and professional choirs for homophonic and polyphonic choir pieces, and suggest new considerations for choir rehearsals and concert performances.

4aMU2. The effects of choir spacing and choir formation on the tuning accuracy and intonation tendencies of a mixed choir.

James F. Daugherty (Div. of Music Education and Music Therapy, The Univ. of Kansas, 1530 Naismith Dr., Ste. 448, Lawrence, KS 66045, jdaugher@ku.edu)

The tuning accuracy and intonation tendencies of a high school mixed choir ($N=46$) were measured from digital recordings obtained as the ensemble performed an *a cappella* motet under concert conditions in $N=3$ singer spacing configurations (close, lateral, circumambient) and $N=2$ choir formations (sectional and mixed). Methods of analysis were modeled on Howard's (2004) pitch-based measurements of the tuning accuracy of crowds of football fans. Results were discussed in terms of (a) previous studies on choir spacing (Daugherty, 1999, 2003) and self-to-other singer ratios (Ternström, 1995, 1999); (b) contributions of choir spacing to vocal/choral pedagogy; and (c) potential ramifications for the design and use of auditoria and portable standing risers for choral performances.

Contributed Papers

10:55

4aMU3. Directivity of singers. Harald Jers (Franzstrasse 33, D-50931 Cologne, Germany, harald.jers@gmx.de)

Studies of acoustical balance between singers within a choir by means of room acoustical measurements have shown that the directional sound propagation of the source is important. For this reason the directivity of female and male singers for different vowels has been measured in this investigation. Measurements of a pilot study and some first measurements in 1998 have been supplemented with new measurements and an enhanced setup. A special measurement setup with reference and recording microphones was used to collect the directivity data. A resolution of 10 deg for azimuth and elevation angle was obtained. The results will be shown in 3D spherical plots with frequency adjustments in semitones from 80 to 8000 Hz. The measurements are compared to an artificial singer's directivity,

Note: Session 4pMU, Acoustics of Choir Singing II, will conclude at Central Lutheran Church with a panel discussion and a short Concert by the St. Olaf Cantorei

and the influence of a sheet music binder in front of a singer will be shown. The results give information on the directivity of singers and are relevant for the prediction of self-to-other-ratios that result from placement and formation aspects within a choir.

11:10

4aMU4. Auditorium design for choral performance. Timothy Foulkes and Christopher Storch (Cavanaugh Tocci Assoc. Inc., 327F Boston Post Rd., Sudbury, MA 01776)

Design for a 500 seat recital hall to support an award winning high school choral program is discussed. Acoustic design strategy, important acoustic parameters (calculated and measured), and photos of the completed project are reviewed.

THURSDAY MORNING, 20 OCTOBER 2005

CONRAD A, 9:00 TO 11:00 A.M.

Session 4aPA**Physical Acoustics: Topics in Atmospheric Acoustics**

Roger M. Waxler, Chair

Univ. of Mississippi, National Center for Physical Acoustics, 1 Coliseum Dr., University, MS 38677

Contributed Papers

9:00

4aPA1. Investigation of two-dimensional numerical shock propagation using a weighted essentially non-oscillatory scheme. Mark S. Wochner and Anthony A. Atchley (Grad. Prog. in Acoust., Penn State Univ., University Park, PA 16802)

This work is focused on the computational simulation of realistic jet noise propagation in air. Consequently, in addition to accurate absorption, dispersion, wave steepening, etc., propagation of multidimensional shocks is necessary. Time-domain propagation of shocks requires some special technique to ensure stability and this is usually accomplished using artificial viscosity. In this investigation, instead of using artificial viscosity, a weighted essentially non-oscillatory (WENO) scheme is used. The WENO scheme is unique in that it uses a combination of multiple stencils to calculate derivatives. Shock propagation is achieved in this method through its use of smoothness indicators, which work to leave out any candidate stencils that contain a discontinuity in the calculation of derivatives. Its ability to handle a wide variety of acoustical conditions and the future potential of such a model is shown. [Work supported by ONR and the Pennsylvania State University Applied Research Laboratory Exploratory and Foundational Program.]

9:15

4aPA2. Modeling volcanic infra-sound propagation. Joseph F. Lingeitch, Geoffrey F. Edelman, Douglas P. Drob, and Michael D. Collins (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, jfl@nrl.navy.mil)

Active volcanoes are significant sources of infra-sound and can be exploited for studying atmospheric sound. The propagation is influenced by the vertical and horizontal wind structure where wind-induced refraction can trap or disperse sound energy, leading to large variations of propagation with wind conditions. Stratospheric winds influence long-range observations and exhibit variations on seasonal, diurnal, and semi-diurnal time scales. These effects impact the observability and interpretation of acoustic signals at remote listening stations. The propagation of infra-sound over propagation paths of several hundred kilometers from volcanoes in the Archipelago of Vanuatu in Oceania is studied over a frequency range of 0.1–4 Hz. Atmospheric wind and sound-speed profiles constructed every 6 h for a 2-year period are used as inputs to a parabolic equation propagation model. The results are compared with experimental observations and other models to assess the validity of our numerical simulations. [Work supported by ONR.]

9:30

4aPA3. Atmospheric sound propagation at higher altitudes. Xiao Di, Kenneth E. Gilbert (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677), and Richard Clark (Millersville Univ., Millersville, PA 17551)

During the daytime, upward refraction dominates atmospheric sound propagation. An elevated sensor can potentially give much higher sound-pressure levels and higher signal coherence. To understand daytime upward atmospheric sound propagation at higher altitudes, a comprehensive sound propagation experiment for receivers at various heights (up to 220 m) was performed in November 1998. In the experiment, both meteorological data and acoustic data were taken. Three single tone sources were used in the experiment (210, 380, and 600 Hz). The horizontal propagation distances were 300, 600, and 900 m. In this presentation, we will present the experimental results and calculations of sound-pressure level and coherence using the measured meteorological data as input. Numerical calculations are compared with the experimental data. Both experimental data and theoretical predictions show much higher sound-pressure levels and higher coherence at higher altitudes.

9:45

4aPA4. The pressure distribution around foam spheres outdoors. Jeremy Webster, Richard Rasket, and Jiao Yu (Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677)

A previous investigation of the pressure fluctuations around foam windscreens outdoors indicated both lower pressure levels and lower correlations between microphones than expected from wind noise theory [J. Acoust. Soc. Am. **116**, 2517]. The measurement set was limited and suffered from uncertainty in the incoming wind direction. In this paper we show results from several new measurements which use a three-axis sonic anemometer to establish the wind direction and speed simultaneous to pressure correlation measurements. [Prepared in part through collaborative participation in the Collaborative Technology Alliance for Advanced Sensors sponsored by the U.S. Army Research Laboratory.]

10:00

4aPA5. Performance comparison of compact cylindrical and spherical windscreens. Qamar A. Shams, Allan J. Zuckerwar, and Bradley S. Sealey (NASA Langley Res. Ctr., M.S. 238, Hampton, VA 23681)

An improved and compact windscreen was conceived for a microphone of a type used outdoors to detect atmospheric infrasound from a variety of natural and artificial sources. This cylindrically shaped compact windscreen, made of closed-cell polyurethane foam, had dimensions of 0.0762 m i.d. \times 0.2286 m height \times 0.0127 m wall ($3 \times 9 \times 0.5$ in.). The low acoustic impedance of the foam showed a transmission coefficient near unity. The same closed-cell polyurethane foam was used to fabricate a spherical windscreen of 0.254 m (10 in.) diameter with 0.0127-m (0.5 in.) wall thickness. In this paper the effectiveness of cylindrical and spherical windscreens is evaluated using a low-speed wind tunnel facility that enables controlled and repeatable experiments. Wind noise levels are characterized for both windscreens as a function of frequency and mean wind velocity.

10:15

4aPA6. Soaker hose versus compact nonporous windscreen: A comparison of performance at infrasonic frequencies. Allan J. Zuckerwar, Qamar A. Shams (NASA Langley Res. Ctr., M.S. 238, Hampton, VA 23681, Qamar.A.Shams@nasa.gov), Krish K. Ahuja, and Robert Funk (Georgia Inst. of Technol., Atlanta, GA 30332)

A compact nonporous windscreen described previously [J. Acoust. Soc. Am. **114**, No. 4, Pt. 2, 2323 (2003)] was tested in the field against a soaker hose array to compare performance at infrasonic frequencies. The cylindrically shaped compact windscreen, made of closed-cell polyure-

thane foam, had dimensions 0.0762 m i.d. \times 0.2286 m height \times 0.0127 m wall ($3 \times 9 \times 0.5$ in.). The low acoustic impedance of the foam permits the propagation of infrasound through the walls of the windscreen with a transmission coefficient near unity. The soaker hoses were 15.24-m (50 ft.) long and coupled to a Chaparral model 5 low-frequency microphone. The hose plenum was removed and replaced with the compact windscreen for testing. A sonic boom simulator, located at a distance of 400 m (1/4 mile) from the microphone, generated tones at 3, 4, 5, and 6 Hz. Analysis of the signals received by the interior microphone revealed that the tones transmitted through the windscreen, as well as the reduction of background noise due to naturally occurring wind-generated turbulence, were nearly identical for the two types of windscreen.

10:30

4aPA7. Low-frequency ground impedance from the surface mode. Roger Waxler, Carrick L. Talmadge, and Kenneth E. Gilbert (The Natl. Ctr. for Physical Acoust., 1 Coliseum Dr., University, MS 38677)

At the last meeting of the Acoustical Society of America we presented data and theoretical analysis showing that pulse arrivals in downward refracting atmospheres have a universal low-frequency tail. This tail was identified with the surface mode. In this presentation we discuss the possibility of obtaining low-frequency ground impedance from the surface mode arrival. Signals received on a 10-m vertical array are analyzed. The vertical waveform of the surface mode is obtained as a function of frequency and used to infer a ground impedance. The inferred ground impedance is compared to impedances predicted by ground impedance models.

10:45

4aPA8. Outdoor sound propagation effects on aircraft detection through passive phased-array acoustic antennas: 3D numerical simulations. Ivan Roselli, Pierluigi Testa, Gaetano Caronna (Dept. of Tech. Phys., Faculty of Eng., Univ. of Rome "La Sapienza," Via Eudossiana 18 - 00184, Rome, Italy), Andrea Barbagelata, and Alessandro Ferrando (D'Appolonia S.p.A., 16145, Genoa, Italy)

The present paper describes some of the main acoustic issues connected with the SAFE-AIRPORT European Project for the development of an innovative acoustic system for the improvement of air traffic management. The system sensors are two rotating passive phased-array antennas with 512 microphones each. In particular, this study focused on the propagation of sound waves in the atmosphere and its influence on the system detection efficiency. The effects of air temperature and wind gradients on aircraft tracking were analyzed. Algorithms were implemented to correct output data errors on aircraft location due to acoustic ray deviation in 3D environment. Numerical simulations were performed using several temperature and wind profiles according to common and critical meteorological conditions. Aircraft location was predicted through 3D acoustic ray triangulation methods, taking into account variation in speed of sound waves along rays path toward each antenna. The system range was also assessed considering aircraft noise spectral emission. Since the speed of common airplanes is not negligible with respect to sound speed during typical airport operations such as takeoff and approach, the influence of the Doppler effect on range calculation was also considered and most critical scenarios were simulated.

4a THU. AM

Session 4aSA

Structural Acoustics and Vibration: Vibration, Radiation and Scattering

Courtney B. Burroughs, Chair

Noise Control Engineering, 1241 Smithfield St., State College, PA 16801

Contributed Papers

9:00

4aSA1. Analysis of broadband acoustic radiation from periodic structures. Pavel Danilov (Continuum Dynam., Inc., 34 Lexington Ave., Ewing, NJ 08618) and Donald Bliss (Duke Univ., Durham, NC 27708)

Radiation from a 2D finite-length plate with periodic discontinuities modeled by line impedances, and with arbitrary boundary conditions, is considered. Under broadband excitation such structures often exhibit a fairly simple directivity pattern that is nearly symmetric and relatively smooth. The differences between single-frequency and broadband response are analyzed. Due to discontinuities the structure exhibits passband and stop-band behavior, with structural waves decaying in stop bands. This behavior leads to a different energy distribution between the structure and the acoustic field for frequencies in different bands. Acoustic radiation from periodic structures is characterized in terms of Bloch waves. Closed-form formulas for harmonic and broadband response from a single structural Bloch wave are obtained. Total response is recovered by superposition of Bloch waves. A single Bloch wave contains a primary harmonic, corresponding to a long-wavelength, smooth global solution, and higher spatial harmonics induced by discontinuities which describe the short-wavelength local solution. Radiation characteristics are discussed in terms of global and local solutions.

9:15

4aSA2. Numerical simulation of an aircraft style panel excited by a random pressure field. Noah H. Schiller, Chris R. Fuller (Dept. of Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA 24061, nschille@vt.edu), and Randolph H. Cabell (NASA Langley Res. Ctr., Hampton, VA 23681)

This study describes the development of a model that predicts the sound radiation from an aircraft style panel excited by a dynamic pressure field with arbitrary spatial correlation. Eventually, this model will be used to develop distributed feedback control strategies for systems with spatially correlated inputs. For this study, the input pressure field is simulated using an array of point forces on the panel. The excitation at each point is defined in the time domain using the spectral representation method [M. Shinozuka and C.-M. Jan, *J. Sound Vibr.* **25**(1), 111–128 (1972)]. This method generates sample functions that match the spatial and temporal correlation characteristics of the desired pressure field. Three different types of excitations are considered: a turbulent boundary layer excitation, a diffuse field, and a spatially uncorrelated disturbance. This approach could be extended to a variety of other types of spatially correlated stochastic inputs as well. A convergence study is also presented to show how many point forces are required to approximate the spatial characteristics of each type of pressure field. [Work supported by NASA.]

9:30

4aSA3. A boundary-element method using broadband vibrating-wall sources to predict high-frequency interior sound fields produced by wall vibration. Linda P. Franzoni and Tracy A. DuVall (Mech. Eng., Duke Univ., Box 90300, Durham, NC 27708-0300)

In the high-frequency limit, vibrating panels subject to spatially random, temporally broadband forcing are shown to have broadband power and directivity properties than can be characterized by a limited set of parameters, based on numerical simulations. The radiated pressure field is

parametrized in terms of direction, wave speed ratio, panel damping, and dimensionless frequency. A source directivity equation dependent on these variables is presented. The radiation properties of this equation are incorporated to simulate vibrating wall panels in an energy/intensity-based boundary-element method (BEM) developed for the prediction of steady-state, broadband, reverberant sound fields in enclosures having either diffusely or specularly reflecting boundaries. The BEM method uses uncorrelated broadband directional intensity sources to construct the source and reflection sound fields and predict mean-square pressure distributions in enclosures. Because uncorrelated broadband directional intensity sources are used, the system does not require a frequency-by-frequency-based solution, thereby reducing computational expense. Simulations are compared to exact solutions obtained by computationally expensive frequency-by-frequency modal methods. When fully developed, the directed application of this method is aircraft interior noise caused by exterior boundary layer excitation on fuselage panels.

9:45

4aSA4. Sound transmission modeling of advanced multilayered composite structures using enhanced T-matrices for two-phase materials. Roland L. Woodcock and Rebecca S. Bryant (United Technologies Res. Ctr., 411 Silver Ln., M.S. 129-17, East Hartford, CT 06108)

Advanced composite structures have been used for many years in the aerospace industry. When designing multilayered structures special attention must be paid to the bonding techniques since the interface conditions have a direct effect on the mechanical coupling between the individual layers. Previous studies have shown the overall acoustical performance such as transmission loss and surface absorption to be sensitive to this structural path mainly in the lower frequency range. The state of the art shows that a comprehensive model is still lacking in the framework of the transfer matrix Method. The present paper proposes a new analytical modeling approach to tackle systems with stiffeners in the low-frequency range. This technique is based on the so-called Series-Parallel network of the transmission line theory in the framework of the classical electro-acoustical analogies. The simulations show that in the typical case of a double plate with stiffeners, with regards to transmission loss the design change due to the mechanical path is captured and the increase of the overall stiffness of the system shifts the resonance to the higher frequencies. The other important acoustical properties of multilayered systems will be highlighted during the presentation with regards to optimizing the overall acoustical performance.

10:00

4aSA5. Axisymmetric acoustic scattering from submerged prolate spheroidal shells. Jeffrey E. Boisvert (NAVSEA Newport, Newport, RI 02841) and Sabih I. Hayek (Penn State Univ., University Park, PA 16802)

The equations of motion for nonaxisymmetric vibration of prolate spheroidal shells of constant thickness were derived using Hamilton's principle [S. I. Hayek and J. E. Boisvert, *J. Acoust. Soc. Am.* **114**, 2799–2811 (2003)]. The shell theory used in this derivation includes shear deformations and rotatory inertias. The shell displacements and rotations were expanded in infinite series of comparison functions. These include associated Legendre functions in terms of the prolate spheroidal angular

coordinate and circular functions in the azimuthal angle coordinate. The shell is insonified by a plane wave incident along the major axis. The external (heavy) fluid loading impedance was computed using an eigenfunction expansion of prolate spheroidal wavefunctions. Far-field scattered acoustic pressure spectra are presented for several shell thickness-to-half-length ratios ranging from 0.005 to 0.1, and for various shape parameters, a , ranging from an elongated spheroidal shell ($a=1.01$) to a spherical shell ($a\sim 100$). The far-field directivity of acoustic scattering is presented at selected frequencies. [Work supported by the ONR/ASEE Summer Faculty Research Program.]

10:15–10:30 Break

10:30

4aSA6. Local parametric identification for *in situ* health monitoring of aircraft using distributed sensors. Philippe Micheau, Jérôme Pinsonnault, and Patrice Masson (GAUS, Mech. Eng. Dept., Université de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada, Philippe.Micheau@USherbrooke.ca)

The objective of this study is to provide a strategy for *in situ* structural health monitoring with MEMS sensors in order to reduce the high costs associated with periodic prescribed inspections of aircraft. The presented strategy focuses on the flexural waves in the medium-frequency range in order to obtain a good trade-off between damage localization and distant propagation, and to be efficient for composite materials. The method consists of performing local parametric identification of equivalent MISO systems in order to detect a parametric discontinuity when a defect is present. Simulations were performed with a hierarchical trigonometric functions set on an aluminum plate (100 cm \times 75 cm \times 1 mm) with a crack (5 cm \times 0.007 in.). For a chirp excitation in the medium-frequency range at 1 kHz, wave propagation simulation shows the scattering around the crack in the plate. A distribution of MEMS sensors over the plate is efficient to detect a significant change in the identified local parameters, and consequently to localize the crack. However, this approach is limited to configuration with high signal-to-noise ratio. Experimental validation of the model is conducted both in frequency and time domains for healthy and cracked beams and plates. [Work supported by the Consortium for Research and Innovation in Aerospace in Quebec.]

10:45

4aSA7. Investigation of the resonance frequency shift in parts with cracks. Krasimir Zahariev, Yulian Kin (Purdue Univ. Calumet, 2200 169th St., Hammond, IN 46323, kin@calumet.purdue.edu), and Alexander Sutin (Stevens Inst. of Technol., Hoboken, NJ 07030)

It is known that development of crack in various parts leads to resonance frequency variation and that phenomena can be used for crack detection and remaining lifetime prediction. We have investigated this effect on a steel specimen (25 \times 150 \times 6 mm). The crack was initiated at the root of preliminary machined notch and propagated under cycling loading on fatigue machine. The finite-element analysis was applied for calculation of frequency shift for three flexural modes of vibration and it was observed that the frequency shift increases with the increase of crack size. The maximum detected frequency shift was 3.8% for the crack size 23% of a sample width. The experimental measurements were conducted by measurements of sound produced by free vibration of the sample after impact excitation (impact resonance acoustic spectroscopy). The relative difference between acoustical measurements and FEA results did not exceed 1%. The conducted research provides a good basis for development of the remaining life prediction methods, for example, by Paris formulation. Parameters of the formulation for our case were determined experimentally. [Research supported by 21 Century Fund of Indiana.]

11:00

4aSA8. Effects of material anisotropy on the amplitudes of waves in cylindrical structures. J. Gregory McDaniel (Dept. of Aerosp. and Mech Eng., Boston Univ., 110 Cummings St., Boston, MA 02215, jgm@bu.edu)

Composite materials may be designed and configured to substantially alter the generation and propagation of elastic waves with a relatively small weight penalty. The anisotropy introduced by composite structures leads one to intuitively expect that wave amplitudes will depend strongly on the orientations of the applied force as well as the composite material. The present work investigates this effect for cylindrical structures with point forces. Using a mixture of analytic and numerical models, an understanding of the relationship between force directivity, material orientation, and wave amplitudes is developed. Results of this study lead directly to design strategies for controlling power flow to the structure. [Work supported by ONR.]

11:15

4aSA9. Tri-axis hybrid (passive-active) vibration isolator using single crystal piezoelectric actuators. Peter Herdic, Robert Corsaro, Brian Houston, and Robert Baden (Naval Res. Lab., Code 7130, Washington, DC 20375)

Results are presented for a laboratory study of a tri-axis hybrid (active-passive) vibration isolator. A previous report [J. Acoust. Soc. Am. **112** (2002)] described control studies performed using single-axis control. It identified the optimum physical control law by examining the placement order within the device consisting of a passive compliant spring, sensing layer, and piezoelectric actuator. Additionally, actuator materials were evaluated by examining the performance levels of three piezoelectric materials having different nonlinearity values: PZT-4, PZT-5A, and PMN-PT single crystal. The study described here extends this previous work to the required, more complicated case of three-axis control. Described are the development, design, and fabrication of a hybrid device that demonstrates stable, robust, and reproducible “local” vibration reductions of ~ 30 – 45 dB simultaneously in all three excitation axes. These results are found for both single and multiple tone tri-axis disturbances and with negligible out-of-band enhancement, where the total harmonic distortion of the system is less than 0.25%. The control approach, using base acceleration minimization, demonstrates ~ 35 dB of “downstream” performance, which is ~ 25 dB higher than that of the passive element alone. The device is sufficiently small and simple to fabrication that it offers potential for being used in practical applications.

11:30

4aSA10. Minimization of resonance excitation of moving actuator and joint structures. Svetlana Kovinskaya (Mechmath LLC, 14530 Bluebird Trail, Prior Lake, MN 55372, mechmath@mechmath.com) and Yuvenaly Khozikov (IEE Russian Acad. of Sci., St. Petersburg 195265, Russia)

Numerous devices have actuators moving from one immobile position to another under action of applied forces. Repositioning of actuators requires its acceleration and deceleration for faster moving. It is provided by subsequent application of constant forces of opposite directions. Duration of their action is much longer than the resonance periods of actuators or joint structures, and such constant forces excite no resonance. However, a sudden change of the force value results in the resonance excitation. Minimization of resonance excitation can be performed by control of the applied force. Considering the problem in the time domain, one must minimize energy at excited resonance by shaping the force profile while keeping the fast actuator reposition. An effective algorithm for shaping of initial profile based on an iterative procedure is suggested. At the first step of the iteration the profile is deformed (to minimize resonance excitation) by shaping functions which are equal unit outside a short instance. At the second step the deformed initial profile is shaped by the polynomial to meet basic requirements on moving (given distance and zero velocity by the end of reposition). Employment of the developed algorithm significantly reduces the residual vibration and noise radiated by a device.

Session 4aSC**Speech Communication: Phonetic Linguistics: Honoring the Contributions of Peter Ladefoged**

Patricia A. Keating, Chair

*Univ. of California, Los Angeles, Dept. of Linguistics, Los Angeles, CA 90095-1543***Chair's Introduction—8:30***Invited Papers***8:35****4aSC1. 50 years of phonetics with Peter Ladefoged.** Jenny Ladefoged (Phonet. Lab., Univ. of California, Los Angeles, CA 90095-1543)

Ladefoged's many different takes on phonetics, in many countries, will be described.

9:00**4aSC2. Peter Ladefoged and phonetics in the field.** Ian Maddieson (Dept. of Linguist., Univ. of California, Berkeley, Berkeley, CA 94720)

Among many distinctive contributions to phonetics by Peter Ladefoged is an insistence on the immense diversity of phonetic phenomena in the languages of the world, particularly at the segmental level. Because of this Peter has maintained a flexible approach to any scheme of classification or description, adapting to both new approaches and new data. Perhaps more than any other phonetician he has expected to find surprises, and has gone to far corners of the world in search of them. His ground-breaking *Phonetic Study of West African Languages* from 1964 laid out a template for synthesizing a large mass of data which is echoed in later works such as *Preliminaries to Linguistic Phonetics* and *Sounds of the World's Languages*. His widely used Course in Phonetics and other textbooks have shown generations of students the richness of spoken sound. But not only novel research results have marked Peter's contributions to this aspect of phonetics; equally significant are his methodological innovations in taking experimental techniques to the field. Some of the most salient steps along both these paths will be reviewed from personal and professional perspectives.

9:25**4aSC3. The influence of Peter Ladefoged.** Kenneth N. Stevens (Res. Lab of Electron, Dept. Elec. Engin. and Comp. Sci., and Div. Health Sci. and Tech., Massachusetts Inst. of Technol., Cambridge, MA 02139)

The research and publications of Peter Ladefoged and his students have been a strong influence in elevating the field of phonetics to a science, and in bringing together phonetics and phonology. I will cite some examples of how Peter's work and that of his colleagues has influenced the research of many of us at MIT and around the world, where attempts to develop unifying theories often need to undergo continuous revision based on the meticulous articulatory and acoustic data that he reports in his books and other publications. These include his descriptions of the involvement of the tongue root in vowel systems, the fricatives of Mandarin Chinese, the inventory of places of articulation he catalogs in the languages of the world, and the many uses of laryngeal states in providing phonological contrasts. [Supported in part by grants from NIDCD.]

9:50–10:05 Break**10:05****4aSC4. Taking the measure of phonetic structure.** Louis Goldstein (Haskins Labs. and Dept. of Linguist., Yale Univ., 370 Temple St., New Haven, CT 06511, louis.goldstein@yale.edu)

From his earliest work, Peter Ladefoged has insisted that phonetic dimensions and categories (e.g., vowel charts, IPA symbols, features) must be measurable if they are to provide an adequate basis for a universal phonetic representation. This view has led him to two questions, the pursuit of which has been enormously informative and has set the agenda for much contemporary phonetic research. (1) What are the appropriate reference frames in which to make these measurements? The debate triggered by Ladefoged as to the correct choice of reference frame is a key component of current theoretical discussion in phonetics. While the empirical data yield tantalizing hints, their interpretation is far from unambiguous, as can be seen even in some of Ladefoged's earliest work. (2) Are there universal fixed phonetic categories within these reference frames from which individual languages choose (and if so, what are they)? Here, Ladefoged's research has been at the leading edge of a growing consensus that there are, in fact, no fixed categories and that the division of continua into categories in particular languages is largely a random process, constrained by the nature of speech production and perception. [Work supported by NIH.]

10:30

4aSC5. Betting on bits: Contextual influences on the perception of “phonetic categories.” Sarah Hawkins (Dept. of Linguist., Univ. of Cambridge, Sidgwick Ave., Cambridge CB3 9DA, UK)

Peter Ladefoged’s outstanding qualities include meticulous description of differences between “the same” phonetic categories in different languages, and a tendency to explore new research areas, not least by investigating things that others judge better left alone. This paper pays tribute to these qualities by discussing the phonetic category as a theoretical construct in speech perception. By demonstrating that *F1* of a preceding phrase influences whether a word is heard as “bet” or “bit,” Ladefoged and Broadbent (1957) stressed that phonetic categorization is plastic: sense depends on context. Many other contextual influences on sound categorization have been identified, yet are poorly integrated into theory, especially when they extend over long durations, or indicate something other than lexical contrast. I argue from (human) perceptual and (animal) neurophysiological and behavioral data that category identification depends on context and perceived relevance—hence on potential meaning. Human data confirm the centrality of context and meaning to categorization. Animal data show that a sound which is behaviorally significant produces long-lasting changes to response characteristics of single neurons in the primary auditory cortex, and probably earlier. Such data encourage radical re-evaluation of how sensory information is used in understanding speech. [Funded by the Leverhulme Trust.]

10:55

4aSC6. Pronunciation models for conversational speech. Keith Johnson (Dept. of Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720-2650)

Using a pronunciation dictionary of clear speech citation forms a segment deletion rate of nearly 12% is found in a corpus of conversational speech. The number of apparent segment deletions can be reduced by constructing a pronunciation dictionary that records one or more of the actual pronunciations found in conversational speech; however, the resulting empirical pronunciation dictionary often fails to include the citation pronunciation form. Issues involved in selecting pronunciations for a dictionary for linguistic, psycholinguistic, and ASR research will be discussed. One conclusion is that Ladefoged may have been the wiser for avoiding the business of producing pronunciation dictionaries. [Supported by NIDCD Grant No. R01 DC04330-03.]

Contributed Papers

11:20

4aSC7. Phonological acquisition of a Korean child: An acoustic study. Sun-Ah Jun (Dept. of Linguist., UCLA, Los Angeles, CA 90095-1543, jun@humnet.ucla.edu)

Studies on child phonology suggest that there exist phonological universals in the timing of phonological events and the ordering of phonological categories, but the acquisition of speech sounds is influenced by the language-specific aspects of the ambient language such as phonetics, phonology, and the frequency of the sound in child-directed speech. This study investigates a Korean child’s phonological acquisition based on tape recordings of longitudinal data (from 2 months to 2 years, recorded in 1- to 2-week intervals). Special attention is given to the change in prosody and the acquisition of the Korean three-way manner contrast (fortis, aspirated, lenis). It is known that Korean fortis and aspirated obstruents trigger high pitch at vowel onset while lenis obstruents trigger low pitch [Jun (1993), (1998)]. Preliminary results suggest that fortis obstruents are acquired first, followed by aspirated, and then lenis. The segmental properties (e.g., voice onset time, breathy phonation) appropriate for the lenis category were acquired later than the pitch. In addition, unlike the universal tendencies, velar and labial consonants were acquired earlier than alveolar consonants. Factors affecting the order of acquisition, including frequency effect and perceptual salience, will be discussed.

11:35

4aSC8. Comparing the acoustics of voiced and voiceless fricatives in Deg Xinag. Richard Wright, Sharon Hargus, and Julia Miller (Dept. of Linguist., Univ. of Washington, Box 354340, Seattle, WA 98195-4340)

Few studies have looked at the acoustic properties of fricative voicing and place in Native American languages despite their relatively rich fricative inventories of rarely studied fricative places. Deg Xinag, an endangered Athabaskan language spoken in Alaska, provides us with a rare opportunity to investigate fricative place and voicing within a single language: it has eight places of articulation for voiceless fricatives, six of which have voiced counterparts, including some rarely studied place contrasts (e.g., palato-alveolar versus retroflex, uvular versus glottal, lateral versus alveolar). In this study, pre- and post-vocalic fricatives were digitally recorded in the field from eight speakers (two males, six females) using a head-mounted mic to control for distance from the source. The segmental context was also controlled for, the neighboring vowel being [a] in all cases. Each speaker produced four repetitions of each word. Each fricative was analyzed qualitatively using impressionistic transcription and spectrographic investigation, and quantitatively using a set of widely employed measures: (a) widely employed spectral measures (center of gravity, skew, kurtosis, standard deviation, lowest spectral peak), peak and rms intensity of frication, overall duration and duration of voicing. [Work supported by NSF.]

4a THU. AM

Session 4aSP**Signal Processing in Acoustics, Underwater Acoustics, Animal Bioacoustics, Noise, Acoustical Oceanography and Engineering Acoustics: Non-Blind Deconvolution in Acoustics**

George B. Smith, Cochair

Naval Research Laboratory, Code 7183, Stennis Space Ctr., MS 39529-5004

Juliette Ioup, Cochair

Univ. of New Orleans, Dept. of Physics, New Orleans, LA 70148

George E. Ioup, Cochair

*Univ. of New Orleans, Dept. of Physics, New Orleans, LA 70148****Invited Papers*****9:00****4aSP1. Introduction to deconvolution.** Peter A. Jansson (College of Optical Sci., Univ. of Arizona, 1630 E. University Blvd., Tucson, AZ 85721)

Deconvolution tasks will always lie at the frontier of human knowledge in many fields, almost by definition. Rising in the latter 20th century from near disreputability, first to usefulness, then necessity in some disciplines, the varied techniques of deconvolution have assumed an important role in the scientists tool kit. This talk will trace deconvolutions development with examples, including many "firsts," drawn from spectroscopy, radio astronomy, photography, cell biology, color science and diverse other fields. Following a tutorial introduction, detail will be provided on modern super-resolving methods and lesser known topics such as selected-ordinate image (SORI) processing.

9:45**4aSP2. Nonlinear deconvolution approaches.** B. Roy Frieden (College of Optics, Univ. of Arizona, Tucson, AZ 85721)

A review of past methods of digital signal restoration is given. Emphasis is upon nonlinear methods, owing to their extra precision in allowing inequalities and other forms of prior knowledge to be entered as constraints. Some specific methods of this type are maximum entropy, maximum bounded entropy, maximum shannon information, recursive median window filtering, Jansson recursion, and minimum Fisher Information. Algorithms and demonstrations will be given.

10:30**4aSP3. Identification of rapidly time-varying wideband acoustic communication channels.** Weichang Li (MIT and Woods Hole Oceanograph. Inst., M.S. 16, Woods Hole, MA 02543, lwc@mit.edu) and James Preisig (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Wideband acoustic communications in very shallow-water are typically confronted with channels that are both rapidly time varying and sparsely structured [J. C. Preisig and G. Deane, *J. Acoust. Soc. Am.* **116**(4), 2067–2080 (2004)]. Techniques for coherent demodulation require either implicit or explicit identification of channel impulse response. Model based channel identification is able to capture the channel dynamics only when certain persistent excitation condition involving both the transmitted symbol sequence and the sequence of channel estimates holds. This is generally not the case for channels with highly sparse impulse response. On the other hand, dimension reduction techniques such as subspace decomposition and sparse processing, inherently assuming that the channel is stationary over certain time period, often fail to track the rapid channel fluctuations. Therefore both the channel dynamics and redundant dimension have to be addressed simultaneously in order to achieve accurate and robust channel identification. This paper presents methods of model based channel tracking with sparse preprocessing. The limitations of model based tracking and sparse processing when applied separately are derived. The performance gain obtained from the combining approach is then demonstrated through experimental results. [Work supported by ONR Ocean Acoustics.]

11:00–11:30**Panel Discussion**

THURSDAY AFTERNOON, 20 OCTOBER 2005

SALON C, 2:00 TO 3:00 P.M.

Session 4pAAa

**Architectural Acoustics: Distinguished Lecture: From Philharmonic Hall to Number Theory:
The Way to More Diffusion**

Peter D'Antonio, Cochair

RPG Diffusor Systems Inc., 651C Commerce Dr., Upper Marlboro, MD 20774

Trevor Cox, Cochair

RPG Diffusor Systems Inc., 651C Commerce Dr., Upper Marlboro, MD 20774

Chair's Introduction—2:00

Invited Paper

2:05

4pAAa1. From Philharmonic Hall to number theory: The way to more diffusion. Manfred R. Schroeder (Univ. of Göttingen, Rieswartenweg 8, 37077 Göttingen, Germany, mrs17@aol.com)

In September 1962, in the presence of Mrs. Jacqueline Kennedy, Philharmonic Hall in New York was inaugurated—the first building of the new Lincoln Center for the Performing Arts. To address the soon-apparent acoustic problems, Lincoln Center turned to Bell Laboratories for help, and I was asked to join a “committee of experts,” chaired by Vern O. Knudsen of UCLA. My work on Philharmonic Hall, assisted by B.S. Atal, G.M. Sessler, and J.E. West, and later, after my move to Göttingen, by my students D. Gottlob, F.K. Siebrasse, and U. Eysholdt, indicated a need for energetic early *lateral* sound. It was clear that better lateral diffusion could improve the acoustic quality and the feeling of “envelopment” by the sound. Knowing some Galois field mathematics, I lucked upon the design of diffusors which scattered incident waves into broad lateral patterns—but only for a single musical octave. Then, in 1977, during a celebration of the 200th anniversary of Gauss's birth, I heard a talk by André Weil on Gauss sums and quadratic residues and, in a flash, it became clear to me that diffusors based on *quadratic residues* were the answer to broadly scattering waves comprising many musical octaves.

2:50–3:00

Question and Answer Period

4p THU. PM

Session 4pAAb**Architectural Acoustics: Reflections on Reflections**

Peter D'Antonio, Cochair

RPG Diffusor Systems Inc., 651C Commerce Dr., Upper Marlboro, MD 20774

Trevor Cox, Cochair

*RPG Diffusor Systems Inc., 651C Commerce Dr., Upper Marlboro, MD 20774****Invited Papers*****3:05****4pAAb1. Thirty years since diffuse sound reflection by maximum length.** Trevor J. Cox (Acoust. Res. Ctr., Univ. of Salford, Salford M5 4WT, UK) and Peter D'Antonio (RPG Diffusor Systems, Upper Marlboro, MD 20774)

This year celebrates the 30th anniversary of Schroeder's seminal paper on sound scattering from maximum length sequences. This paper, along with Schroeder's subsequent publication on quadratic residue diffusers, broke new ground, because they contained simple recipes for designing diffusers with known acoustic performance. So, what has happened in the intervening years? As with most areas of engineering, the room acoustic diffuser has been greatly influenced by the rise of digital computing technologies. Numerical methods have become much more powerful, and this has enabled predictions of surface scattering to greater accuracy and for larger scale surfaces than previously possible. Architecture has also gone through a revolution where the forms of buildings have become more extreme and sculptural. Acoustic diffuser designs have had to keep pace with this to produce shapes and forms that are desirable to architects. To achieve this, design methodologies have moved away from Schroeder's simple equations to brute force optimization algorithms. This paper will look back at the past development of the modern diffuser, explaining how the principles of diffuser design have been devised and revised over the decades. The paper will also look at the present state-of-the-art, and dreams for the future.

3:35**4pAAb2. Measuring, predicting, and characterizing reflections.** Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774) and Trevor J. Cox (Univ. of Salford, Salford M5 4WT, UK)

Over the past 100 years, since the founding of architectural acoustics by Sabine, there has been considerable effort devoted to studying surface absorption. Over this period, a considerable library of absorption coefficients has been tabulated, based on accepted standards of measurement, and the role of absorptive surfaces is now well understood. In contrast, scientific knowledge about how scattering surfaces should be measured and characterized and how, where, when, and why these surfaces should be used is still evolving. Over the past three decades, how to design, optimize, predict, measure, and characterize the performance of scattering surfaces has been learned. This presentation will describe the procedures developed to measure, predict, and quantify scattering surfaces. Standardized procedures for determining scattering coefficient and correlation scattering coefficient, which indicate the amount of sound scattered away from the specular direction, and the diffusion coefficient, which indicates the uniformity of scattering, will be described and examples given. Three-dimensional balloons, which display the polar distribution of scattered sound, will also be displayed. The goal is that, as these standards become more widely understood, a growing library of coefficients will develop and soon absorption, scattering, and diffusion coefficients will all find their way into architectural acoustic specifications.

4:05**4pAAb3. Reflection of sound from finite-size plane and curved surfaces.** Jens H. Rindel (Oersted-DTU, Tech. Univ. of Denmark, Bldg. 352, DK-2800 Kgs. Lyngby, Denmark)

The author's research on reflectors over nearly 25 years is summarized. The influence of curvature was analyzed by a geometrical model in order to quantify the attenuation by a simple expression. Reflection from a finite-size plate was studied using the Kirchhoff-Fresnel approximation, and the design frequency for a single reflector was derived. Above the design frequency the attenuation due to the finite size can be neglected and the reflection is efficient in the specular direction. The method was extended to the case of a reflector array, and it was demonstrated that the performance of a reflector array can improve if the size of the panels is decreased. The same design frequency applies to a single reflector and a reflector array, but with different meaning; in the latter case the design frequency is the upper limit for useful reflections. This design rule was first used in the refurbishment of the concert hall of the Danish Radio in Copenhagen 1989, and later in many other halls. In order to describe the scattering due to edge diffraction, the directional characteristic of reflections from a finite-size plate has been studied and a simple approximation valid for octave bands has been derived.

4:25

4pAAb4. Virtual reflections in electronic acoustic architecture. Bjorn van Munster (P.O. Box 720, 5400 AS UDEN, The Netherlands b.v.munster@siap.nl)

In the era of the ancient Greeks and Byzantines, the first attempts for increasing reverberation time are noted. In the 1950s, the Ambiophonic system accomplished this by means of an electronic device, for the first time. The early systems only increased the reverberation time by delaying the picked-up reverberation. With the introduction of multichannel feedback-based systems, the reverberation level also could be increased. Later, it was understood that it was important to also fill in the missing reflections, address reflection density, frequency dependence, etc. This resulted in the development of the SIAP concept. Current DSP technology led to the development of a processor whereby density, length, level, and the frequency content can be controlled for different areas in the same room or different rooms, leading to the concept of the acoustic server. Electronic acoustic architecture has become the current state-of-the-art approach for solving acoustic deficiencies in, among others, rehearsal rooms, theaters, churches, and multipurpose venues. Incorporation of complementary passive acoustic solutions provides an optimum solution for all room problems. This paper discusses the utilization of virtual reflections in the new approach of electronic acoustic architecture for different environments. Measurements performed in the Sejong Performing Arts Centre, Seoul, South Korea, show the power of this approach.

4:45

4pAAb5. Reflections in computer modeling of rooms: Current approaches and possible extensions. U. Peter Svensson (Acoust. Group, Dept. of Electron. and Telecommunications, Norwegian Univ. of Sci. and Technol., Trondheim, NO-7491 Trondheim, Norway)

Computer modeling of rooms is most commonly done by some calculation technique that is based on decomposing the sound field into separate reflection components. In a first step, a list of possible reflection paths is found and in a second step, an impulse response is constructed from the list of reflections. Alternatively, the list of reflections is used for generating a simpler echogram, the energy decay as function of time. A number of geometrical acoustics-based methods can handle specular reflections, diffuse reflections, edge diffraction, curved surfaces, and locally/non-locally reacting surfaces to various degrees. This presentation gives an overview of how reflections are handled in the image source method and variants of the ray-tracing methods, which are dominating today in commercial software, as well as in the radiosity method and edge diffraction methods. The use of the recently standardized scattering and diffusion coefficients of surfaces is discussed. Possibilities for combining edge diffraction, surface scattering, and impedance boundaries are demonstrated for an example surface. Finally, the number of reflection paths becomes prohibitively high when all such combinations are included as demonstrated for a simple concert hall model. [Work supported by the Acoustic Research Centre through NFR, Norway.]

5:05

4pAAb6. A largely diffuse small room. George Massenburg (Blackbird Studios, 2806 Azalea Pl., Nashville, TN 37204) and Peter D'Antonio (RPG Diffusor Systems, Inc., Upper Marlboro, MD 20774)

An 8 m × 10 m rectangular room is described that will have broad bandwidth diffusion completely covering 5 of its 6 surfaces, i.e., all walls and ceiling surfaces. The dimensions of the room were optimized for minimal standard deviation of the modal response. The wall surfaces are treated with a large prime, single period, number theoretic diffusor 1.2 m deep that wraps around the entire room. The ceiling consists of a 12 × 13 low frequency diffusor 2 m deep, which is further treated with mid-high frequency diffusors to form a nested, diffractal surface. These surfaces will be described further and illustrated. The room is intended to be deployed as a monitor room for mixing surround sound. It is hoped that the unique combination of a reduced number of specular surfaces and very neutral, wide-band ambience will improve localization (particularly for virtual sources) and offer greater support for (and speed up) the balancing/mixing of multitrack sources. The perceived effectiveness of the diffusor in the ceiling, particularly in the 2 low octaves will be described. The room will also be used as a massively diffuse recording room for various musical formats.

4p THU. PM

Session 4pAB**Animal Bioacoustics, Psychological and Physiological Acoustics and ASA Committee on Standards:
Frequency Weighting For Animal Species**

Larry L. Pater, Cochair

U.S. Army Engineering Research and Development Ctr., 2909 Farber Dr., Champaign, IL 61822

Ann E. Bowles, Cochair

*Hubbs Sea World Research Inst., 2595 Ingraham St., San Diego, CA 92109***Chair's Introduction—1:00*****Invited Papers*****1:05****4pAB1. The 60-dB rule for birds: An example of the application of a weighting function in environmental impact mitigation.**

Ann E. Bowles (Hubbs-SeaWorld Res. Inst., 2595 Ingraham St., San Diego, CA 92109) and Sheyna Wisdom (URS Corp., San Diego, CA 92108)

Over the last decade U.S. Fish and Wildlife Service managers in California have required millions of dollars in added expenditure for NEPA consultation, mitigation barriers, and project delays to reduce the effects of noise from construction activities on endangered passerine birds when the hourly A-weighted Leq is expected to exceed 60 dB. The rule was originally intended to prevent masking of species-typical songs of endangered birds such as the Coastal California Gnatcatcher. However, no research is available to demonstrate the effectiveness of the rule for any noise-related impact. Although A-weighting is probably a conservative estimator of bird exposure in the range from 125 Hz to 8 kHz, it may underestimate exposure at very low frequencies. Its utility as a weighting function has not been tested against other possible weighting procedures, such as use of the species-typical auditory threshold function. Additionally, where sources are intense but intermittent, Leq is unlikely to be a useful metric. These issues should receive more technical scrutiny before the 60-dB rule becomes entrenched in law. It is in widespread use for NEPA consultations, and is already being extended to other species, including large mammals.

1:35**4pAB2. The problem of frequency weighting functions and standards for birds.**

Robert Dooling, Elizabeth Brittan-Powell, Amanda Lauer (Dept of Psych., Univ. of Maryland, College Park, MD 20742), Micheal Dent (Univ. at Buffalo, Buffalo, NY 14260), and Isabelle Noirot (Univ. of Maryland, College Park, MD 20742)

Frequency weighting functions in humans are widely used as a single-figure guess to assess noise problems and aid in making decisions with regard to noise limitations when no other data exist. However, this use of frequency weightings invariably results in a loss of precision in assessing the likelihood of a sound to produce hearing damage or sound annoyance. There is a growing interest in developing frequency weighting functions in animals presumably to assist in judging the risk of hearing damage, interference with acoustic communication, or habitat suitability. Laboratory studies reveal many parallels between humans and animals on a variety of psychoacoustic measures, such as equal loudness contours. However, differences between humans and animals on specific tests argue against using standards developed for humans to gauge the effect of noise on animals. Here we review data which show this same problem exists among birds. That is, the differences in the effects of noise among bird species can be as large as the differences between humans and birds. These results suggest that whereas frequency weighting functions and acoustic standards for a specific species might be useful, generalizing across species is likely not practical.

2:05**4pAB3. Source levels of northern elephant seal vocalizations in-air.**

Stephen J. Insley (Long Marine Lab., UCSC, 100 Shaffer Rd., Santa Cruz, CA 95060, sinsley@ucsc.edu) and Brandon L. Southall (NOAA Fisheries Acoust. Program, Silver Spring, MD 20910)

Accurate measurements of vocalization sound-pressure levels are necessary to determine the acoustical active space of animals in natural and human-altered ambient noise conditions. Despite this basic need, such data are limited or nonexistent for most species. Our study characterized aerial ambient noise and vocalization source levels for northern elephant seals during the breeding season. Subjects were adult males, lactating females, and dependent offspring (pups) at Año Nuevo State Reserve. Source level measurements were made using a Type 1 sound level meter and calibrated microphones on-axis: (1) at 1 m; (2) at several known distances (laser measured); and (3) simultaneously at 1 m and a second known distance. Concurrent ambient noise conditions were measured *in situ* (non-weighted 5 min Leq integrated averages) and recorded for later spectral analysis. Measurements were made at two sites, one relatively noisy and the other relatively quiet, to determine whether animals compensate for higher noise conditions by increasing source levels (Lombard effect). Results indicate a wide range in signal strength, particularly for adult males whose vocalization source levels appear to be correlated with dominance rank and related to ambient noise conditions. The Lombard effect was not observed for adult females or elephant seal pups.

4pAB4. Strategies for weighting exposure in the development of acoustic criteria for marine mammals. James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett Bay Campus Narragansett, RI 02882, miller@uri.edu), Anne E. Bowles (Hubbs-Sea World Res. Inst., San Diego, CA 92109), Roger L. Gentry (NOAA Acoust. Prog., Silver Spring, MD 20910-6233), William T. Ellison (Marine Acoust., Inc., Litchfield, CT 06759), James J. Finneran (Space and Naval Warfare Systems Ctr., San Diego, CA 92152-5000), Charles R. Greene Jr. (Greeneridge Sci., Inc., Santa Barbara, CA 93110), David Kastak (Long Marine Lab., Univ. of California at Santa Cruz, Santa Cruz, CA 95060), Darlene R. Ketten, Peter L. Tyack (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Paul E. Nachtigall (Hawaii Inst. of Marine Biol., Kane'ohe, HI 96744), W. John Richardson (LGL Ltd. Environ. Res. Assoc., King City, ON, Canada L7B 1A6), and Jeanette A. Thomas (Western Illinois Univ., Moline, IL 61265)

The Noise Exposure Criteria Group has been developing noise exposure criteria for marine mammals. Although the primary focus of the effort is development of criteria to prevent injury, the Group has also emphasized the development of exposure metrics that can be used to predict injury with accuracy and precision. Noise exposure metrics for humans have proven to be more effective when they account for psychophysical properties of the auditory system, particularly loudness perception. Usually noise is filtered using the A-weighting function, an idealized curve based on the human 40-phon equal loudness function. However, there are no empirical studies to show whether a comparable procedure for animals will improve predictions. The Noise Exposure Criteria Group panel has proposed to weight noise data by functions that admit sound throughout the frequency range of hearing in five marine mammal groupings—low frequency cetaceans (mysticetes), midfrequency cetaceans, high-frequency cetaceans, pinnipeds in air, and pinnipeds in water. The algorithm for the functions depends only on the upper and lower frequency limits of hearing and does not differentially weight frequencies based on sensitivity within the range. This procedure is considered conservative. However, if the human case may be taken as a model, it is not likely to produce precise predictions. Empirical data are essential to finding better estimators of exposure.

3:05–3:20 Break

Contributed Papers

3:20

4pAB5. Experimental validation of a species-specific behavioral impact metric for underwater noise. Jeremy R. Nedwell, Joe Lovell, and Andrew W. H. Turnpenny (Subacoustech Ltd, Chase Mill, Winchester Rd., Bishop's Waltham, Hampshire SO32 1AH UK)

The regulation of noise from offshore activities in the UK requires a metric allowing the behavioral effects on underwater animals of man-made underwater noise on a wide range of species to be objectively assessed. The dBht(species) metric is a pan-specific metric incorporating the concept of “loudness” by using a frequency-weighted curve based on the species’ hearing threshold as the reference unit for a dB scale. A large number of controlled laboratory measurements have been made of the avoidance of a range of idealized noises, using fish with greatly different hearing as a model. Additional data, of many thousands of individuals, has been obtained by re-evaluation of fish avoidance of a large acoustic fish deflection system at an estuarine power station. All data, irrespective of source or species, indicate a dependence of avoidance reaction on the dBht(species) level. The data indicates three regions, “no reaction” below 0dBht (i.e., below the species’ threshold of hearing), a “cognitive avoidance” region where increasing numbers of individuals will avoid the noise from 0 to 90 dBht, and “instinctive reaction” at and above 90 dBht where all animals will avoid the noise. This probabilistic model allows the behavioral impact of any noise source to be estimated.

3:35

4pAB6. Frequency weighting in the feature extraction process: Effects of parameter choice on generalized perceptual linear prediction coefficients. Patrick J. Clemins and Michael T. Johnson (Speech and Signal Processing Lab., Marquette Univ., P.O. Box 1881, Milwaukee, WI 53201-1881)

The generalized perceptual linear prediction (gPLP) feature extraction model incorporates information about the perceptual abilities of the species under study to generate features relevant to that species. The gPLP feature extraction model is based on the source-filter model of vocalization production and quantifies the general shape of the spectral envelope. gPLP coefficients remove excitation information for vocalizations with a low fundamental frequency, but capture harmonic information for vocaliza-

tions with a higher fundamental frequency through the use of a filter bank analysis component. Frequency warping, frequency masking, and equal loudness normalization, or frequency weighting, are some of the psychophysical phenomena modeled in the gPLP model. The effects of accounting for these phenomena in the feature extraction process are explored using perceptual spectrograms, statistical tests, and classification tasks. Experiments show that using this perceptual information can provide insights during the analysis of vocalizations and improve classification accuracies. The contribution of other feature extraction parameters including the number of cepstral coefficients and number of filters in the filter bank is also examined. [Work supported by the National Science Foundation under Grant No. IIS-0326395.]

3:50

4pAB7. Detection sensitivity of frequency modulated (FM) and constant frequency (CF) signals in temporal masking conditions in the Mongolian gerbils. Tomofumi Taketani, Yasunari Sasaki, and Hiroshi Riquimaroux (Grad. School of Eng., Doshisha Univ., 1-3, Miyakodani, Tataru, Kyotanabe, Kyoto 610-0321, Japan, dtf0757@mail4.doshisha.ac.jp)

The auditory systems would be sensitive to temporal changes in frequency of sounds. We hypothesized that frequency modulation would play an important role in perception. And in this study, the difference in detection sensitivity between frequency-modulated signals (FM) and constant frequency signals (CF) in temporal masking conditions was examined in three Mongolian gerbils (*Meriones unguiculatus*) by GO-NOGO procedure. We focused on their detection sensitivity of three types of signals; upward FM, downward FM, and CF. The frequency range of each FM was from 4 to 7 kHz, and CF was 6 kHz. We set $S+$ that signals were presented between white noise bursts (BW: 0 to 17.5 kHz), and then signals were temporally masked by white noise. In contrast, we set $S-$ in which silence was presented instead of signals. Subjects were trained to detect whether signals were presented or not. After subjects reached criterion (80%), each stimulus was presented randomly and attenuated in amplitude. We investigated their response rate to each $S+$. Result showed that their detection sensitivity to FM was higher than that of CF. It suggests that FM is superior to CF in recognition for subjects. [Research was supported with financial assistance from MEXT.]

Session 4pMU**Musical Acoustics and Architectural Acoustics: Acoustics of Choir Singing II**

Sten O. Ternstrom, Cochair

Kungliga Tekniska Hogskolan, Speech Music and Hearing, Lindstedtsvagen 24, S-10044 Stockholm, Sweden

K. Anthony Hoover, Cochair

*Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA 01776***Chair's Introduction—2:00*****Invited Papers*****2:05****4pMU1. Choir singing in Subsaharan Africa: Acoustic factors of a regional style in southern Mozambique.** Joao Soeiro de Carvalho (Universidade Nova de Lisboa, Av. de Berna 26c, 1069-061 Lisboa, Portugal, jsоеiro@fsh.unl.pt)

Choir singing is a most prominent form of expressive behavior in Subsaharan Africa. A vast majority of expressive modes involves multipart singing, both within the framework of European tonal system as well as other structured ways of combining vocal sounds of different frequencies. Vocal improvisation stands as an important process for the course of performance; individual voice ranges, as well as issues of social status and musical competence, determine the ways musicians participate in performance. Aesthetic validation is often expressed by the use of a nonverbal expressive mode, "kulungwani," a vocal technique involving the action of the lower maxillae and tongue in order to produce a low-frequency interruption of sound emission. Choral singing intonation processes seem to rely on harmonic results, rather than melodic. A regional choral style in southern Africa seems to have developed, where a particular distribution of formant frequencies and an emphasis on low-frequency energy play a significant role.

2:30**4pMU2. Listener perception of and acoustic differences between girl and boy choristers in an English cathedral choir.** David Howard (Dept. of Electron., Univ. of York, Heslington, York, YO10 5DD, UK), and Graham Welch (Univ. of London, London, WC1H 0AL, UK)

For centuries, boy choristers have been singing the top (treble) line in English cathedrals. Girl choristers were first admitted in 1991, and there is a long-running debate as to whether they can carry out this role appropriately. This paper will detail the results from two listening experiments designed to establish whether or not listeners can tell the difference between girl and boy choristers singing the top line in cathedral music. In the first experiment, 189 listeners took part and on average they were able to tell the difference 60% of the time; this was statistically significant over chance. The results suggested that repertoire played a significant part in this ability, and the second experiment was carried out in which the boys and girls sang the same repertoire. Nearly 170 listeners have completed this experiment and, on average, they are making guesses (correct 52% of the time). The paper will discuss the acoustic differences between the stimuli with respect to the singing of boy and girl choristers, while placing the discussion in the context of the English cathedral tradition.

2:55**4pMU3. Making an anechoic choral recording.** Ron Freiheit (Wenger Corp., 555 Park Dr., Owatonna, MN 55050, ron.freiheit@wengercorp.com), John Alexander (3M Ctr., St. Paul, MN 55144-1000), and John Ferguson (St. Olaf College, Northfield, MN 55057)

The utilization of auralization as a tool for acoustic analysis continues to grow and develop. An important element for successful auralization listening experiences is the selection of anechoic source material. In researching the current library of anechoically recorded source material, it was discovered that choral material was not readily available. The Wenger Corporation, St. Olaf College,

and 3M undertook a joint project to create an anechoic choral recording. The paper describes the challenges of this recording project—from the technological, logistical, and musical standpoints—and the solutions that were successfully implemented.

3:20–3:50
Panel Discussion

After the session at the Hilton, the session will move to the Central Lutheran Church which is a short walk from the hotel.

4:10–4:30

The session will continue with a panel discussion “Acoustical issues relevant to choral singing” with Drs. Anton Armstrong and John Ferguson, who direct the St. Olaf Cantorei, and the ASA session organizers.

It will end with a short concert by the St. Olaf Cantorei from St. Olaf College in Northfield, Minnesota. The 90-voice liturgical Choir will be accompanied by organ, brass quartet, handbells, and percussion. The performance, at Central Lutheran Church, which is also open to the public, will feature a hymnsing in which the audience will be invited to participate.

The St. Olaf Cantorei, which regularly performs in the very reverberant St. Olaf College Chapel, recently made an anechoic recording under the sponsorship of the Wenger Company.

THURSDAY AFTERNOON, 20 OCTOBER 2005

CONRAD A, 2:00 TO 4:15 P.M.

Session 4pPA

Physical Acoustics: Topics in Seismic Acoustics

William C. K. Alberts, II, Chair

Univ. of Mississippi, National Ctr. for Physical Acoustics, 1 Coliseum Dr., University, MS 38677

Contributed Papers

2:00

4pPA1. Technique for measurement of characteristic impedance and propagation constant for porous materials. Ki Won Jung and Anthony A. Atchley (Grad. Program in Acoust., Penn State Univ., University Park, PA 16802)

Knowledge of acoustic properties such as characteristic impedance and complex propagation constant is useful to characterize the acoustic behaviors of porous materials. Song and Bolton’s four-microphone method [J. Acoust. Soc. Am. **107**, 1131–1152 (2000)] is one of the most widely employed techniques. In this method two microphones are used to determine the complex pressure amplitudes for each side of a sample. Muehleisen and Beamer [J. Acoust. Soc. Am. **117**, 536–544 (2005)] improved upon a four-microphone method by interchanging microphones to reduce errors due to uncertainties in microphone response. In this paper, a multiple microphone technique is investigated to reconstruct the pressure field inside an impedance tube. Measurements of the acoustic properties of a material having square cross-section pores is used to check the validity of the technique. The values of characteristic impedance and complex propagation constant extracted from the reconstruction agree well with predicted values. Furthermore, this technique is used in investigating the acoustic properties of reticulated vitreous carbon (RVC) in the range of 250–1100 Hz.

2:15

4pPA2. Vibration signature of human footsteps on the ground and in buildings. James Sabatier and Alexander Ekimov (NCPA, The Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677)

Vibration and acoustic responses of the ground and floor in a building to human footsteps were experimentally investigated in the broad frequency range, starting from 0.5 Hz up to ultrasonic frequencies. It was experimentally shown that human footsteps have low- and high-frequency

components. The low-frequency component is well known in the literature and generated by the force component normal to the ground/floor. This force is a function of a person’s weight and a manner of motion (walking, running), or gait. The second high-frequency vibration and acoustical components of human footsteps are generated by the tangential to the ground/floor force from footstep and ground reaction, or friction force. The interactions of these two forces produce the friction noise. The low-frequency forces due to tangential motion between two contacted surfaces generate the high-frequency friction noise. The parameters of this friction noise, such as frequency band and vibration, and sound magnitudes as functions of a distance and walking style (shoes or barefoot) and ground/floor structures were studied. The results of indoor and outdoor tests are presented and discussed. [Work supported by Department of the Army, Army Research Office Contract W911NF-04-1-0190.]

2:30

4pPA3. Nonlinear acoustic behaviors of soils. Zhiqu Lu (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, Coliseum Dr., University, MS 38677, zhiqulu@olemiss.edu)

Soil, like other porous materials such as rock, sand, sandstone, sedimentation, and concrete, exhibits remarkable nonlinear behaviors. In this work, the nonlinear acoustic behaviors of soils are studied by measuring both amplitude and phase frequency responses as a function of dynamic strain. Typical nonlinear phenomena such as resonant frequency shift and harmonics generation are observed. It is also found that the phase difference between the received and reference signals changes with strain. New methods for measuring the nonlinearity parameters, based on resonant frequency shift, phase shift, and phase slope measurements, are developed and compared. This suggests that phase could be a very useful parameter for determining the nonlinearity of porous materials. [This work is supported by the USDA Agricultural Research Service under Specific Cooperative Agreement 58-6408-0-108.]

4p THU. PM

4pPA4. Nonlinear acoustic experiments involving landmine detection: Connections with mesoscopic elasticity and slow dynamics in geomaterials, Part III. Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD 21402) and James M. Sabatier (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

In nonlinear acoustic detection schemes, airborne sound at two primary tones, f_1, f_2 (closely spaced near an 80-Hz resonance) excites the soil surface over a buried landmine. Due to soil wave interactions with the landmine, a scattered surface profile can be measured by a geophone. Profiles at $f_1, f_2, f_1 - (f_2 - f_1)$ and $f_2 + (f_2 - f_1)$ exhibit single peaks; those at $2f_1 - (f_2 - f_1), f_1 + f_2$ and $2f_2 + (f_2 - f_1)$ involve higher order mode shapes for a VS 2.2 plastic, inert, anti-tank landmine, buried at 3.6 cm in sifted loess soil [J. Acoust. Soc. Am. **116**, 3354–3369 (2004)]. Near resonance, the bending (softening) of a family of increasing amplitude tuning curves, involving the vibration over the landmine, exhibits a linear relationship between the peak particle velocity and corresponding frequency. Results are similar to nonlinear mesoscopic/nanoscale effects that are observed in granular solids like Berea sandstone. New experiments show that first sweeping up through resonance and then immediately sweeping back down result in different tuning curve behavior that might be explained by “slow dynamics” where an effective modulus reduction persists following periods of high strain. Results are similar to those described by TenCate *et al.* [Phys. Rev. Lett. **85**, 1020–1023 (2000)]. [Work supported by U.S. Army RDECOM CERDEC, NVESD.]

3:00

4pPA5. Nonlinear acoustic landmine detection: Comparison of “off target” soil background and “on target” soil-mine nonlinear effects. Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD 21402)

When airborne sound at two primary tones, f_1, f_2 (closely spaced near a resonance) excites the soil surface over a buried landmine, soil wave motion interacts with the landmine generating a scattered surface profile which can be measured over the “target.” Profiles at $f_1, f_2,$ and $f_1 - (f_2 - f_1), f_2 + (f_2 - f_1), 2f_1 - (f_2 - f_1), f_1 + f_2$ and $2f_2 + (f_2 - f_1)$ (among others) are measured for a VS 1.6 plastic, inert, anti-tank landmine, buried at 3.6 cm in sifted loess soil. It is observed that the “on target” to “off target” contrast ratio for the sum frequency component can be ~20 dB higher than for either primary. The vibration interaction between the top-plate interface of a buried plastic landmine and the soil above it appears to exhibit many characteristics of the mesoscopic/nanoscale nonlinear effects that are observed in geomaterials like sandstone. Near resonance, the bending (softening) of a family of increasing amplitude tuning curves, involving the vibration over the landmine, exhibits a linear relationship between the peak particle velocity and corresponding frequency. Tuning curve experiments along with two-tone tests are performed both on and off the mine in an effort to understand the nonlinearities in each case. [Work supported by U.S. Army RDECOM CERDEC, NVESD.]

3:15

4pPA6. Nonlinear acoustic detection of weathered, low compliance landmines. James M. Sabatier, W. C. Kirkpatrick AlbertsII (NCPA, Univ. of Mississippi, University, MS 38677), and Murray S. Korman (U.S. Naval Acad., Annapolis, MD 21402)

Two potential impediments to acoustic landmine detection are soil weathering processes and low compliance landmines. To bury landmines, the soil within a mine diameter is removed and replaced such that bulk density, compression, and shear strength all decrease, leaving an acoustic scar detectable with the linear acoustic measurement technique. After a few soil wetting and drying cycles, this contrast is reduced. Linear acoustic mine detection measurements were made on a low impedance contrast landmine before the first rainfall on several occasions over the subsequent 5 years. During this period of time, both the spatial and frequency resolution had to be increased to maintain an on/off target velocity ratio that allowed detection. In some cases, the landmine remains undetectable. To address this, two-tone nonlinear acoustic measurements have been made

on these landmines. When the landmine is detectable with linear acoustics, two tones are broadcast at the frequency where the on/off target velocity ratio is the largest. For the cases when the landmine is undetectable, a two-tone sweep is performed and the operator observes the real-time velocity FFT, noting nonlinear sidebands. Next, two-tone tests are conducted at these sidebands to determine nonlinear velocity profiles. [Work supported by U.S. Army RDECOM, NVESD.]

3:30

4pPA7. A study of the acoustic behavior of a plastic, blast-hardened, antitank landmine. W. C. Kirkpatrick Alberts II, Roger Waxler, and James M. Sabatier (NCPA, Univ. of Mississippi, University, MS 38677)

Mechanical resonances in landmines are often exploited by some acoustic-to-seismic detection schemes. As such, a study of an acoustically active landmine has been performed. Measurements of the mechanical resonances of the pressure plate reveal seven modes below 1.6 kHz and a large frequency shift of the first symmetric mode of the pressure plate, such that it occurs higher in frequency than the first asymmetric mode. The pressure plate is considered to act as an elastically supported thin plate, and the observed perturbation of the first symmetric mode is believed to be caused by the landmine’s blast-hardening mechanism. This is confirmed by a lumped acoustic element model of the system of pressure plate and blast-hardening mechanism. Mass-loading experiments in water show a decrease in resonance frequency with increasing depth that saturates at shallow depths as predicted by Velea *et al.* [J. Acoust. Soc. Am. **115**, 1993 (2004)]. Similar mass-loading measurements in sand reveal an interesting phenomenon; flush burial of the landmine effectively removes the effect of the blast-hardening mechanism on the pressure plate. A qualitative theoretical model of this phenomenon will be presented.

3:45

4pPA8. Effect of particle size on the vibration of plates loaded with granular material. Joseph A. Turner, Liyong Yang, and Wonmo Kang (Dept. of Eng. Mech., W317.4 Nebraska Hall, Univ. of Nebraska, Lincoln, NE 68588)

Acoustic methods of landmine detection are emerging as reliable techniques that are especially well suited for non-metallic mines. These methods rely on the vibrations of the top plate of the mine as it responds to the acoustic excitation. The plate response is complicated by the interaction with the surrounding soil. If the loading soil is granular (e.g., sand), it is expected that particle size will influence the mine response. Experimental results related to this hypothesis are presented here. The first resonant frequency of a sand-loaded plate is measured as a function of sand mass for dry sand of various sizes covering a range from hundreds of microns to a few millimeters. For low values of sand mass, the plate resonance decreases and eventually reaches a minimum. In this regime, there is no dependence on particle size. After the minimum, the frequency increases with additional mass. In this regime, a particle size effect is observed. Models are also presented to support and explain experimental results. The results are expected to impact quantitative aspects of mine detection by acoustic means. [Work supported by ARL.]

4:00

4pPA9. Ultrasound sensing of ground displacement in the presence of surface roughness for landmine detection. Purnima Ratilal, Ninos Donabed, Mark Andrews, Carey Rappaport (Ctr. for Subsurface Sensing and Imaging Systems, Northeastern Univ., 409 Dana Res. Cnt., Boston, MA 02115), and Douglas Fenneman (U.S. Army RDECOM CERDEC Night Vision and Electron., Fort Belvoir, VA 22060)

The presence of surface roughness often degrades the performance of the ultrasound sensor. Here, Green’s theorem is applied to analytically model the field reflected off the vibrating ground in the presence of surface

roughness for a pair of ultrasound transducers. In the presence of surface roughness, the total field measured by the sensor is the sum of the reflected wave from the vibrating ground and that scattered off the roughness elements. The scattered field from the rough surface randomizes the phase of

the received field. The performance of the sensor is examined as a function of the area of ground insonified, angle of incidence and frequency of the transducer, measurement time, correlation lengths, and amplitude of surface roughness.

THURSDAY AFTERNOON, 20 OCTOBER 2005

SALON G, 2:00 TO 5:00 P.M.

Session 4pSC

Speech Communication: Speech Production: Models and Processes (Poster Session)

Peter J. Watson, Chair

Univ. of Minnesota, Dept. of Speech Language Hearing Sciences, 164 Pillsbury Dr., SE, Minneapolis, MN 55455

Contributed Papers

All posters will be on display from 2:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:00 p.m. to 3:30 p.m. and contributors of even-numbered papers will be at their posters from 3:30 p.m. to 5:00 p.m.

4pSC1. Reliability of measurements from ultrasound images. Sarah M. Hardin and Stefan A. Frisch (Univ. of South Florida, 4202 East Fowler Ave., PCD1017, Tampa, FL 33704)

As ultrasound imaging gains popularity in phonetic and speech science research, examining the reliability of measures taken from ultrasound images becomes important. This study assesses the reliability of hand measures of ultrasound data collected by graduate student researchers at the University of South Florida ultrasound imaging lab. Speech production data from two different experiments, "Ultrasound analysis of velar fronting" (Wodzinski, 2004) and "Ultrasound study of errors in speech production" [Frisch, (2003)] were analyzed by two different researchers to obtain inter-rater reliability measures. In addition, one data set was measured twice by the same researcher, once when inexperienced with ultrasound analysis and 7 months later after considerable experience had been gained. The study compared researcher's choice of image to analyze, the measures of the location of articulatory landmarks, and the measures used to quantify articulatory postures. Overall, hand measures of ultrasound images were found to be reliable. There were some differences in the absolute measures obtained, however, different researcher's measures of the same data led to the same conclusions about articulation. In addition, it was found that the measurements of different researchers became more similar to one another with experience.

4pSC2. Midsagittal tongue motion patterns in English: More than pivots and arches. Melissa A. Epstein and Maureen Stone (Biomed. Sci., Univ. of Maryland Dental School, 666 W. Baltimore St., Rm. 5A12, Baltimore, MD 21201, maepstein@alumni.upenn.edu)

The transitional motion patterns of the tongue in the midsagittal plane appear to consist primarily of pivoting and arching. A pivot is a region of minimal motion and an arch is a region of maximal motion. It has been found that the location and number of pivots and arches is dependent on the phonemic content of the articulation. The aim of the present study is to systematically investigate these phonemic effects by constraining the transitions between consonants and vowels using CVCVs. Early results of this work [Epstein and Stone, *J. Acoust. Soc. Am.* **117**, 2575 (2005)] indicate that these two patterns are extreme descriptions of a continuum of shape patterns in which arches and pivots occur simultaneously. This is because the tongue is a nonrigid body and therefore tongue motions are not homogeneous. Canonical tongue shape alone was not enough to indicate which motion pattern would occur. Therefore, this study considers the effect of

tongue height, place, and shape on transitional patterns. The study will track locations of minimum (pivot) and maximum (arch) motions in the vocal tract to determine the effects of physiological and linguistic constraints.

4pSC3. Muscular hydrostat mechanism for lip protrusion in speech. Emi Z. Murano, Maureen Stone (Dept. Biomed. Sci., Univ. of Maryland Dental School, 666 West Baltimore St., 5A-12, Baltimore, MD 21201, ezm001@dental.umaryland.edu), and Kiyoshi Honda (ATR Human Information Sci., Kyoto 619-0288, Japan)

The lip is an organ consisting mostly of muscle similar to the tongue. While the tongue is known as a muscular hydrostat, it is unclear whether the lip is also. In this paper the muscular hydrostat issue was explored from the anatomical and functional point of view using high-resolution static MRI (hr-MRI; 0.125 mm/pixel) and tagged-cineMRI (t-MRI). A 3-D reconstruction of the lips and its muscles was obtained from hr-MRI during sustained vowels /i/ and /u/. The muscular geometry of the orbicularis oris, mentalis, and depressor labii inferior muscles were superimposed onto the principal strains that depicts compression and expansion of the internal tissue obtained from t-MRI. It is shown that (1) orbicularis oris muscle shape can predict both the borderline of glabrous and hairy skin and the manner in which the lips are protruded; (2) the lips volume is almost identical for both speech tasks; and (3) direction and intensity of compression of orbicularis oris and mentalis muscle bundles imply the role of these muscles in the protrusion appearance. These results indicate that the muscular architecture and volume preserving characteristics of the lips are consistent with a muscular hydrostat. [This work was supported by NIH (USA) and NiCT (Japan).]

4pSC4. Smoothing spline analysis of variance (ANOVA) for tongue curve comparison. Lisa Davidson (Dept. of Linguist., New York Univ., 719 Broadway, 4th Fl., New York, NY 10003)

Ultrasound imaging of the tongue is an increasingly common technique in speech production research. One persistent issue regarding ultrasound data is how to quantify them. Researchers often want to determine whether the tongue shape for an articulation under two different conditions (e.g., consonants in phrase-initial versus phrase-medial position) is the same or different. To address this issue, a method for comparing tongue

curves using a smoothing spline ANOVA has been developed (SSANOVA) [Gu, 2002, Smoothing spline ANOVA models]. The SSANOVA is a technique for comparing curve shapes (splines) for two sets of data to determine whether there are significant differences between the curve types. Data sets contain 8–10 repetitions of each of the relevant tongue curves being compared. If the interaction term of the SSANOVA model is statistically significant, then the groups have different shapes. Since the interaction may be significant even if only a small section of the curves is different (i.e., the tongue root is the same, but the tip of one group is raised), Bayesian confidence intervals are used to determine which sections of the curves are statistically different. SSANOVAs are illustrated with some data comparing obstruents produced in word-final and word-medial coda position.

4pSC5. A local vector coding for high-quality voice analysis/synthesis. Masashi Ito and Masafumi Yano (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai 980-8577, Japan, itojin@iec.tohoku.ac.jp)

Line-type spectrum is observed in frequency responses for voiced sound. The spectrum can be characterized by physical parameters: instantaneous amplitude, frequency, and phase for each component. It is difficult to estimate these parameters for natural utterances accurately by power spectrogram because the sound is usually unstationary. A new method, termed local vector coding (LVC), has been proposed to analyze these sounds. LVC assumes that the time-varying parameters for the input sound can be approximated by simple quadratic functions in a short analysis window. Utilizing the phase responses, LVC can estimate not only instantaneous amplitude and frequency for each component of the input but also their time derivatives. The validity of LVC method is examined by using naturally uttered voiced speech. The averaged estimation errors, defined by the differences between the input and resynthesized signals, are lower than 30 dB of the input energy. It indicates that LVC method is very useful for analyzing natural sounds. In addition, since the parameters of each component obtained by LVC method characterize the vowel quality, any kind of voice can be synthesized/transformed by changing each parameter independently, such as a voice of a male adult to a female voice.

4pSC6. The effect of speaking rate on supersegmentals: An acoustic and perceptual analysis. Hsin-Huei Chiou and Peter Watson (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Shevlin 115, Minneapolis, MN 55455)

Rate manipulation has been used to study change in prosodic contrasts such as emphatic stress. Timing contrasts in stressed words are reduced or eliminated when speaking rate is increased. However, reports of intonation and rate change are mixed. Some studies have reported an increase of F_0 movement [M. Stepling and A. Montgomery, *J. Phonetics* **64**, 451–461 (2002)], and other reports have found that F_0 movement is decreased at faster speaking rates [C. Fougeron and S. Jun, *Percept. Psychophys.* **26**, 45–69 (1998)]. This study examined the effect of speaking rate on F_0 and duration in sentences produced with emphatic stress in different sentential position and in declarative and interrogative forms. Essentially, durational contrasts were reduced at faster speaking rates and were more pronounced at slower speaking rates. Intonation, on the other hand, was more pronounced for the fast rate and somewhat reduced for the slow rate. A perceptual component will also be reported that examines a listener's ability to recognize stressed words and mode of sentence production (declarative and interrogative) at different speaking rates.

4pSC7. Tongue movements in vowel-consonant-vowel (VCV) sequences: The effect of consonant length. Anders Lofqvist (Haskins Labs., 300 George St., New Haven, CT 06511, lofquist@haskins.yale.edu)

This study examined the effect of consonant duration on the tongue movement from the first to the second vowel in VCV sequences, where the consonant is a short or long labial nasal consonant. Lip, tongue, and jaw movements were recorded in native speakers of Japanese using a magnetometer system. Measurements were made of the duration, path, and speed of the tongue movement trajectory between the two vowels. The coordination of the onsets of the lip closing and tongue movements was also studied, as well as the relative part of the trajectory that occurred during the consonant and the vowels. Preliminary results show a robust difference in duration between the long and short consonants, with the long ones about twice as long. The duration of the tongue movement was longer in the long than the short consonants. Both the peak and average speed of the tongue movement were slower in the long consonants. The tongue movement path was slightly longer in the long consonants. These results suggest that speakers adjust the tongue movement trajectory so that a similar relationship between the movement and the consonant closure is maintained in both the long and the short consonants. [Work supported by NIH.]

4pSC8. Laryngeal adjustments for devoicing of /h/: A within-speaker study. Laura Koenig (Haskins Labs., 300 George St., New Haven, CT 06511)

Past work has investigated cross-speaker and cross-gender differences in voicing of /h/ in English speakers. The purpose of this study was to see whether a phonetically sophisticated speaker could intentionally alter his /h/ voicing patterns, and, if so, how he would effect any changes. One adult male speaker of American English, a trained phonetician and dialectologist, produced approximately 500 repetitions of intervocalic /h/ in short carrier phrases, with differing vowel contexts and loudness levels. In the first block, the speaker produced the utterances normally (i.e., without specific instructions on /h/ production); in the second, he was explicitly asked to devoice his /h/'s. Results indicated that the incidence of devoiced /h/ increased from 2% in the first block to 69% in the second block. On average, the /h/'s in the second block were produced with higher baseline airflows, indicating more extreme laryngeal abduction. This alone did not account for the speaker's devoicing behavior, however, since the soft condition, which had the lowest peak airflows in the second block, had the most devoicing. Voice source measures will be compared between the two blocks to clarify how the speaker altered his laryngeal setting to achieve more devoicing. [Work supported by NIH.]

4pSC9. Simulating and understanding the effects of velar coupling area on nasalized vowel spectra. Tarun Pruthi and Carol Y. Espy-Wilson (Dept. of Elec. Eng. and Inst. of Systems Res., Univ. of Maryland, College Park, MD 20742)

MRI-based area functions for the nasal cavity of one speaker were combined with the area functions for the vowels /iy/ and /aa/ to study nasalized vowels. The oral cavity was compensated for the falling velum by decreasing the oral cavity area by an amount equal to the increase in the nasal cavity area. Susceptance plots were used along with the simulated transfer functions to understand the effects of velar coupling on nasalized vowel spectra. Susceptance plots of $-(B_p + B_o)$ and B_n suggested significant deviation from the rules suggested by O. Fujimura and J. Lindqvist [*J. Acoust. Soc. Am.* **49**(2), 541–558 (1971)]. In particular, the plots showed that: (1) the frequency of zero crossings of the susceptance plots changes with a change in the coupling area, and (2) formant frequencies need not shift monotonically upward with an increase in coupling area. Further, as a consequence of (1), and the fact that an increase in the coupling area results in a shift of B_n to the right and $-(B_p + B_o)$ to the left, it is postulated that zero crossings of the two plots can cross each other. [MRI data from Brad Story. Work supported by NSF Grant No. BCS0236707.]

4pSC10. Vowel production in Korean, Korean-accented English, and American English. Jimin Lee and Gary Weismer (Waisman Ctr. and Dept. of Commun. Disord., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705)

The current study compares vowel formant frequencies and durations produced by ten native speakers of Korean, those same speakers producing American English vowels, and ten native speakers of American English. The Korean speakers were chosen carefully to have a minimum of 2 years, and maximum of 5 years residence in the United States; all speakers were between the ages of 22 and 27. In addition, the native speakers of Korean were chosen, by means of a small-scale dialect-severity experiment, from a larger pool of speakers to achieve some homogeneity in their mastery of English phonetics. The full vowel systems of both languages were explored, and a rate condition was included (conversational versus fast) to test the hypothesis that the English vowel space is modified by rate differently for native speakers of Korean who produce English, versus native speakers of English. Results will be discussed in terms of language- and rate-induced adjustments of the vowel systems under study.

4pSC11. Common modes of vocal tract articulation for vowels. Brad Story (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, P.O. Box 210071, Tucson, AZ 85721-0071)

The purpose of this study was to investigate the spatial similarity of vocal tract shaping patterns across speakers and the similarity of their acoustic effects. Vocal tract area functions for eleven American English vowels were first obtained from six speakers, three female and three male, using magnetic resonance imaging (MRI). Each speaker's set of area functions was then decomposed into mean area vectors and representative modes (eigenvectors) using principal components analysis (PCA). Three modes accounted for more than 90% of the variance in the original data sets for each speaker. The general shapes of the first two modes were found to be highly correlated across all six speakers. To demonstrate the acoustic effects the modes, both isolated and combined, a mapping between the mode scaling coefficients and $[F1, F2]$ pairs was generated for each speaker. The mappings were unique for all six speakers in terms of the exact shape of the $[F1, F2]$ vowel space, but the general effect of the modes was the same in each case. The results tend to support the idea that the modes provide a common system for perturbing a unique underlying neutral vocal tract shape. [Work supported by NIH R01-DC04789.]

4pSC12. Production of Mandarin tone contrasts by musicians and non-musicians. Terry L. Gottfried and Grace Yin-Hwei Ouyang (Dept. of Psych., Lawrence Univ., Appleton, WI 54912-0599)

Previous research has shown that native speakers of American English who are musicians perform better than non-musicians when identifying and producing the four phonemic tones of Mandarin. The present study corroborates these findings and analyzes acoustic properties of non-natives' tonal imitations. Listeners imitated Mandarin two-syllable word phrases that varied the vowel (/li/, /la/, /lu/) and tone (high-level, mid-rising, low-dipping, high-falling) of the first syllable. Four native Mandarin speakers rated the musicians, on average, to be better in their imitation of Mandarin tone 4 (high-falling) than non-musicians. There were no significant differences between groups in how they were rated on the other three tones. Acoustical analyses revealed that non-natives failed to match native speakers both in differences in initial $F0$ and in $F0$ contour (change from initial to final $F0$) across tones. Imitations by musicians did not show significant acoustic differences from non-musicians, except for tone 4, where musicians' imitations showed a significant decrease in $F0$ from initial to final portions of the syllable; the decrease in $F0$ for non-musicians was smaller and not significant. Creaky voice (often present in natives' tone 3 and 4) was observed in many non-native imitations, but was not restricted to tones 3 and 4.

4pSC13. An experimental investigation of velocity fields in divergent glottal models of the human vocal tract. Byron D. Erath and Michael W. Plesniak (Purdue Univ., 500 Allison Rd., West Lafayette, IN 47907, plesniak@ecn.purdue.edu)

In speech, sound production arises from fluid-structure interactions within the larynx as well as viscous flow phenomena that is most likely to occur during the divergent orientation of the vocal folds. Of particular interest are the flow mechanisms that influence the location of flow separation points on the vocal folds walls. Physiologically scaled pulsatile flow fields in 7.5 times real size static divergent glottal models were investigated. Three divergence angles were investigated using phase-averaged particle image velocimetry (PIV). The pulsatile glottal jet exhibited a bimodal stability toward both glottal walls, although there was a significant amount of variance in the angle the jet deflected from the midline. The attachment of the Coanda effect to the glottal model walls occurred when the pulsatile velocity was a maximum, and the acceleration of the waveform was zero. The location of the separation and reattachment points of the flow from the glottal models was a function of the velocity waveform and divergence angle. Acoustic analogies show that a dipole sound source contribution arising from the fluid interaction (Coanda jet) with the vocal fold walls is expected. [Work funded by NIH Grant RO1 DC03577.]

4pSC14. Blind inversion and the perception/production link. John Hogden (CCS-3, Los Alamos Natl. Lab., Los Alamos, NM 87545) and Philip Rubin (Yale Univ. School of Medicine, New Haven, CT 06511)

Whether humans recover articulator positions from acoustics in the course of speech perception has been debated for years. Some of the arguments against recovering articulator positions are relevant for machine speech recognition as well. However, techniques for blind inversion, in which the mapping from acoustics to articulation is learned without known articulator trajectories, were not considered when some of these arguments were being developed. We discuss the impact of blind inversion on the debate, and conclude that there are pragmatic reasons to recover articulation from acoustics, particularly when blind inversion is used and the mapping from articulation to acoustics is many-to-one.

4pSC15. Some articulatory details of emotional speech. Sungbok Lee, Serdar Yildirim, Murtaza Bulut, Abe Kazemzadeh, and Shrikanth Narayanan (Speech Anal. & Interpretation Lab., USC, Los Angeles, CA 90089, sungbokl@usc.edu)

Differences in speech articulation among four emotion types, neutral, anger, sadness, and happiness are investigated by analyzing tongue tip, jaw, and lip movement data collected from one male and one female speaker of American English. The data were collected using an electromagnetic articulography (EMA) system while subjects produce simulated emotional speech. Pitch, root-mean-square (rms) energy and the first three formants were estimated for vowel segments. For both speakers, angry speech exhibited the largest rms energy and largest articulatory activity in terms of displacement range and movement speed. Happy speech is characterized by largest pitch variability. It has higher rms energy than neutral speech but articulatory activity is rather comparable to, or less than, neutral speech. That is, happy speech is more prominent in voicing activity than in articulation. Sad speech exhibits longest sentence duration and lower rms energy. However, its articulatory activity is no less than neutral speech. Interestingly, for the male speaker, articulation for vowels in sad speech is consistently more peripheral (i.e., more forward displacements) when compared to other emotions. However, this does not hold for female subject. These and other results will be discussed in detail with associated acoustics and perceived emotional qualities. [Work supported by NIH.]

4pSC16. Recognition of voice onset time for use in pronunciation modeling. Abe Kazemzadeh, Sungbok Lee, and Shrikanth Narayanan (Speech Analysis and Interpretation Lab., Univ. of Southern California, University Park Campus, Ronald Tutor Hall 320, Los Angeles, CA 90089)

This study examines methods for recognizing native and accented voiceless stops based on voice onset time (VOT). These methods are tested on data from the Tball corpus of early elementary school children, which includes both native English speakers and Spanish speakers learning English, and which is transcribed to highlight pronunciation variation. We examine the English voiceless stop series, which have long VOT and aspiration, and the corresponding voiceless stops in Spanish accented English, which have short VOT and little aspiration. The methods tested are: (1) to train hidden Markov models (HMMs) based on native speech and then extract the VOT times by post-processing phone-level alignments, (2) to train HMMs with explicit aspiration models, and (3) to train, for each phoneme, different HMMs for native and accented variants. Error rates of 23%–53% for distinguishing phone VOT characteristics are reported for the first method, 5%–57% for the second method, and 0%–36% for the third. The error rates varied depending on the different phones examined. In general, the /p/ and /k/ phones had results that varied more than /t/. These results are discussed in light of each method's usefulness and ease of implementation, and possible improvements are proposed.

4pSC17. Influence of prosodic factors on segment articulations and acoustics in English. Patricia Keating (UCLA, Los Angeles, CA 90095-1543, keating@humnet.ucla.edu) and Taehong Cho (Hanyang Univ., Seoul, Korea)

This study investigated effects of three prosodic factors—prosodic boundary, lexical stress, and accent—on articulatory and acoustic realizations of two CV syllables, /nE/ and /tE/. These syllables occurred at the beginning of trisyllabic English nonwords; their position in the larger phrase (prosodic boundary conditions), and whether they were lexically stressed and/or accented (prominence conditions) were varied. Articulatory measurements included linguopalatal contact (by electropalatography) for both C and V, stop consonant seal duration, and C-to-V contact difference; acoustic measurements include nasal duration and energy for /n/; VOT, burst energy and spectral center of gravity for /t/; and F1, vowel duration and vowel amplitude for /E/. We tested whether domain-initial strengthening occurs in the C and/or the V segments independently of stress or accent conditions. We found that the effects of position and of stress/accent can be distinguished in the production and the acoustics of these syllables. One domain-initial effect (greater consonant contact domain-initially) was complementary to one stress/accent effect (greater vowel opening with stress/accent); in other cases the effects overlapped (greater vowel energy and tendency to longer consonant both domain-initially and with stress/accent); in one case they conflicted (less consonant energy domain-initially, more consonant energy with stress/accent).

4pSC18. Mucosal wave velocity. David A. Berry, Zhaoyan Zhang, and Juergen Neubauer (The Laryngeal Dynam. Lab., UCLA Div. of Head and Neck Surgery, 1000 Veteran Ave., Rm. 31-24, Los Angeles, CA 90095-1794)

Mucosal wave velocity correlates with the elastic properties of the laryngeal mucosa, and can often discriminate normal phonation from pathology. In the past, mucosal wave velocity has been measured in the laboratory using two-point measurements. However, more recent laboratory investigations using human excised hemi-larynges have yielded quantitative vibratory data along a large portion of the medial/superior surface of the vocal fold. These data will be used to probe mucosal wave velocity as a function of (1) location on the medial and surface surfaces of the vocal fold, (2) subglottal pressure, (3) elongation force, and (4) adductory force. Such data reveal significant aspects of the mucosal wave, and the impact of this wave on phonation.

4pSC19. Observations of the near-field structures of the glottal flow. Juergen Neubauer, Zhaoyan Zhang, and David Berry (The Laryngeal Dynam. Lab., UCLA Div. of Head and Neck Surgery, 1000 Veteran Ave., Los Angeles, CA 90095)

The three-dimensional glottal jet was measured downstream from a self-oscillating physical model of the vocal folds using several techniques, including grid-based hotwire anemometry, high-speed flow visualization, and particle image velocimetry. Coherent flow structures were extracted from the spatio-temporal data using principal component analysis (PCA). The glottal jet resembled planar jets through static slitlike orifices. The direction of the jet core axis oscillated in the medial-lateral direction. Also, there was a positive correlation between the dynamics of shear layers around the laminar jet core and flow structures in the transitional region. This finding was similar to the dynamics of planar jets where jet flapping was observed previously. This can be understood in terms of an antisymmetric array of large-scale vortices being convected downstream. This antisymmetric mode of coherent vortex structures acts as an acoustic dipole that could affect the dynamics of the shear layers in the laminar core region. Thus, our observations help to understand the dynamics of the acoustic near-field of the glottal oscillator and suggest a possible feedback mechanism for source-tract coupling.

4pSC20. Effect of intraoral air pressure on the release of an alveolar stop closure. Lan Chen (Speech Commun. Group, Res. Lab. of Electron., MIT, Cambridge, MA 02139)

The effect of air pressure on the release of an alveolar stop consonant closure is simulated with a 2-D finite-element model of the front part of the midsagittal tongue in the vocal tract. Active movement of the tongue is estimated from x-ray microbeam recordings, and intraoral air pressure is also applied to the surface of the tongue as an external force. Air flow in the vocal tract is then coupled to the tongue movement by representing the upper tongue surface as part of the flow boundary. At release, the upward movement of the tongue tip due to the decrease in air pressure introduces an oscillation at the release trajectory of the supraglottal constriction. This oscillation is realized in a quasiarticulatory speech synthesizer HLSyn and causes the appearance of a plateau in the time course of the output airflow. This hesitation in tongue tip movement results in an increase in the duration of the frication noise by about 10 ms, consistent with acoustic data on the stop burst. The simulation shows how the mechanical properties of the tongue and the intraoral air pressure buildup can influence the duration of the frication noise. [Supported by Grant NIDCD DC00075 from NIH.]

4pSC21. An experimental study of the aeroacoustic sound generated by vortex shedding in static and dynamic models of the glottis. Bogdan R. Kucinski, Kenneth J. DeWitt, Ronald C. Scherer (Dept. of Commun. Disord., Bowling Green State Univ., 200 Health Ctr., Bowling Green, OH 43402, bkucins@bgnet.bgsu.edu), and Terry T. Ng (Univ. of Toledo, Toledo, OH 43606)

Flow-visualization with smoke particles illuminated by a laser sheet, performed with a high-speed camera, was used to obtain the flow patterns in two models of the human larynx, both scaled by a factor of 7.5. The first model was static, while the second was a driven dynamic mechanical model, which simulated the motion of the vocal folds during phonation. For the dynamic model, the glottal motion frequency was of the order of the average fundamental frequency of males. It was found that the vortical flow structures were shed at different locations within and downstream the glottis, depending on the transglottal pressure, geometrical configuration and glottal motion frequency. The vortex shedding frequency, which was much higher than the motion frequency of the vocal folds, was measured optically based on the high-speed visualization images, and compared to the frequency peaks in the sound spectra, determined by microphone measurements. Generally, it was observed that the vortex shedding frequency corresponds to the main peak in the sound spectrum. The intensity of noise, however, was found to be dependent of the location of vortical

structures relative to the glottal walls, indicating the presence of quadrupole- or dipole-type sound sources, due to ordered flow structures.

4pSC22. Identification of final fall in subglottal pressure contours of speech utterances. Helen M. Hanson, Janet Slifka, Stefanie Shattuck-Hufnagel (Speech Commun. Group, MIT RLE, 36-511, 77 Massachusetts Ave., Cambridge, MA 02139, helen.hanson@alum.mit.edu), and James B. Kobler (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, Boston, MA 02114)

Subglottal pressure (P_s) contours for speech are described as having three phases: initial rise, constant or declining working phase, and final fall. The current work is part of a project to relate characteristics of the P_s contour to prosodic events. To that end, one must identify the three phases in a P_s contour. In past work, it was found that the initial phase is relatively easy to identify, but the transition from the working phase to final fall is less clear [J. Slifka (2000)]. Confounding issues could include segmental impedance, pitch accents, and phrase and boundary tones, all of which can have local effects on P_s . In this work, it is attempted to control tones and segments at the ends of utterances in order to better identify final fall. Lung pressure is estimated from esophageal pressure (corrected for lung volume). Pilot data from one subject indicate that the beginning of final fall is easier to identify when the phrase and boundary tones are low than when they are high. Results will be presented for additional subjects and it will be attempted to relate them to the distribution of pitch accents. [NIH grant DC04331.]

4pSC23. The articulatory and acoustical characteristics of the “apical vowels” in Beijing Mandarin. Wai-Sum Lee (Dept. of Linguist., Univ. of Hong Kong, Pokfulam Rd., Hong Kong, Hong Kong, wsleeba@hku.hk)

The study investigates the articulatory and acoustical characteristics of the two so-called “apical vowels” in Beijing Mandarin, which have been referred to as “apical anterior vowel” and “apical posterior vowel” by the linguists in China. The “apical posterior vowel” has also been described as a retroflex. The results of an EMA (electromagnetic articulograph) analysis show that both vowels are apical, with the tip of tongue approaching the alveolar region for the “anterior vowel” and the postalveolar region for the “posterior vowel.” The “posterior vowel” is pharyngealized, as the body of tongue in particular the posterodorsal portion is pulled backward toward the pharynx. Acoustical data obtained using the CSL4400 speech analysis software show that the two “apical vowels” have similar $F1$ value. The $F2$ value is slightly larger for the “posterior vowel” than “anterior vowel.” Thus, the correlation between a larger $F2$ and the advanced tongue position is not applicable to these “apical vowels.” The main difference between the two “apical vowels” is in $F3$, where the value is much smaller for the “posterior vowel” than “anterior vowel.” It is assumed that the smaller $F3$ value for the “posterior vowel” is due to pharyngealization.

4pSC24. Vocal tract length development during the first two decades of life: A magnetic resonance imaging study. Hourii K. Vorperian (Waisman Ctr., Univ. of Wisconsin-Madison, Madison WI 53705), Moo K. Chung (Univ. of Wisconsin-Madison, Madison, WI 53705), Lindell R. Gentry (Univ. of Wisconsin Hospitals and Clinics, Madison, WI 53792), Ray D. Kent, Celia S. Choib, Reid B. Durtschi and Andrew J. Ziegert (Univ. of Wisconsin-Madison, Madison, WI 53705)

As the vocal tract length (VTL) increases more than twofold from infancy to adulthood, its geometric proportions change. This study assesses the developmental changes of the various hard and soft tissue structures in the vicinity of the vocal tract (VT), and evaluates the relational growth of the various structures with VTL. Magnetic resonance images from 327 cases, ages birth to age 20, were used to secure quantitative

measurements of the various soft, cartilaginous and bony structures in the oral and pharyngeal regions using established procedures [Vorperian *et al.* (1999), (2005)]. Structures measured include: lip thickness, hard- and soft-palate length, tongue length, naso-oro-pharyngeal length, mandibular length and depth, and distance of the hyoid bone and larynx from the posterior nasal spine. Findings indicate: (a) ongoing growth of all oral and pharyngeal structures with changes in growth rate as a function of age; (b) a strong interdependency between structure orientation and its growth curve; and (c) developmental changes in the relational growth of the different VT structures with VTL. Findings provide normative data on the anatomic changes of the supra-laryngeal speech apparatus, and can be used to model the development of the VT. [Work supported by NIH-NIDCD Grants R03-DC4362 R01-DC006282, and NIH-NICHHD P30-HK03352.]

4pSC25. Improving automatic speech recognition via better analysis and adaptation. Douglas O’Shaughnessy, Wayne Wang, William Zhu, Vincent Barreard, T. Nagarajan and R. Muralishankar (INRS-EMT, 800 de la Gauchetiere West, Ste. 6900 Montreal, QC, Canada H5A 1K6)

One way to improve automatic speech recognition (ASR) systems is to reduce the mismatch between system training and operating conditions, as such mismatch seriously degrades performance. We have developed model adaptation techniques able to adapt to various speech environments without modifying ASR systems, and have developed an appropriate feature transformation scheme for the Mel-frequency cepstral coefficients (MFCC), a popular front-end feature of ASR systems. We use maximum a posteriori model adaptation and a method based on Bayesian parametric representation. Feature transformation aims to maximize the desired source of information for a given speech signal in the front-end features and to minimize undesired sources. Frequency-domain autoregressive modeling and a segmentation algorithm are being developed, e.g., to segment a speech signal into syllablelike units. We also introduce a new speech-processing front-end feature that performs better than the existing MFCC, as well as a log-energy dynamic range normalization technique for ASR in adverse conditions. In addition, we have developed a continuous ASR method that exploits the advantages of syllable and phoneme-based subword unit models. [Work supported by NSERC-Canada and Prompt-Quebec.]

4pSC26. The effects of the glottal geometry on intraglottal pressure distributions. Li Sheng (Dept. of Biomed. Eng. School of Life Sci. and Technol., Xi’an Jiaotong Univ., 28 West Xianning Rd., Xi’an, 71004R, P.R. China), C. Ronald Scherer (Bowling Green State Univ., Bowling Green, OH 43403), MingXi Wan, and SuPin Wang (Xi’an Jiaotong Univ., China)

The purpose of this study is to explore the effects of the glottal geometry on intraglottal and transglottal pressures using a Plexiglas model and a commercially computational fluid dynamics code, FLUENT. Nine glottal angles (uniform, as well as convergent and divergent 5, 10, 20, and 40 deg), 18 inferior vocal-fold angles varied from 87.5 to -10 deg, and 19 superior vocal-fold surface angles varied from -85 to 45 deg for uniform, convergent 10- and divergent 10-deg glottal angle, and a wide range of entrance radii varied from 0.26 to 0.005 cm for different divergent glottis were selected separately to examine their pressure distribution effects. The empirical data were supported by computational results using FLUENT. The results suggest that the 10-deg divergence angle may correspond to least flow resistance, the vocal-fold surface pressures are essentially independent of the inferior and superior vocal-fold surface angles realistic for human phonation, and a small glottal entrance radius tends to lower the transglottal pressure, move the minimal pressure near the glottal entrance more upstream, and make the pressure dip more negative in value. These results suggest that the glottal geometry should be well specified when using physical, mathematical phonatory models.

Session 4pSP**Signal Processing in Acoustics, Underwater Acoustics, Animal Bioacoustics, Noise, Acoustical Oceanography and Engineering Acoustics: Blind Deconvolution in Acoustics**

George B. Smith, Cochair

Naval Research Lab., Code 7183 Stennis Space Center MS 39529-5004

George E. Ioup, Cochair

Univ. of New Orleans, Dept. of Physics, New Orleans, LA 70148

Juliette Ioup, Cochair

*Univ. of New Orleans, Dept. of Physics, New Orleans, LA 70148****Invited Papers*****3:15****4pSP1. Blind deconvolution and source separation in acoustics.** Leon H. Sibul, Michael J. Roan, and Christian M. Coviello (Appl. Res. Lab., Penn State Univ., P.O. Box 30. State College, PA 16804-0040, lhs2@psu.edu)

Blind deconvolution (BDC) and blind source separation (BSS) are active research topics with many important applications in acoustics. The goal of deconvolution is to recover original input signal from the output of a convolution filter. In blind deconvolution details of the convolution filter and input signals are not known. The fundamental assumption in BDC is that the input signal is a non-Gaussian stochastic process. A topic closely related to BDC is BSS. BSS is a process that is an inverse operation to a mixing process. In BSS it is assumed that inputs to the mixing systems are statistically independent stochastic processes, where only one input may be Gaussian, others must be non-Gaussian. Standard criterion functions for BDC and BSS are reviewed. Limitations of the second-order statistics and need for higher-order statistics (HOS) or information theoretic criteria that lead to nonlinear optimization algorithms are pointed out. Advantages of various information theoretic criteria for BDC and BSS are discussed. Because gradients of these criteria are nonlinear, resulting optimization algorithms are nonlinear. Linear and non-linear algorithms for BDC and BSS are examined. [Work supported by ONR Codes 321US and 333.]

4:00**4pSP2. Blind deconvolution in ocean sound channels via artificial time reversal.** David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48109-2133), Karim G. Sabra (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0238), and Mark R. Stevenson (NATO Undersea Res. Ctr., La Spezia, Italy)

A signal that travels through a sound channel is typically distorted when recorded by a remote listener because of interference arising from multiple propagation paths. If the distorted signal is recorded with a transducer array, a novel passive acoustic technique, artificial time reversal (ATR), can be used for blind deconvolution to remove sound-channel-induced distortion and recover the original signal. The technique is based on artificial backpropagation of the recorded signals using source-to-array Green's functions synthesized directly from the recorded signals. The technique relies on separating out the contribution of a propagating mode or ray path having a phase that depends linearly on frequency. Analysis of the technique shows that it is simultaneously a matched filter and an inverse filter in the limit of high signal-to-noise ratio. Computational examples employing a single source are presented and compared with results obtained from oceanic measurements. The temporal correlation between an ATR-compressed signal and the original signal approaches 100% in the computational examples, and commonly exceeds 90% for the oceanic data. Potential extension of this technique to source-array-range estimation, and to multiple sources emitting simultaneously in the same frequency band are also presented. [Work supported by ONR.]

4:30**4pSP3. Maximum-length-related sequences and a specialized deconvolution for multiple acoustic channel estimation.** Ning Xiang (Architectural Acoust. Program, School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180) and John N. Daigle (Univ. of Mississippi, University, MS 38677)

A wide variety of acoustic systems have multiple sources and receivers. This paper discusses a deconvolution technique for making acoustic measurements simultaneously with multiple sources. The deconvolution is based on a collection of known excitation signals of the maximum-length sequence (MLS) related classes, widely used in the spread spectrum communication, but less well known in acoustics applications. This technique is particularly valuable when characterization of multisource, multireceiver system must be accomplished in a limited time period. Since lengths of periodic MLS-related source signals are not suitable for FFT-based deconvolutions, and the fast MLS transform based on fast Hadamard transform is not applicable to the MLS-derived combined signals, our effort has been given to an efficient deconvolution algorithm. The simulation and experimental results demonstrate the feasibility, fidelity, and potential applications of the deconvolution techniques in characterizing acoustic systems.

4pSP4. Mechanisms proposed for spectrogram correlation and transformation deconvolution in FM bat sonar. James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI 02912)

Big brown bats use time/frequency distributions to represent FM biosonar pulses and echoes as a consequence of reception through frequency tuned channels of the inner ear and subsequent processing by similarly tuned neural channels in the auditory pathway. Integration time is 350 μ s, yet delay resolution is 2–10 μ s, which must be based on detecting changes in the echo spectrum caused by interference between overlapping reflections inside the integration time. However, bats perceive not merely the echo interference spectrum but the numerical value of the delay separation from the spectrum, which requires deconvolution. Because spectrograms are the initial representation, this process is spectrogram correlation and transformation (SCAT). Proposed SCAT deconvolution mechanisms include extraction of echo envelope ripples for time-domain spectrometry, cepstral analysis of echoes, use of coherent or noncoherent reconstruction with basis functions, segmentation of onsets of overlapping replicas at moderate to long time separations, and localization of the occurrence of spectral interference ripples at specific times within dechirped spectrograms. Physiological evidence from single-unit recordings reveals a cepstral-like time–frequency process based on freqlets, both single-unit and multiunit responses reveal which may prove to be time-domain basis functions, and multiunit responses exhibit modulations by onset and envelope ripple. [Work supported by NIH and ONR.]

Contributed Paper

5:30

4pSP5. Deconvolution for interference cancellation in underwater acoustics using Gibbs sampling. Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ 07102)

Loud interfering sources often hinder efforts to localize a quiet source in shallow-water environments. Interference cancellation methods are sought that remove the field component from the extraneous sources, leaving the contribution of the source of interest intact as much as possible; this latter contribution can then be used for source localization using matched-field methods. In this work we develop a method that estimates

locations and spectral characteristics (amplitude and phase) of multiple sources; the number of sources is estimated as well. The procedure is optimized with Gibbs sampling and is, in essence, a deconvolution process, since source amplitude and phase information is recovered. We find that the parameters of the strongest sources can be reliably estimated within a few iterations. The field components attributed to those interfering sources are then estimated and removed from the total acoustic field; the remaining field component can be readily used for localization of the quiet source(s) using a simple Bartlett processor. [Work supported by ONR.]